Brief Summary of BCM

BY

Sam Hassan

The Nortel Networks Business Communications Manager (BCM) system provides private Network and telephony management capability to small and medium-sized businesses. The Business Communications Manager integrates voice and data capabilities, VoIP gateway functions and quality of service (QoS) data-routing features into a single telephony system. Business Communications Manager is a compact system that allows you to create and provide telephony applications for use in a business environment.

The Business Communications Manager current version software is 3.6 and is available in the Following three product configurations:

• BCM200
• BCM400 Standard (STD)
• BCM400 Redundant feature option (RFO), which includes dual, hot-swappable power supply, dual chassis cooling fan and RAID (Redundant array of independent disks) mirrored hard disk drive redundancy.

The main component of the BCM is the base unit. The base unit contains the following powerful parts:

- Intel® Pentium® III 700 MHz (BCM400) or a Pentium Celeron 850MHz (BCM200), CPU
- 256 MB SDRAM
- 20 GB hard drive
- 2 10/100 BaseT Ethernet ports (on-board)
- 1 V.90 embedded modem (North America units only)
- 2 PCI slots (one used by the Media Services Card and one for adding a WAN interface card)
- 4 media bays in BCM400, 2 media bays in BCM200
- 350 watt power supply (PS)
- Windows NTE 4.0.

The BCM base unit controls all tasks, including call processing, voice messaging and data Routing. The base unit also contains telephony hardware and data networking hardware Components. Making and receiving calls is crucial to any business. The call processing capability of BCM has been designed to process calls even when the Windows NTE operating system is out of service.

Software Key codes
All BCM applications are loaded onto the system when it is shipped; some of them are standard and work immediately after the system is installed, while other applications are optional and must be enabled using software key codes.
The following table describes the software applications that are standard and any other application required a key code.

<table>
<thead>
<tr>
<th>Standard Applications</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Telephony</td>
<td>√</td>
</tr>
<tr>
<td>Auto Attendant</td>
<td>√</td>
</tr>
<tr>
<td>Call Detail Recording</td>
<td>√</td>
</tr>
<tr>
<td>Hunt Groups</td>
<td>√</td>
</tr>
<tr>
<td>BRI S/T</td>
<td>√</td>
</tr>
<tr>
<td>PRI</td>
<td>√</td>
</tr>
<tr>
<td>Integrated QoS Routing Plus</td>
<td>√</td>
</tr>
<tr>
<td>WAN Routing</td>
<td>√</td>
</tr>
</tbody>
</table>

**BCM200 platform base chassis**

The BCM200 platform base chassis design provides multiple points of access to the base platform hardware components. The front of the chassis has three assemblies that house one base function tray and two media bay modules (MBMs). The rear of the chassis provides mount points for the Fan and power supply. The rear of the chassis also has a removable panel to provide access to the...
Hard disk. The top cover has a removable section to allow access to the cables, connectors, power supply, hard disk and cooling fan. The **BCM200** cannot be expanded using an Expansion Cabinet and is designed to meet the needs of customers with **32 digital extensions** or fewer users per system and up to 90 IP phones.

**BCM400 Platform Base Hardware**

The BCM400 platform is available in either a standard (STD) or RFO configuration. Nortel Networks recommends that you know the location of the different components before attempting to install or maintain the system. The BCM400 platform base chassis design provides multiple points of access to the base platform hardware components. The front of the chassis has two, sliding tray assemblies that house the base function tray and advanced function tray. Four bays accommodate the media bay modules (MBMs).

When coupled with an Expansion Cabinet, the system can grow to support a maximum of **192 digital stations** or up to **90 IP telephones**; however, this is configuration is lines (trunks) dependent. **240 stations** is the maximum capacity, with a mix of IP and digital stations, when 100% IP trunking is used.
Business Communications Manager Expansion Unit

The Business Communications Manager Expansion unit contains six additional bays for media bay modules. The Business Communications Manager Expansion unit is available for use only with the BCM400.

The supplied DS256 cable is 5 m (16 ft.) long. Use of any other cable is not supported. The cable connects into a DS256 port on the MSC of the base function tray, and into a DS256 port on the Center panel of the expansion unit.

Platform Media Bay Module Bays and Backplane

The number and configuration of the media bay modules depend on the number of bays available in the platform base chassis and DS30 system resources. The BCM200 platform base chassis provides two media bay module bays. The BCM400 platform base chassis provides four media Bay module bays.
All media bay module bays must contain either a media bay module or a MBM filler blanking plate. Populate the bays with media bay modules as required. Fill unpopulated media bay module Bay openings in the platform base chassis with the MBM filler blanking plate.

**Media services card (MSC)**

The Media Services Card (MSC), a PCI card, performs call processing and media processing of the voice channels for the Business Communications Manager system, including the VoIP trunks. This card also offers connections for auxiliary features, including external, customer-supplied Hardware for paging and music-on-hold.

The MSC faceplate offers the following optional connections:

- **DS256 connector (BCM400)** — The Business Communications Manager expansion unit connects to the base function tray through the DS256 jack on the MSC faceplate. The DS256 cable to make this connection is provided with the purchase of a BCM1000e Expansion chassis.
- **Auxiliary ringer jack** — the base function tray uses the auxiliary ringer jack to control the Cadence of an auxiliary ringer (customer supplied). You must use this output in a low current, low voltage application only. Do not use this output for switching the auxiliary ringer directly.
- **Page relay jack** — when you use the page signal output jack to connect an external paging Amplifier, you also use the page relay jack. The page relay jack connects a floating relay Contact pair. The base function tray uses this jack to control the external paging amplifier.
- **Page output jack** — the base function tray uses the page output to connect an internally Generated voice paging signal to an external paging amplifier (customer supplied).
- **Music on hold jack** — the base function tray uses the Music on hold input to connect an External music source that supplies a signal to held lines (music on hold) or telephone speakers (Background music). The input source can be any available radio or music source approved for Connection to the network.
### Business Communications Manager maximum capacity

<table>
<thead>
<tr>
<th>Voice capabilities</th>
<th>Voice applications suite</th>
</tr>
</thead>
<tbody>
<tr>
<td>• Over 150 core PBX telephony features</td>
<td>• Auto attendant and custom call routing</td>
</tr>
<tr>
<td>• ISDN trunks – PRI and BRI</td>
<td>• CallPilot voicemail</td>
</tr>
<tr>
<td>– Analogue available in some markets</td>
<td>• CallPilot 2.0 unified messaging for:</td>
</tr>
<tr>
<td>• Voice networking over ISDN</td>
<td>– Outlook, Outlook Express, Lotus Notes, Qualcomm,</td>
</tr>
<tr>
<td>– QSig, DPNSS and MCDN</td>
<td>Eudora Pro, Novell Groupwise, Netscape Messenger</td>
</tr>
<tr>
<td>• 5 digital telephone models – Norstar based</td>
<td>• Fax messaging</td>
</tr>
<tr>
<td>– M7000, T7100, T7208</td>
<td>• BCM contact centres</td>
</tr>
<tr>
<td>– T7316e and T24 key expansion module</td>
<td>– 1 to 80 active agents</td>
</tr>
<tr>
<td></td>
<td>– 2 to 50 queues</td>
</tr>
<tr>
<td></td>
<td>– 10 to 150 recorded announcements</td>
</tr>
<tr>
<td></td>
<td>– Integrated MIS reporting for real time and historical data</td>
</tr>
<tr>
<td></td>
<td>– IP and soft wall boards</td>
</tr>
<tr>
<td></td>
<td>– Web voice button</td>
</tr>
<tr>
<td></td>
<td>• TAPI 2.1 server for computer telephony integration</td>
</tr>
<tr>
<td></td>
<td>• MRS 100 based interactive voice response support (integrated)</td>
</tr>
<tr>
<td></td>
<td>• CDR raw data output via FTP or in real time, over IP</td>
</tr>
</tbody>
</table>
| IP Internet telephones and networking | • Support for 12002, 12004 and 12050 Internet telephones.  
• Support for up to 60 VoIP trunks  
• Voice networking between BCM systems, Meridian 1 IP enabled systems via ITG or Succession 1000, using MCDN protocols over IP  
• Support for Symbol IE 802.11 wireless IP telephones, with T7100 digital set feature emulation. |
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>DECT mobility*</td>
<td>• Up to 32 cordless handsets and 8 radio base stations</td>
</tr>
</tbody>
</table>
| Data capabilities                    | • IP/IPX router  
• RIP, RIP2, OSPF, static  
• DHCP server  
• DNS cache server  
• Web cache server  
• NAT - public to private network address translations  
• Netlink Manager for WAN back up |
| WAN access services                 | • PPP over frame relay over X.21 or V.35  
• PPPoE  
• PPP/IP over demand over ISDN |
| Security services                   | • IPSec - 3DES 128 bit encryption  
• PAP/CHAP - password and challenge handshake authentication protocol  
• RAS  
• Integrated firewall - stateful or basic packet filtering |
| VPN services                        | • Contivity server support  
• Up to 16 VPN tunnels  
• BCM to BCM, BCM to Contivity, BCM to client PC (SOHO)  
• Contivity Extranet client supports up to 16 simultaneous IPSec clients  
• PPTP - up to 10 tunnels |
| System management and IP services   | • Web browser system management interface  
• SNMP traps  
• Network configuration manager for multi site management and configuration of up to 2000 BCM nodes  
• DiffServ quality of service  
• DHCP server |
## Technical Specifications

### Base unit

<table>
<thead>
<tr>
<th>Physical dimensions</th>
<th>BCM400</th>
<th>BCM200</th>
</tr>
</thead>
<tbody>
<tr>
<td>Depth</td>
<td>16.3 in; 41.5 cm</td>
<td>16.3 in; 41.5 cm</td>
</tr>
<tr>
<td>Width</td>
<td>17.5 in; 44.5 cm</td>
<td>17.5 in; 44.5 cm</td>
</tr>
<tr>
<td>Height</td>
<td>7.1 in; 18 cm</td>
<td>3.5 in; 9 cm</td>
</tr>
<tr>
<td>Weight</td>
<td>Std. 20.4 lbs; 9.2 Kg</td>
<td>Std. 10.3 lbs; 4.6 Kg</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Components</th>
<th>BCM400</th>
<th>BCM200</th>
</tr>
</thead>
<tbody>
<tr>
<td>Media Bay/Module Bays</td>
<td>Four</td>
<td>Two</td>
</tr>
<tr>
<td>CPU Processor</td>
<td>Intel Core i7 1.2 GHz</td>
<td>Intel Core i7 1.2 GHz</td>
</tr>
<tr>
<td>Memory (RAM)</td>
<td>256 MB SDRAM</td>
<td>256 MB SDRAM</td>
</tr>
<tr>
<td>Hard Drive</td>
<td>20 GB IDE</td>
<td>20 GB IDE</td>
</tr>
</tbody>
</table>

### Mechanical requirements

- **NS13 (60 Hz/50 Hz)** compliant for Transportation and Operational Vibration and Shock resistance per IEC 68-2-27, Transportation Source to IEC 68-2-55 and Unpackaged Drop to ISTA Project 3A.

### Regulatory compliance

#### Electromagnetic emissions

- **Radiated**
  - Australia AS/NZS Class A
  - North America C63.22 Class A
  - United Kingdom EN55022 Class A

- **Conducted Power Leads**
  - Australia AS/NZS Class A
  - North America C63.22 Class A
  - United Kingdom EN55022 Class A

#### Immunity (Narrow band RF interference)

- **Radiated**
  - North America Customer driven (based on EN 61000-4-3)
  - United Kingdom/International EN55024: 1998

- **Conducted**
  - North America Customer driven (Based on EN 61000-4-8)
  - United Kingdom/International EN55024: 1998

### Immunity to electrostatic discharge

- **Indirect**
  - No Functional Impairment up to +/- 8 kV
  - No damage up to +/- 20 kV

- **Direct**
  - Un-terminated I/O ports and Connectors: No damage up to +/- 8 kV
  - Mated Connector and Cords: No functional impairment up to +/- 15 kV, No damage up to +/-15 kV

### Network protection

- Australia T50/81/9/91/04/01
- North America/CEA FCC Part 68, CS-03 Issue B
Digital Drop and Insert MUX Media Bay Module (DDIM)
- One 8-pin RJ-45 modular jack
- T1 trunk interface with integrated CSU
- 24 B channels with T1 interface (supports DSX1 and DS1 interfaces)
- LEDs: Power, Status, In-service, Loop-back Test, Receive Alarm, Receive Error, Transmit Alarm, Transmit Error
- LEDs: Transmit, Receive, RTS, CTS, DCD, DSR, TM
- V.35-0625 miniature connector
- Cable: V.35, BCI, WANI, Nortel Networks, E1-60 cable
- Utilizes two DS-30s

Basic Rate Interface Media Bay Module (BRIM S/Y)
- Four modular RJ-45 jacks
- Supports four S/T interfaces (8 b-diameters)
- T-Interface to connect to a T1 device or a S-Interface to connect DS1 terminals
- Supports ETSI and National E164 BRI
- Two LEDs: Power, Status
- Utilizes 1 DS-30

Global Analog Trunk Module (GATM 4)
- One Analphak modulator for analog North American, UK, and Australia Standard
- Four loop-start CASS/CMS lines plus one auxiliary port for V.90 modem, fax, analog telephone connection, or Power-Fail Transfer
- Two LEDs: Power, Status
- Utilizes 1 DS-30

Global Analog Trunk Module (GATM 8)
- One Analphak modulator for analog North American, UK and Australia Standard
- Eight loop-start CASS/CMS lines plus one auxiliary port for V.90 modem, fax, analog telephone connection, or Power-Fail Transfer
- Two LEDs: Power, Status
- Utilizes 1 DS-30

4x66 Combo Module (66 Caller ID trunks + 16 Station sets, CMB6616)
- Modular RJ-45 jacks to support four loop-start CASS/CMS lines plus one auxiliary port for V.90 modem, fax, analog telephone connection, or Power-Fail Transfer
- One Analphak modulator (26 pair) to support 16 digital phone ports
- Individual interfaces are currently limited to 80 mA
- Two LEDs: Power, Status
- Utilizes 1 DS-30

Fiber Expansion Media Bay Module (TEM)
- Six fiber ports
- Connects up to six Nonstar fiber-based trunk or station modules
- Two LEDs: Power, Status
- Utilizes 1 DS-30 for each Nonstar fiber trunk or station module connected (up to 6)

DECT Mobility Media Bay Module (DECT, A-Law & Mu-Law — Europe & Taiwan only)
- Four BRI/ISDN 5 loops
- Supports up to 8 radio base stations through eight RJ-45 connectors
- Each BCI system can support one DECT module with a maximum of 32 DECT handsets
- Each DECT radio base station supports up to 4 simultaneous calls
- A maximum of 8 simultaneous calls can be established between DECT handsets and the BCM core
- Available in either A-Law or Mu-Law compliant firm version of firmware
- Two LEDs: Power, Status
- Utilizes 1 DS-30
Data networking components
Embedded v. 0.2 Modern (North America only)
- North America and UK only for Dial Backup or Remote Admin
- V.92 56 Kbps ITU standard
- V.92 33.6 Kbps ITU standard
- RS-485 contactor
- V.42/V.42bis 2:1 error control
- V.42/V.42bis V.5 data compression
- Capable of receiving data at 16 Kbps and sending data at 3.2 Kbps
10/100 Ethernet LAN interface
- 10/1000BASE-T Ethernet ports (on board the mainboard)
- Supports IEEE 802.3 Ethernet frame format
- Uses Carrier Sense Multiple Access with Collision Detection (CSMA/CD)
- 100BASE-TX with RJ-45 connector
- 10/100 Auto-sensing
- Full duplex support
- Fast LAN to LAN routing
- LAN traffic smoothing
- ISK support
- PRIME (enabled using keycode)

WAN interface
Two port PCI card (field installable)
- Each port can be independently configured to Frame Relay or PPP
- STAC compression is available
- One serial port/2 ports (V.92) and one RJ-11 parallel (Integrated CSU and DSU connectivity) - N.A. only
- Two serial ports (V.92) - N.A. only
- Two serial ports (V.35 and X.21) - EMEA only
- ISDN via MMAs
- Up to six ISDN B-channels (PRI or BRI) (optional)
- Dial on Demand, Persistent, or WAN Backup
- MLPPP

Expansion Cabinet (BCM400 and BCM1000 only)
Connections
- Six MediaBay Module slots
- An 8-pin modular OK256 connector for the interface to the Business Communications Manager base unit (5-meter cable)

Standard Expansion Cabinet
- Depth: 18.1 in., 45.5 cm
- Width: 13.5 in., 34.3 cm
- Height: 5.4 in., 13.6 cm
- Expansion Cabinet with no MediaBay Modules: 24.75 lb., 11.65 kg
- Expansion Cabinet with six MediaBay Modules: 30 lb., 13.6 kg

Redundant Expansion Cabinet (redundant power supply and fans)
- Depth: 20 in., 50.8 cm
- Width: 13.5 in., 34.3 cm
- Height: 5.4 in., 13.6 cm
- Expansion Cabinet with no MediaBay Modules: 21 lb., 9.5 kg
- Expansion Cabinet with six MediaBay Modules: 26.2 lb., 12 kg

Power requirements
Standard Power Supply | Redundant Power Supply
---|---
Auto-sensing | Auto-sensing
300 Watts | 300 Watts
90/264 VAC | 90/264 VAC
6.0 A/9.0 A | 7.0 A/9.5 A
60/50 Hz | 60/50 Hz

Environmental ranges
Operating temperature: 32°F to 104°F (0°C to 40°C)
Operating humidity: 10% to 90% relative humidity, non-condensing
Storage temperature: -67°F to 158°F (-55°C to 70°C)
Storage humidity: up to 95% relative humidity

Mounting options
- Rack-mount (standard 19-inch rack), stand alone (unit included), wall mount (optional wall mount bracket available separately)

Telephones and Adapters
Station Sets

<table>
<thead>
<tr>
<th>Business Series</th>
<th>North America</th>
<th>Dimensions-ft</th>
<th>Loop Length (26G)</th>
<th>With SAPS sets</th>
</tr>
</thead>
<tbody>
<tr>
<td>T7100</td>
<td>8.1D x 7W x 1SH</td>
<td>1,000 ft.</td>
<td>2,600 ft.</td>
<td></td>
</tr>
<tr>
<td>T7500</td>
<td>8.1D x 7W x 1SH</td>
<td>1,000 ft.</td>
<td>2,600 ft.</td>
<td></td>
</tr>
<tr>
<td>T7316E</td>
<td>8.1D x 7W x 1SH</td>
<td>1,000 ft.</td>
<td>2,600 ft.</td>
<td></td>
</tr>
<tr>
<td>T74 KIM</td>
<td>7.7D x 6W x 1SH</td>
<td>1,000 ft.</td>
<td>2,600 ft.</td>
<td></td>
</tr>
<tr>
<td>T74 KIM</td>
<td>7.7D x 6W x 1SH</td>
<td>1,000 ft.</td>
<td>2,600 ft.</td>
<td></td>
</tr>
<tr>
<td>N44Q</td>
<td>12.5D x 7W x 1SH</td>
<td>Connects to T7316E</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

IP Stations
- Nortel Networks IP Phone 2004
- Nortel Networks IP Phone 2002
- Nortel Networks IP Phone 2001
- Nortel Networks IP Softphone 2050 (PC or Laptop)

T24 KIM—requires T7316E
- T24 ERM (Enhanced KIM—used as CAP)—max. 12 positions per system, max. 4 ERMAs per position
- T24 OLM (Ordinary KIM—used for answering/EL/BL)—unlimited per system, max. 4 OLMAs without power supply per position, max. 9 OLMAs with power supply per position

Mobility—Symbol VoIP Wireless 802.11B
- N4Vision Phone (Wireless VoIP)
- DataVision Phones (Wireless VoIP with Integrated Fax Data Scanner)

Mobility—Cordless
- T46: 3 handset per base station, 2 basestations per system

Accessories
- B17, Desktop
- Nortel Audio Conferencing Unit (NACU)
- Station Auxiliary Power Supply (SAPS)
- ATA 2 Analog Terminal Adapter (separate models for NA, Europe and Australia)
System and component reliability indicators

The table below provides Mean Time Between Failure (MTBF) figures for selected BCM 3.5 systems and modules or spares. MTBF figures are one measure of the reliability of systems. In addition, estimates of the annualized failure rates for these pieces are provided, which are related to the installed base of systems or components at any point in time and converted to a twelve-month period. These figures allow service plans and sparing strategies to be set.

<table>
<thead>
<tr>
<th>Description</th>
<th>PEC</th>
<th>CPC</th>
<th>MTBF (hours)</th>
<th>MTBF (years)</th>
<th>Failure rate (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>BCM 400 Base System: 0x0 with Motherboard, CPU, RAM, HDD, PS, Cooling Fan</td>
<td>NT7B10AAH</td>
<td>A0610880</td>
<td>60,686</td>
<td>6.9</td>
<td>1.5</td>
</tr>
<tr>
<td></td>
<td>NT7B10AAA</td>
<td>A0610891</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>BCM 400 Base System: 0x0 with Motherboard, CPU, RAM, HDD, PS, Cooling Fan</td>
<td>NT7B10AAJ</td>
<td>A0610869</td>
<td>60,686</td>
<td>6.9</td>
<td>1.5</td>
</tr>
<tr>
<td></td>
<td>NT7B10AAA</td>
<td>A0610883</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>BCM Expansion Cabinet: 0x0 with HUB, PS, Cooling Fan</td>
<td>NT7B14AAA</td>
<td>A0691975</td>
<td>86,334</td>
<td>9.9</td>
<td>1.5</td>
</tr>
<tr>
<td></td>
<td>NT7B14AAG</td>
<td>A0691987</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>BCM 400 Base System: 0x0 with Motherboard, CPU, RAM, Redin HDD, Redin PS, Redin Cooling Fan</td>
<td>NT7B10AAB</td>
<td>A0610891</td>
<td>67,951</td>
<td>7.8</td>
<td>1.5</td>
</tr>
<tr>
<td></td>
<td>NT7B10AAG</td>
<td>A0611057</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>BCM Expansion Cabinet: 0x0 with HUB, Redin PS, Redin Cooling Fan</td>
<td>NT7B14AAA</td>
<td>A0699916</td>
<td>96,454</td>
<td>11.0</td>
<td>1.5</td>
</tr>
<tr>
<td></td>
<td>NT7B14AAGH</td>
<td>A0699918</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>BCM 400 MSC Media Services Card</td>
<td>NTAB8829</td>
<td>A0636061</td>
<td>541,739</td>
<td>621</td>
<td>1.0</td>
</tr>
<tr>
<td>BCM 200 MSC Media Services Card</td>
<td>NTAB8930</td>
<td>A0635562</td>
<td>541,739</td>
<td>621</td>
<td>1.0</td>
</tr>
<tr>
<td>BCM-GATM4 – Global Analog Trunk Module</td>
<td>NSB44BAA</td>
<td>A0697997</td>
<td>302,094</td>
<td>35.0</td>
<td>1.0</td>
</tr>
<tr>
<td>BCM-GATM8 – Global Analog Trunk Module</td>
<td>NSB44AAA</td>
<td>A0697994</td>
<td>248,869</td>
<td>28.4</td>
<td>1.0</td>
</tr>
<tr>
<td>BCM-COMBO 4x16 – CLD Trunk &amp; Digital Station Media Bay Module</td>
<td>NSB422AAA</td>
<td>A0786241</td>
<td>128,324</td>
<td>14.6</td>
<td>1.0</td>
</tr>
<tr>
<td>BCM-BRI 5T – Media Bay Module</td>
<td>NT7B64AAB</td>
<td>A0603598</td>
<td>625,938</td>
<td>71.1</td>
<td>1.0</td>
</tr>
<tr>
<td>BCM-DM – Digital Trunk Media Bay Module</td>
<td>NT5B04AAD</td>
<td>A0603597</td>
<td>786,275</td>
<td>90.0</td>
<td>1.0</td>
</tr>
<tr>
<td>BCM-DMS8+ – 16 Digital Station Media Bay Module</td>
<td>NT7B08AAL</td>
<td>A06802619</td>
<td>855,685</td>
<td>98.0</td>
<td>1.0</td>
</tr>
</tbody>
</table>
MSC IP call processing hardware

If your system requires a high volume of IP telephones and/or more IP trunks than the standard eight trunks, you have the option to switch a DS30 bus setting on the MSC from providing service for a media bay module, to providing digital processing service for additional IP telephones and/or trunks. To ensure adequate data flow from the system, you can increase the number of PEC III cards (BCM200 has a maximum of 2 cards; BCM400 has a maximum of 4).

- DS30 channels are internal communication paths. Each DS30 bus provides a possible 32 Signaling channels and 32 media channels.
- Two DS30 buses are exclusively dedicated to MSC data resources. Five paths within these Channels have hard-coded applications. The other paths can be assigned to various data Applications such as voice mail, dialup ISDN WAN, VoIP trunks, or IP telephony.
- Five DS30 channels are exclusively reserved for the media bay modules
- The sixth DS30 bus can be switched to accommodate media bay modules or more Channels for IP telephones or VoIP trunks. You control the use of the channel by your Choice of using either a 2/6 or 3/5 DS30 bus split.

PEC III — The Business Communications Manager 3.5 uses PEC III to deliver increased Capacity for digital signal processing for voice mail, call center, FAX, VoIP trunks, IP
Telephony and dialup ISDN WAN. The BCM200 platform uses one PEC III card (expandable to 2). The BCM400 platform uses two PEC IIIIs (expandable to 4) to accommodate increased Requirements for media processing.

**Media bay modules (MBMs)**
The MBMs connect with external devices to implement various types of voice trunks and stations. Install the MBMs in the media bay module bays in the BCM200, BCM400 base platforms and the expansion unit. There are three types of media bay modules.

<table>
<thead>
<tr>
<th>Module type</th>
<th>Media bay module name</th>
<th>Faceplate acronym</th>
</tr>
</thead>
<tbody>
<tr>
<td>Trunk media bay module</td>
<td>Digital trunk interface</td>
<td>DTM</td>
</tr>
<tr>
<td></td>
<td>Caller ID trunk (CLID) 4 line trunk</td>
<td>CTM 4</td>
</tr>
<tr>
<td></td>
<td>Caller ID trunk (CLID) 8 line trunk</td>
<td>CTM 8</td>
</tr>
<tr>
<td></td>
<td>ISDN BRI S/T Interface</td>
<td>ISDN BRI</td>
</tr>
<tr>
<td></td>
<td>Global analog trunk module</td>
<td>GATM8</td>
</tr>
<tr>
<td>Station media bay module</td>
<td>16 digital station interface (DSI) double</td>
<td>DSM 16+</td>
</tr>
<tr>
<td></td>
<td>density</td>
<td>DSM 32+</td>
</tr>
<tr>
<td></td>
<td>Combination CTM4 x DSM16</td>
<td>4X16</td>
</tr>
<tr>
<td></td>
<td>Analog Station Interface</td>
<td>ASM 8</td>
</tr>
<tr>
<td>Specialized media bay module</td>
<td>DECT Base Station Module</td>
<td>DECT8</td>
</tr>
<tr>
<td></td>
<td>DECT Base Station Module (u-law)</td>
<td>DECT8</td>
</tr>
<tr>
<td></td>
<td>Fibre Expansion Module</td>
<td>FEM 6</td>
</tr>
<tr>
<td></td>
<td>Digital Drop &amp; Insert MUX</td>
<td>DDI Mux</td>
</tr>
</tbody>
</table>

The BCM200 holds a maximum of **two media** bay modules. The BCM400 holds a maximum of **four media** bay modules. The expansion unit holds a maximum of six media bay modules.

The BUS is a collection of communication lines that carry electronic signals between components in the system. Besides internal communications, the Business Communications Manager MSC uses buses to support media bay modules and IP telephony components.

**DS30 numbers**
A DS30 bus is a block of virtual pathways on the media services card (MSC). On a default system, six buses of DS30 channel blocks can be assigned to media bay modules. Which block the module is assigned to determines the range of line (trunk) numbers or extension Numbers (DNs) that can be allocated by the module to the equipment connected to that module.
The other two blocks are permanently routed to the PEC digital signal processors (DSPs) to Support internal Business Communications Manager Functions such as voice mail, VoIP trunks, IVR, and IP telephony functions. This configuration is called a **2/6 channel split**. You can change the DS30 allocation to a **3/5 split** to accommodate increased IP telephony or VoIP Trunk requirements. You do this by assigning bus 7 to the voice data sector. This choice should be made at system startup, but a default system can be changed through the Unified Manager to a 3/5 split after startup if IP requirements increase.

**DS30 Split**

A DS30 bus is a group of **32** signaling channels and **32** media channels. The DS30 split determines how these channels are assigned on Business Communications Manager. You have a choice of a 2/6 or a 3/5 split. If you choose a 2/6 split, two DS30 buses are assigned to the MSC and six are assigned to the Media Bay Modules. If you choose a 3/5 split, three DS30 Buses are assigned to the MSC and five are assigned to the Media Bay Modules. The split you choose is determined by the number of signaling channels you require for Applications such as voice mail, IVR, IP trunks, IP telephones and dialup ISDN WAN connections. If you need 58 signaling channels or less for these applications, use a 2/6 DS30 split. If you need 59 signaling channels or more, use a 3/5 DS30 split.

**Warning:** If you change the channel split from 3/5 to 2/6 after your system is configured, You will lose all the data and optional application connections.

## Setting offsets

Each offset is one-quarter of a DS30 bus. Each bus supports 16 lines (32 time slots) for most Modules.
Set Media Bay Module Dip Switches

There are six DIP switches that you use to set the DS30 channels and offsets are found either on the bottom of the module or on the rear, behind the power connector.

Assign Dip switches settings before you install a media bay module. The settings determine which line numbers (trunks) or DNs (extensions) the equipment connected to the module will have access to. The DIP switches are located on the back or underside of the media bay module.

Switches on the media bay module

FEM switch settings

The DIP switches on the underside of the FEM module are used to turn the six ports on the front of the module on or off. You need to turn a port on for each Norstar expansion module you want to connect to the Business Communications Manager. Each port also occupies one full DS30 Channel. Therefore, if you have a fully-configured, six-module Norstar system to convert, you need to turn on all six ports on the FEM, and, therefore, no other module can be installed in the Business Communications Manager.
**Double Density**

The BCM 3.5 software configures the MSC so that it supports 32 ports for digital Telephones on DS30 buses 02 to 05.

The 32 time slots are important when you are working with station media bay modules. The DSM 16+, DSM 32+, and ASM 8 modules can be configured, using the offset dip Switches, to use each of these time slots as separate telephone lines. This, essentially, Doubles your system telephone capacity. On a default 3.5 system, this feature, called Partial double density (PDD), is available on DS30 2, 3, 4, and 5. DS30 6 and 7 maintain The current two time slots per line configuration, which supports the Companion Application. For systems where Companion is not required, you can use the Unified Manager to change all six DS30 buses to full double-density (FDD).

Exception: If your system has a 3/5 channel split, only DS30 6 becomes double density And is available to media bay modules when the system is changed to FDD. However, 16 More channels on DS30 7 are also made available for IP telephones.

*You cannot regress from a Full Double Density (FDD) system to a Partial Double Density (PDD) system.*

---

**Rules for Assigning DS30 Resources**

Media bay modules are assigned to DS30 buses in a specific hierarchical manner. These are Some of the preferred order of positioning for each type of module

*The following are some general notes about assigning modules:*

• The DIP switches on the DDIM module are used to set the DS30 designation for the DTM part Of the module. The module automatically assigns an additional DS30 for the data module part Of the DDIM. You cannot choose DS30 7 for the DDIM module, because the data module Would not be accessible. The same applies to DS30 6 if your system is set to a 3/5 split.

• If you chose a 3/5 channel split for your system, DS30 7 cannot be used by any module. For Modules that require two buses, this means that you cannot set the DIP switches to DS30 6 for Those modules, because the second level of lines would fall into DS30 7, which would not be Accessible.

DSM 32 modules require two DS30 numbers. When you assign the first DS30 number to a DSM 32, the module automatically adds the next DS30 number. For example, if you assign DS30 2 to a DSM 32, it uses DS30 2 and 3. However, you cannot choose DS30 7 for the DSM32 module, because the second level of DSM lines would not be accessible. The same Applies to DS30 6 if your system is set to a 3/5 split.

DSM 32+ modules can be set to either **single or double density**. When they are set to double Density, the module only requires one DS30 bus.

• The DIP switches on the 4X16 module are used to set the DS30 designation and offset for the CTM part of the module. The module automatically assigns an additional DS30 for the 16 DSM lines. However, you cannot choose DS30 7 for the 4X16 module, because the DSM lines Would not be accessible. The same applies to DS30 6 if your system is set to a 3/5 split.

**Companion** sets configure the DSM or DSMs handling Business Communications Manager
Companion to DS30 6 or 7. You must change the module number of any trunk media Bay modules configured to module 6 or 7 to an unassigned module number to prevent conflicts With Companion.

**Companion DS30 split restrictions:**
If you choose a 3/5 channel split for your system, the second module cannot be assigned. Therefore, you can add a maximum of 16 Companion base stations, which support a Maximum of 30 handsets. This means you can only use a DSM 16 on DS30 6. You cannot Assign a DSM 32.

**If your system is set to full double density (FDD), DS30 6 and/or 7 do not support Companion.**

- The **CTM8** module, when set to single density, uses two offsets on a DS30 bus. You assign the First offset to the module, and the second offset is automatically selected. This means that you Can choose offset pairs 0-1, 1-2, or 2-3. Because the module requires two offsets on the same DS30, you cannot select offset 3.

- The **CTM8** module, when set to single density, uses two offsets on a DS30 bus. You assign the First offset to the module, and the second offset is automatically selected. This means that you Can choose offset pairs 0-1, 1-2, or 2-3. Because the module requires two offsets on the same DS30, you cannot select offset 3.

- ASM8 module in a single density configuration, such as for DS30 6 or 7 when they are set to the default PDD, only offset 1 and 2 are available to ASM8s. In a double-density configuration, you can install four ASM8s per DS30 bus.

- DSM 16s have one connector which connects to 16 lines (telephones). These modules require A full DS30 number each (single density), or half a bus (double density).

- DSM32s have two connectors, each of which connects to 16 lines (telephones). These Modules require two full, consecutive DS30 numbers (single density) or one full but (double density).

The DECT module supports a maximum of eight DECT radio base station connections. This Module occupies one full DS30 bus, which can support a maximum of 32 cordless handsets when All eight base stations are deployed. The DECT module should be installed on DS30 6 or 7.

**DSM modules deployed with 2.5 systems are all single density and cannot be set to Double density. The DSM 16+ and DSM32+ modules can be set to either density.**

Choose the assigned order for modules

Station modules are assigned starting with **DS30 2**. This allows telephones to start numbering from the system Start DN (Default: 221). The exception to this is a DSM used for Companion, which must be installed on **DS30 6 (DSM32)** or **DS30 6 and 7 (two DSM 16s)**. If your system is set to a 3/5 split, you can only assign a DSM16 to DS30 6 for Companion.
Trunk modules are assigned starting at **DS30 7**, in a system with a 2/6 DS30 split, and at **DS30 6** in a system with a 3/5 split. The exception to this is the 4X16 module and the DDIM, which require two DS30 buses, so it must be set to a DS30 that has the next channel open.

The Following are some examples of single and double density configuration

**3/5 channel split**
- DS30 7 is not available to any module if your system has been configured with a 3/5 channel split.
- Modules that require two DS30 buses, such as the DSM32 and the 4X16, must be assigned to DS30s higher than 6, to allow accommodation of all the resource requirements.
- Companion: Only a maximum of 16 Companion base stations (30 handsets) can be registered on a system configured with a 3/5 split.

**Combination and specialized media bay modules**
- **4X16 module**: 2 DS30 buses; offset set to 0, 1, 2, or 3.
  - 1 offset of 1 full DS30 for lines
  - 1 full DS30 for telephone and equipment connections
- **DDIM module**: 2 DS30 buses; offset set to 0.
  - 1 full DS30 for DTM module
- **DECT module**: 1 DS30 bus; offset set to 0.
  - 1 DECT per DS30
  - (1 DECT per system)

**Space requirements for special media bay modules**
Space requirements for media bay modules, on a per-DS30 configuration
Assigning single-density modules to the DS30 channel hierarchy

After you choose your modules, choose where to assign them on the DS30 buses.

Station modules are assigned starting at the top (DS30 2) of the available media bay module DS30 buses.

Exception: DSM 16 or DSM 32 used for Companion. In this case, DSM 32 must be set to DS30 3. DSM 16 can be installed in both DS30 2, 3, and 4. If DS30 3 is not available you cannot use a DSM 32.

Trunk modules are assigned starting with the last available media bay module. DS30 bus (DS30 2 or 3, depending on the channel split is effective).

Exception: a 4X16 module or a DDM cannot be assigned to the last DS30 bus.

Lines start at 61 on DS30 7.
Assigning double density modules to the DS30 channel hierarchy

Double-density example
(system configured as Partial Density)

Partial density
Systems configured with Partial double density (PDD), allow Companion telephones on DS30 6 and 7 (if the system is set to a 2/6 split). In this configuration, DS30 6 and 7 only allow single-density modules. DS30 2 to 5 are set to allow double density modules.

Double density
Systems configured with Full double density (FDD), do not allow Companion telephones. All DS30s are set to allow double density modules.

3/5 channel split
Works in same as shown in the single-density diagram. If the system is set to a 3/5 split, DS30 7 is not available to any media bay modules.
<table>
<thead>
<tr>
<th>Programmed Bus Type</th>
<th>Hardware unit</th>
<th>Capacity</th>
<th>Available line types (some line types are region-dependent)</th>
</tr>
</thead>
</table>
| Station module      | Digital Station Media Bay Module (DSM 16/16+ or DSM 32/22+) | Single density  
- DSM16/16+ = 1 per bus/16 digital sets per module  
- DSM 32/22+ = 2 buses/32 digital sets per module  
- Double density  
- DSM16+ = 2 per bus/16 digital sets per module  
- DSM 32+ = 1 per bus/32 digital sets per module  | N/A |
|                     | 4X16 Media Bay Module (4X16)  
(Counts as one DSM 16) |  
- 4X16 = 1 offset (frunk) and additional bus/16 digital sets | N/A |
|                     | Nonstar station module (SM) connected to a FEM |  
- SM = 1 bus/16 digital sets | |
| Analog station module | Analog Station Media Bay Module (ASM 6) | Single density  
- ASMB = 2 per bus/8 digital sets for each module  
- ASMB = 4 per bus/8 digital sets for each module  | N/A |
|                     | Nonstar analog station module connected to a FEM |  
- FEM = 1 per bus/16 digital sets | |

<table>
<thead>
<tr>
<th>Programmed Bus Type</th>
<th>Hardware unit</th>
<th>Capacity</th>
<th>Available line types (some line types are region-dependent)</th>
</tr>
</thead>
</table>
| Station module      | Digital Station Media Bay Module (DSM 16/16+ or DSM 32/22+) | Single density  
- DSM16/16+ = 1 per bus/16 digital sets per module  
- DSM 32/22+ = 2 buses/32 digital sets per module  
- Double density  
- DSM16+ = 2 per bus/16 digital sets per module  
- DSM 32+ = 1 per bus/32 digital sets per module  | N/A |
|                     | 4X16 Media Bay Module (4X16)  
(Counts as one DSM 16) |  
- 4X16 = 1 offset (frunk) and additional bus/16 digital sets | N/A |
|                     | Nonstar station module (SM) connected to a FEM |  
- SM = 1 bus/16 digital sets | |
| Analog station module | Analog Station Media Bay Module (ASM 6) | Single density  
- ASMB = 2 per bus/8 digital sets for each module  
- ASMB = 4 per bus/8 digital sets for each module  | N/A |
|                     | Nonstar analog station module connected to a FEM |  
- FEM = 1 per bus/16 digital sets | |
Types of MSC resources

Media Services Card (MSC) resources are required for the following features:

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

When you configure the MSC resources, you are configuring how Business Communications Manager shares the MSC resources between these features.

There are several resources that you must check when you are configuring the MSC resources:

- Signaling channels
- Media channels
- DSP resources
- Voice bus paths
- Media gateways
**Signaling channels**

Signaling channels are the communication channels used to send control signals to and from the MSC. You must have one signaling channel for each device you have connected and feature port you have enabled. Signaling channels are the MSC resource that determines how many IP telephones you can connect to your system. If you have a system that does not use IP telephones, the number of signaling channels does not affect your configuration.

- The total number of signaling channels available to the MSC depends on the DS30 split you have configured.

If you have a 2/6 DS30 split, the total number of signaling channels is 64. **Actual is 58**
If you have a 3/5 DS30 split, the total number of signaling channels is 96. **Actual is 90**

**Here are some examples to show you the numbers required by some applications:**

Dial-on-Demand ISDN WAN uses 27 signaling channels.
All 27 signaling channels are used, regardless of the number of WAN channels configured.
- Voicemail requires one signaling channel for each voicemail port enabled. You can enable up to 32 voicemail ports.
  Both voicemail and call center use Voicemail ports.
- IP Telephony clients require one signaling channel for each IP telephone connected to the system.
- IP Telephony trunks require one signaling channel.
  Only one signaling channel is required regardless of the number of IP Telephony trunks enabled.
- IVR requires 1 signaling channel for each IVR port enable.
- Up to 24 ports enabled. Maximum of 32 ports between IVR and voicemail.

**Media channels**

Media channels are the communication channels used to send voice and data information between the devices and feature ports. Media channels are required only when a device or feature is sending or receiving voice or data information. For this reason, the devices and feature ports can share media channels. Management functions use five media channels. These five channels are reserved for management functions and are always in use.

If you have a 2/6 DS30 split, the total number of media channels is **58**
If you have a 3/5 DS30 split, the total number of media channels is **90**

**Here are some examples to show you the numbers required by some applications:**

Dial-on-Demand ISDN WAN uses 27 media channels.
All 27 media channels are used, regardless of the number of WAN channels configured. The Maximum number of WAN channels is 16.
Voicemail and call center use one media channel for each active session.
- DECT mobility requires one media channel.
  **Note:** If your system also has Dial-on-Demand WAN, DECT uses one of the 27 WAN media Channels, so an additional channel is not required.
- A call between an IP telephone and a digital or analog telephone or a PSTN line uses a media Channel for the duration of the call.
• A call from a digital or analog telephone that uses an IP trunk uses a media channel for the
  Duration of the call.
• A call between two IP telephones on the same Business Communications Manager uses a
  Media channel during call setup. After the call is established, the media channel is released.
• A call on an IP telephone using an IP trunk uses a media channel during call setup. After the
  Call is established, the media channel is released.
• IVR needs 1 media channel for each active session.

DSP resources
Digital Signal Processors (DSP) provide the voice processing functions on Business
Communications Manager. Voice processing is required to convert voice information to and from
Digital format for voicemail, call center and IVR. Voice processing is also required to handle
Encoding and decoding of IP telephony calls. The DSPs are located on the MS-PEC cards
installed in your MSC.

The number of DSP resources you have depends of the number of type of MS-PEC you have
Installed.
For the purposes of calculating DSP resources, we can estimate the relative power of each
Configuration as follows:
• 4 MS-PEC I 24 units
• 2 MS-PEC III 64 units
• 4 MS-PEC III128 units

Here are some examples to show you the numbers required by some applications:

The number of DSP resources you need depends on the features and type of codec you are using.
• Dial-on-Demand WAN uses 1 unit for each 64Kbit/s channel
• Voicemail, IVR, and call center use 1 unit for each active session
• Fax uses 6 units for each active fax channel
• IP telephone or IP trunk using G.711 codec uses 1 unit
• IP telephone or IP trunk using G.729 codec uses 3 units
• IP telephone or IP trunk using G.723 codec uses 4 units

Voice bus paths
The voice bus paths are the communication channels between the DSPs on the MS-PECs and the
Master DSP on the MSC. One voice bus path is required for each voice-processing task that is
Operating on the DSPs.
There are 62 voice bus paths available on Business Communications Manager

Here are some examples to show you the numbers required by some applications:
• Voicemail and IVR use one voice bus path for each active session.
• Dial-on-Demand WAN uses one voice bus path for each 64Kbit/s channel that is active.
• IP telephones and IP trunks require one voice bus path when ever a media channel is required.

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

Here are some examples to show you the numbers required by some applications:
One media gateway is required for each call:
• From an IP telephone to an analog or digital telephone
• From an IP telephone using a PSTN line
• From an analog or digital telephone using an IP trunk
Telephones and adapters

The following telephones and devices can be used with the Business Communications Manager system.

**Business Series Terminal T7100**
- one-line display, one memory button without indicator.
- T7000 (not shown) (International only) - four memory buttons, without display or indicators.

**Business Series Terminal T7316**
- two-line display, three display buttons, 16 memory buttons with indicators, eight memory buttons without indicators. Supports separate route key and a headset key under the dial pad.

**Business Series Terminal T7316E**
- two-line display, three display buttons, 16 memory buttons with indicators, eight memory buttons without indicators. Hands-free, mute, and headset buttons are located under the dial pad. The default button assignment is the same as the T7316 when the T7316E is installed on a system running software previous to BCM 3.5 software. The default button assignment is different than the T7316 when the T7316E is installed on a system running BCM 3.5 or later software.

**Business Series Terminal T7208**
- one-line display, eight memory buttons with indicators

**Business Series Terminal (BST) Doorphone**
The BST Doorphone is used as an intercom device to control access to your building. It provides call notification and hands-free communication from a site entry location to assigned telephones on the Business Communications Manager system.

**Business Series Terminal T7316E + Key Indicator Module (KIM)** — all the features of the T7316E plus 24 extra memory buttons with icons, per KIM. Can be configured as an enhanced Central Answering Position (CAP) that supports line and hunt group appearances the eKIMs, as an ordinary CAP that only supports memory button programming on the OKIMs. Supports a maximum of four eKIMs and up to nine OKIMs.
BCM IP-enabled platform

Supports converged voice networks. IP telephony adds additional flexibility and capabilities to your network as the network expands and specific challenges arise, IP telephony provides an additional level of flexibility for solving specific business challenges that cannot be met by conventional digital telephony. The ideal solution is to mix-and-match digital and IP telephony to create a solution that precisely mirrors the needs of your business. For example, an IP phone set can be placed at an employee’s home office, and with secure IP connectivity, it will function as part of the enterprise network. Need to place a handset at a guard post that’s over a mile away? No problem. Or one of your roving employees might need to connect to the enterprise voice network from various campuses, or even over a customer’s network. Imagine the flexibility of connecting via a wireless IP hotspot at any remote location, even an airport, and receiving

**i2004 IP telephone** — connects through an internet link to the Business Communications Manager. 6-line text display with a row of display keys on the 8th display line. Six memory keys with indicators. It can be used to call through VoIP or PBX lines. Not shown: i2002 IP telephone and i2050 Software Phone.

**Mobility options**

**Companion** (region-specific) — provides twinning capability between a stationary set and a wireless mobile set. These handsets communicate through a stationary base station, which is wired to a digital station media bay module on the Business Communications Manager. Depending on your system configuration, you can have up to 64 sets assigned to your system. For installation instructions, refer to “Companion Hardware Installation” on page 273.

Provides two-line display, but no line, memory or display buttons. The handset accesses a restricted set of system features.

**DECT** (region-specific) — provides cordless access to the system through a DECT media bay module. The cordless handsets can be twinned with a stationary set, or configured to act as an independent set. You can register up to 32 sets on a module. Each Business Communications Manager system can support one DECT module.

Has display, but no line, display or memory buttons and has access to a restricted list of system features. For base station installation and handset registration instructions, refer to the **DECT Installation and Maintenance Guide**.

**T7406 Cordless Telephone system** — provides cordless mobility in a small office environment. Each base station supports three telephones. Function is based on the T7316/M7310 telephone. The base station connects to a digital station media bay module on the system. Provides six memory buttons with indicators and a two-line display with three display buttons.

For installation instructions, refer to the **T7406 Cordless Telephone Installation Guide**.

**Symbol® NetVision and NetVision Data telephones** — H.323+ based IP telephones provide eMobility access through a LAN/WAN connection via a wireless access point. A display menu provides access to user and call feature.

Provides multi-line display capability, but no line, memory or display buttons.

**Audio Conference Unit** (ACU) — provides large-room audio conferencing. The keypad provides many of the set features of the basic Norstar M-series telephones without display or memory buttons. This set comes with three microphones. Installation instructions are provided by the vendor.
your calls as if you were seated at your desk at the central site IP telephony consists of two different types of interfaces:
1) IP stations, which function as normal phone sets or wireless phones, but are connected directly to an IP interface such as a BayStack 460 10/100 switch.
2) IP trunks, which allow the system to connect to other devices, including other Business Communications Manager, IP-enabled Meridian 1 PBX, or Succession 1000 IP-PBX systems. IP trunks can be established using either the H.323 protocol or SIP (BCM to BCM only). The Nortel Networks IP telephone portfolio currently includes:
• i2001 internet telephone, IP-based phone set.
• i2002 Internet Telephone, which is a small, economical IP-based phone set
• i2004 Internet Telephone provides a full-featured IP-based phone set with a large LCD display
• i2050 IP softphone client, a software based PC client that brings full featured IP telephony to your desktop or laptop PC
• Support for a wide variety of third-party wireless IP-based handsets by combining wireless IP-based handsets with wireless IP, customers can now take advantage of wireless voice over IP. This lets customers use the same WiFi 802.11 infrastructure that they already have to support voice mobility, opening the door for wireless phones, wireless barcode scanners, and wireless laptop telephony.
To simplify calling between conventional telephones and IP-based phones, Nortel Networks provides gatekeeper technology to deliver automatic translation between phone numbers and IP addresses. Just key in the phone number of the person you wish to reach, and the call will ring through—even if the person you are calling is connecting over a wireless IP network at a remote location. Gatekeeper databases can either be established locally within each Business Communications Manager, or the system can work with a centralized gatekeeper database maintained within Nortel Networks Succession devices or other third-party products.

**Figure 1: Business Communications Manager single-site solution**

**Integrated data services** designed to meet the needs of your business. Comprehensive data services for small-to-medium-sized sites. In addition to its comprehensive telephony solutions, Business Communications Manager can also provide a rich array of data services capable of meeting the requirements of most small- to medium-sized sites. By leveraging tight, system wide integration of both
telephony and data networking features, Business Communications Manager delivers a level of reliability and ease of management that cannot be provided by solutions that rely on multiple components. The system provides support for the following advanced data networking features:

- **VPNs** (Virtual Private Networks) are secure private networks created over the public Internet. The system’s VPN implementation adheres to Microsoft’s Point-to-Point Tunneling Protocol (PPTP), and supports IPsec and 128-bit triple DES encryption. By using branch-to-branch tunneling, the system can extend VPNs to all devices on the network, and extend a secured tunnel over the public Internet to another Business Communications Manager, Nortel Networks Contivity® Secure IP Services Gateway, Nortel Networks Shasta® 5000 Broadband Service Node, or other IPsec-compliant device.

- **IPsec Support** enables the system to provide secure connectivity to mobile workers with an IPsec client installed on their laptops, or to a home office with a small IPsec-compliant branch device such as a Contivity 1000 Series gateway. This approach supports both encrypted data and encrypted voice, and is capable of satisfying stringent governmental security requirements.

- **Firewall with Stateful Packet Filtering**
  Can be quickly configured to allow or deny network access based on time-of day, application, IP address, port range, or other attributes. Ultra-granular control enables Web or data traffic to be restricted, while still permitting VoIP calls to pass through.

- **Integrated IP Router** built-in to the system can be configured with an optional WAN interface to provide full-service IP routing. Two WAN Media Bay Modules are available: one with two serial interfaces, and one with one serial interface and one T1 interface with integrated CSU.

- **QoS (Quality of Service)** is essential for supporting IP telephony or other latency-sensitive traffic. Even if you are using an external router, the system can recognize and prioritize voice traffic, and in many instances this approach can help your company avoid the cost of replacing your router. QoS prioritization takes place within the Business Communications Manager, and voice traffic is then sent through the second LAN interface to the legacy router.

- **NAT (Network Address Translation)**
  Allows a single public IP address to be shared by multiple internal users. Up to 250 private IP addresses can be issued by the DHCP server, providing additional security because the user’s private IP address is not visible to other sites on the public Internet. NAT can also be used to share static addresses across a group of users, who need exclusive, but temporary, use of a static IP address. Since static IP addresses can be costly, this can translate into hard cash savings or your company.

- **DHCP (Dynamic Host Control Protocol)**
  Automatically issues IP addresses on an as-needed basis. Ideal for sharing IP addresses through NAT, DHCP is also useful for supporting mobile workers who want to use their laptop-enabled i2050 Software Phone, but don’t want to go through the hassle of obtaining and entering a static IP address. With DHCP enabled, a user with a wireless IP telephony handset can simply walk into a wireless IP hot spot, and their phone will be ready to make and receive calls.

- **DNS (Domain Name System)** provides name-to-IP address correlation, allowing users to enter the name of Frequently-visited sites and have the system call up the appropriate Web page. This eliminates the need for the system to perform lookups over the Internet every time a site is accessed, maximizing performance and Preserving available bandwidth.

- **Web Caching Services** also maximizes system performance, and preserves available bandwidth by laminating the need to pull down frequently-visited Web pages, such as the corporate home page. Cached Web pages are stored locally on the system, and then updated automatically.
VoIP Gateway

The VoIP gateway allows communication with other supported H.323 v2 gateways via system to system (trunk) calls. The VoIP gateway is treated as a trunk and uses the trunking and routing functionality of the BCM product portfolio. The IP trunks are an integral part of the telephony services, making Voice over IP transparent to users. The gateway provides dialing plan support, allowing customers to set up the routing tables to direct calls to appropriate destinations based on the dialed digits. In addition, it supports voice calls (but does not support fax or modem calls). Routing codes and the destination code table allow the core BCM telephony services to identify which trunking facilities are used for calls and when they are used. Routing codes are associated with line pools. More than one routing code may be assigned to each destination code, depending on factors such as least-cost routing.

Voice over Internet Protocol (VoIP) technology is a system for transmitting telephone calls over data networks, such as the ones that make up the Internet.

A VoIP gateway involves the conversion of voice from its traditional circuit-switched telephony format into a packet format that can be transported over an IP network. The BCM VoIP gateways are key code-enabled options that support both IP trunks to communicate with other systems, as well as IP stations, such as the i2004, i2002, and i2050 and mobility wireless VoIP solutions. Subscribers can activate a maximum of 60 simultaneous VoIP trunk gateway sessions on the BCM with the purchase of software key codes and two additional PEC III DSP processors. The VoIP trunk gateway key codes are available in increments of 2, 4, 8, 16 and 32. VoIP stations or clients are available in increments of 1, 4, 8, 16, 32 and 64. The BCM can support as many as 90 IP stations, depending on the other capabilities planned to be configured.
**IP clients:** IP Clients are IP telephones like mentioned earlier. The IP station portfolio consists of the i2001, i2004, the i2002 and the i2050 software-based phone, in addition to other H.323 client devices such as the wireless Voice over IP handset. The i2004 and i2002 support paging through the set and hands-free intercom. Since the i2050 is a “software phone,” it has no set or speaker and cannot receive pages or hands-free intercom calls. The i2050, however, can originate paging and hands-free intercom calls.

IP stations provide the same functionality as Nortel Networks traditional digital stations, yet do not require digital station modules. IP stations are connected directly to the customer’s LAN and in some cases WAN equipment.

**DSP** resources are required only when the IP telephone is in use (for example, to make a call, receive a call, listen to voicemail).

**IP trunks:** IP Trunks are communication channels that Business Communications Manager uses to send and receive IP telephony calls using the Public Data Network. You can use IP trunks to connect the Business Communications Manager system to:

- Another Business Communications Manager system
- A Meridian 1 IPT system
- A third-party H.323 end point or gateway

**Media Gateway:** Media Gateways provide the connection between IP telephony devices (IP trunks, i2004 telephones, i2050 telephones, and H.323 terminals) and normal telephony devices (PSTN lines, T7316 telephones, T7208 telephones, T7100 telephones, etc.).

**Things you should know and be aware of:**

**IP Addresses and the Internet**

Because TCP/IP networks are interconnected across the world, every machine on the Internet must have a unique address to make sure that transmitted data reaches the correct destination. Blocks of addresses are assigned to organizations by the Internet Assigned Numbers Authority (IANA). Individual users and small organizations may obtain their addresses either from the IANA or from an Internet service provider (ISP). The Internet Protocol (IP) uses a 32-bit address structure. The address is usually written in dot notation (also called dotted-decimal notation), in which each group of eight bits is written in decimal form, separated by decimal points. For example, the following binary address

```
11000011 00100010 00001100 00000111
```

Is normally written as: 195.34.12.7

The latter version is easier to remember and easier to enter into your computer.

In addition, the 32 bits of the address are subdivided into two parts. The first part of the address identifies the network, and the second part identifies the host node or station on the network. The dividing point may vary depending on the address range and the application.

**Private IP Addresses**

If you’re local network is isolated from the Internet (for example, when using NAT), you can assign any IP addresses to the hosts without problems. However, the IANA has reserved the following three blocks of IP addresses specifically for private networks:

- 10.0.0.0 - 10.255.255.255
- 172.16.0.0 - 172.31.255.255
- 192.168.0.0 - 192.168.255.255
**Single IP Address Operation Using NAT**

In the past, if multiple PCs on a LAN needed to access the Internet simultaneously, you had to obtain a range of IP addresses from the ISP. This type of Internet account is more costly than a Single-address account typically used by a single user with a modem, rather than a router. If you utilize Firewall/VPN Router that employs an address-sharing method called Network Address Translation (NAT). This method allows several networked PCs to share an Internet account using only a single IP address, which may be statically or dynamically assigned by your ISP. The router accomplishes this address sharing by translating the internal LAN IP addresses to a single address that is globally unique on the Internet. The internal LAN IP addresses can be either Private addresses or registered addresses.

![Diagram of IP address translation](image)

**What is a VPN?**

A VPN can be thought of as a secure tunnel passing through the Internet, connecting two devices such as a PC or router, which form the two tunnel endpoints. At one endpoint, data is encapsulated and encrypted, then transmitted through the Internet. At the far endpoint, the data is received, unencapsulated and decrypted. Although the data may pass through several Internet routers between the endpoints, the encapsulation and encryption forms a virtual “tunnel” for the data.
The tunnel endpoint device, which encodes or decodes the data, can either be a PC running VPN Client software or a VPN-enabled router or server. Several software standards exist for VPN data Encapsulation and encryption, such as PPTP and IPSec. To set up a VPN connection, you must configure each endpoint with specific identification and connection information describing the other endpoint. This set of configuration information defines a security association (SA) between the two points. Two common applications of VPN are:

- secure access from a remote PC, such as a telecommuter connecting to an office network
- secure access between two networks, such as a branch office and a main office

**Accessing Network Resources from a VPN Client PC**

VPN client remote access allows a remote PC to connect to your network from any location on the Internet. In this case, the remote PC is one tunnel endpoint, running VPN client software.

In some cases, the client PC may connect to the Internet through a local non-VPN-enabled router, as shown below:

If the non-VPN router is performing NAT, it must support “VPN-pass through” of IPSec-encoded Data.

If the non-VPN router is performing NAT, it must support “VPN-pass through” of IPSec-encoded Data. For a PC to act as a tunnel endpoint to your Firewall/VPN Router, the PC must run a VPN Client program based on the IPSec protocol.

**What Is IPSec?**
IPSec—or Internet Protocol Security—is a suite of protocols that provides security for IP traffic at the network layer. It defines how to provide data integrity, authenticity and confidentiality across a public network like the Internet. It accomplishes these goals through tunneling, encryption and authentication, but allows enterprises to select the specific security policy appropriate for their business. Configuration choices include:

- **Tunneling.** Authentication Header (AH) or Encapsulating Security Payload (ESP)
- **Encryption.** 56-bit DES, 112- or 168-bit 3DES, 128-, 192- or 256-bit AES, or none
- **Authentication.** Username and password (such as RADIUS), username and token + pin (such as RSA SecurID), or X.509 digital certificates (such as Entrust or VeriSign). But with flexibility also comes complexity. For two entities to communicate via an IPSec connection, both must agree to the same security policy, called a security association, which must be configured in the devices on both ends of the IPSec connection. A single IPSec tunnel secures all communications between the devices, regardless of traffic type (TCP, UDP, SNMP) or application (e-mail, client-server, database). Tunnels can be established from server-to-server and user-to-server. An IPSec server can secure traffic for many devices and is referred to as a gateway; an IPSec user (an individual device) is referred to as a host. Because IPSec operates at the network layer, users gain access to all company resources as if they were physically in the office connected to the corporate LAN. Special-purpose software is available to create IPSec connections. This software is typically available for user workstations, PCs and mobile devices, as well as edge routers and firewalls. Some vendors offer special purpose VPN appliances with the IPSec software integrated.

**What is a Router?**
A router is a device that forwards traffic between networks based on network layer information in the data and on routing tables maintained by the router. In these routing tables, a router builds up a logical picture of the overall network by gathering and exchanging information with other routers in the network. Using this information, the router chooses the best path for forwarding network traffic.

**IP Configuration by DHCP**
When an IP-based local area network is installed, each PC must be configured with an IP address. If the PCs need to access the Internet, they should also be configured with a gateway address and one or more DNS server addresses. As an alternative to manual configuration, there is a method by which each PC on the network can automatically obtain this configuration information. A device on the network may act as a Dynamic Host Configuration Protocol (DHCP) server. The DHCP server stores a list or pool of IP addresses, along with other information (such as gateway and DNS addresses) that it may assign to the other devices on the network. Some Firewall/VPN Router has the capacity to act as a DHCP server. Firewall/VPN Router sometimes also functions as a DHCP client when connecting to the ISP. The router can automatically obtain an IP address, subnet mask, DNS server addresses, and a gateway address if the ISP provides this information by DHCP.

**What is a Firewall?**
A firewall is a device that protects one network from another, while allowing communication between the two. A firewall incorporates the functions of the NAT router, while adding features for dealing with a hacker intrusion or attack. Several known types of intrusion or attack can be recognized when they occur. When an incident is detected, the firewall can log details of the attempt, and can optionally send email to an administrator notifying them of the incident. Using information from the log, the administrator can take action with the ISP of the hacker. In some Types of intrusions, the firewall can fend off the hacker by discarding all further packets from the Hacker’s IP address for a period of time.
Router? Bridge? Switch? Hub? What's the difference?

A hub is a **repeater**, which is an **OSI model** device, the simplest possible. A hub takes the data that comes into a port and sends it out all the other ports in the hub. It doesn’t perform any filtering or redirection of data. Although it’s actually a little more complicated, you can think of a hub like a piece of wire. A better analogy might be that of an Internet Chat room. Everything that everyone who joins a particular chat is seen by everyone else. If there are too many people trying to chat, things get bogged down.

**Bridge**
Bridges (sometimes called "Transparent bridges") work at OSI model **Layer 2**. This means they don’t know anything about protocols, but just forward data depending on the destination address in the data **packet**. This address is not the IP address, but the MAC (Media Access Control) address that is unique to each network adapter card.

With a Bridge, all your computers are in the same network **subnet**, so you don’t have to worry about not being able to communicate between computers or share an Internet connection. DHCP servers will work fine across Bridges, or if you assign your own IP addresses, you’ll use the same first 3 "octets" of the IP address (Example: 192.168.0.X).

However, the only data that is allowed to cross the bridge is data that is being sent to a **valid address on the other side of the bridge**. No valid address, no data across the bridge. Bridges don’t require programming. They learn the addresses of the computers connected to them by listening to the data flowing through them.

Bridges are very useful for joining networks made of different media types together into larger networks, and keeping network segments free of data that doesn't belong in a particular segment.

**Switches**
Switches are the same thing as Bridges, but usually have multiple ports with the same "flavor" connection (Example: 10/100BaseT).

Switches can be used in heavily loaded networks to isolate data flow and improve performance. In a switch, data between two lightly used computers will be isolated from data intended for a heavily used server, for example. Or in the opposite case, in "auto sensing" switches that allow mixing of 10 and 100Mbps connections, the slower 10Mbps transfer won’t slow down the faster 100Mbps flow.

Although switch prices are dropping so that there is very little difference from hub prices, most home users get very little, if any, advantage from switches, even when sharing "broadband" Internet connections. "Broadband" connections for most users are in the 1-2Mbps range, far below even 10Mbps speeds. Since you share that bandwidth, you can see that your speedy 100BaseT connection isn't even breaking a sweat when you're using the Internet.

**Router**
Routers forward data packets from one place to another, too! However routers are OSI model **Layer 3** devices, and forward data depending on the **Network address**, not the Hardware (MAC) address. For TCP/IP networks, this means the IP address of the network interface.

Routers isolate each LAN into a **separate subnet**, so each network adapter’s IP address will have a different third "octet" (Example: 192.168.1.1 and 192.168.2.1 are in different subnets.). They are necessary in large networks because the TCP/IP addressing scheme allows only 254 addresses per (Class C) network segment.

Routers, like bridges, provide bandwidth control by keeping data out of subnets where it doesn't belong. However, routers need to be set up before they can get going, although once set up, they can communicate with other routers and learn the way to parts of a network that are added after a router is initially configured.
Routers are also the only one of these four devices that will allow you to share a single IP address among multiple network clients.

**QoS (Quality of Service)**

refers to the capability of a network to provide better service to selected network traffic over various technologies, including Frame Relay, Asynchronous Transfer Mode (ATM), Ethernet and 802.1 networks, SONET, and IP-routed networks that may use any or all of these underlying technologies. The primary goal of QoS is to provide priority including dedicated bandwidth, controlled jitter and latency (required by some real-time and interactive traffic), and improved loss characteristics. Also important is making sure that providing priority for one or more flows does not make other flows fail. QoS technologies provide the elemental building blocks that will be used for future business applications in campus, WAN, and service provider networks. QoS is essential for supporting IP telephony or other latency-sensitive traffic. Even if you are using an external router, the system can recognize and prioritize voice traffic, and in many instances this approach can help your company avoid the cost of replacing your router. QoS prioritization takes place within the Business Communications Manager, and voice traffic is then sent through the second LAN interface to the legacy router. To ensure optimum performance, larger companies with high-density IP Telephony environments will need to replace their existing shared 10 Mbps Ethernet hubs with Quality of Service (QoS) capable switches that support the DiffServ, 802.1P, and 802.1Q standards. This higher-bandwidth architecture provides consistent access to the bandwidth required by latency-sensitive telephony traffic. To maximize bandwidth availability across the network.