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Telephony

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Messaging

Voice Over IP (VoIP)

Voice Networking

Call Center

Interactive Voice Response (IVR)

Mobility Solutions

Computer Telephony Integration (CTI)

Virtual Private Networks (VPN)

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Introduction

The Business Communications Manager (BCM) Handbook is a reference tool designed for representatives who sell the Nortel Enterprise product portfolio in North America. The Business Communications Manager product is referred to as BCM throughout this handbook.

This handbook is intended for reference purposes only. Please consult Configuration and Pricing Support, specifically the Global Product Price Catalogue (GPPC) or the Nortel Networks Enterprise Configurator (NNEC). The configurations and applications mentioned in this handbook may not be standard offerings with your company.

The Business Communications Manager

Nortel delivers global, industry-leading enterprise solutions for businesses of all types and sizes. These solutions include communications systems, call center and multimedia messaging applications, in addition to data access products. Our customer-driven solutions increasingly take advantage of Internet and computer integration, helping drive business performance and creating a greater competitive advantage for our customers. One of the Nortel enterprise communications solutions is the BCM.

Originally introduced to the market in early 2000, Nortel BCM solution is one of the most sophisticated and reliable converged voice and data solutions available for branch offices and small- to medium-sized businesses. The BCM is a converged communications system that delivers Nortel reliable and proven voice processing, feature-rich business telephony applications and data networking services over a single platform that is managed via a browser-based tool. As a highly reliable, scalable and integrated voice and data solution, BCM is designed to align with the universal core objectives of the single-site, medium-sized and multisite enterprise to increase revenues, improve customer service, streamline costs and expand market reach.

BCM has been designed to allow customers to build on the platform by adding applications as needed. Risk-averse customers can start with PSTN and migrate to IP when the time is right. With BCM, disparate systems, equipment, and applications such as voice messaging, auto attendant and call center functionality can be consolidated in one box in order to reduce costs and to create a consistent customer and employee experience across the organization.
The BCM Portfolio

BCM comes in a portfolio of three models, each of which has been designed for maximum flexibility and scalability. All of them support pluggable and interchangeable Media Bay Modules that provide the basis for delivering BCM’s many applications. Here are the three models:

- BCM50 supports up to 20 users, and comes in a streamlined combination of a main module plus optional expansion modules.

- BCM200 provides two bays for Media Bay Modules, and has a lower removable tray, similar to that in the BCM400, for improved serviceability of the platform in a lower-cost version. It’s designed to support 32 users or fewer per system.

- BCM400 is available in a standard model or a redundant feature option model, which includes four Media Bay Modules, which simplifies configurations that require four modules and eliminates the need for an Expansion Cabinet. When coupled with an Expansion Cabinet, the BCM 400 can grow to support a maximum of 192 digital stations or up to 90 IP telephones, depending on the configuration.
For multi-site organizations, BCM offers advanced voice and data networking capabilities that allow employees to effectively collaborate, independent of their office location. And, with Unified Manager, or Element Manager on BCM50, SMBs can maximize cost effectiveness and simplify management by using a single point to control and program individual BCM systems. These single-point management tools minimize onsite visits and can reduce or even eliminate the need for dedicated IT staff at each office location.

BCM is a voice and data communications system that delivers Nortel’s reliable and proven voice processing, feature-rich business telephony applications and data networking services over a single platform. BCM’s approach to IP telephony literally transforms multiple networks into a single multi-service network while driving simplicity to the desktop. BCM offers increased application performance and enhancements in the areas of core telephony, data, mobility, management and serviceability. And, it provides unparalleled ROI for vertical markets spanning manufacturing to retail to healthcare.
Introduction to Digital Mobility

The Digital Mobility Solution is a high quality, integrated in-building cellular solution that is ideal for office, industrial, and campus environments. Using a series of base stations and repeaters to extend mobility across a workplace, the Digital Mobility Solution enables handoffs between access points, allowing people to stay connected on a call while moving across the office or campus. The Digital Mobility Solution is based on the Digitally Enhanced Cordless Telephony (DECT) technology. It is supported on the BCM200/400 and scales from 1 to 64 users and covers an area up to 1.5 million square feet for true campus-wide mobility.

For more information, see the chapter entitled “Mobility Solutions”.

Introduction to BCM50

BCM50 is a smaller but robust member of the BCM family aimed at businesses with fewer than 20 employees. While the BCM50 can not offer some advanced capabilities in Call Center and IVR as both the 200/400 do,. Because of the price-sensitivity of typical BCM50 target customers, some applications, like Professional Call Center, Multimedia Call Center, IVR, Digital and WLAN mobility, are not currently enabled on BCM50. Despite this, it fulfills most business needs for small enterprises without incurring excessive costs or equipment. BCM50 is the solution for customers with smaller sites because it delivers on price, while still enabling them to adopt key advanced services when they need them, offering increased platform scalability up to 44 Digital users and/or 32 IP users and core high value applications including Messaging, Unified Messaging and Call Center.

BCM50 has all applications loaded, so that businesses can activate applications as needed using Keycodes, reducing the front-end cost of implementation. BCM50 extends the classic BCM capabilities, previously available only to larger-sized organizations, into a converged, small-site solution that can serve as few as three desktops. BCM50 can be deployed as a pure IP solution, a converged solution, or as a traditional digital solution.

The BCM50 Main Unit is available in three versions:

- A BCM50 with no router
- A BCM50 with Ethernet Router (BCM50e), and
- A BCM50 with ADSL Router (BCM50a).

Customers can choose Ethernet or ADSL, depending on what type of data service they use now. If they want to adopt more advanced data options, they can select a BCM50 without an integrated router and combine that with a more advanced Nortel routing solution, such as Contivity Secure Routing.
Customers can install up to two expansion units, which support optional Media Bay Modules. BCM50 is desk-, wall-, and rack-mountable, so it can fit easily into any workspace.

About This Handbook

This Handbook is organized into short and concise sections that are intended for reference use. This format allows you to easily locate and use the most relevant sections for a current project or request for information. The information you find here will help make your sales easier, quicker and more professional.

This Handbook contains the following chapters:

- Introduction
- Hardware
- Telephony
- Data Capabilities
- Messaging
- Voice Over IP (VoIP)
- Voice Networking
- Call Center
- Interactive Voice Response (IVR)
- Mobility Solutions
- Computer Telephony Integration (CTI)
- Virtual Private Networks (VPN)
- System Management and Software Options
- Glossary
Introduction

> Hardware

Telephony
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BCM Handbook

Hardware

BCM takes advantage of today’s technology and has the following components:

- Base units
- Media Bay Modules (MBM)
- Media Services Card (MSC)
- Business Series Terminals (BST)
- Legacy Terminals
- IP Terminals

*Figure 2.*

- BCM400
- BCM200
- BCM50
BCM200/400 Hardware

The BCM200 and BCM400 chassis comes fully equipped with a 1.2 GHz Intel Celeron processor, 256Mb of RAM, a 20-GB hard drive and a 350-watt power supply, all housed in a 19-inch, rack-mountable chassis. The chassis also comes equipped with integrated features like voice, data and management applications working in concert with Microsoft Windows NT 4.0 Embedded operating system, fully supported into 2006.

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

- 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 (BCM400/BCM200), Linux OS (BCM50)

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 NT 4.0 Embedded operating system is out of service.

The BCM has been designed for flexibility and scalability, with support for pluggable and interchangeable Media Bay Modules. The BCM400 platform has availability for four Media Bay Modules, which simplifies configurations and eliminates the need for the Expansion Cabinet. It also reduces the total cost of ownership for this configuration. When coupled with an Expansion Cabinet, the system can grow to support a maximum of 192 digital stations or up to 90 IP telephones. This is configuration dependent. 240 stations is the maximum capacity, with a mix of IP and digital stations, when 100% IP trunking is used. The BCM400 is available in a standard model or a redundant feature option model, which includes dual, hot-swappable power supply, dual chassis cooling fan and RAID mirrored hard disk drive redundancy.
BCM is also available on the BCM200 platform, which provides two bays for Media Bay Modules. The BCM200 has a lower removable tray, similar to that in the BCM400, for improved serviceability of the platform in a lower-cost version. The BCM200 cannot be expanded using an Expansion Cabinet and is designed to meet the needs of customers with 32 or fewer users per system. The BCM200 comes in the standard version and may be upgraded in the field with an RAID upgrade kit with dual mirrored hard drive.
BCM50 Hardware

For customers with need for a small-site converged solution for 20 or fewer desktops, the BCM is also available on the BCM50 platform.

*Figure 5.*

BCM50 is delivered in a compact, plastic enclosure designed to ensure that there is no need to add additional hardware to enable all features and applications. The compact size and flexible installation options support fast installs and recognize the diverse environmental / physical conditions that can be found in small businesses. BCM50 main module and BCM50 expansion module share the following packaging attributes:

- Approximate dimensions 12” x 8.5” x 2.5”
- External power supply
- Units may be stacked and include design details to “lock” the units together in the stack
- Can be installed on desktop or shelf. Includes rubber feet.
- Optional Wall Mount bracket.
- Optional Rack Mount Shelf
Physical Interfaces

BCM50 is tailored to meet the capacity requirements of many small businesses using the interfaces on the main unit without having to add any further hardware. This optimizes cost effectiveness of the system into locations with less than 10 stations which are often the businesses with the highest price sensitivity. Each system ships with no paths open. You activate them by purchasing Keycodes. The following interfaces are provided on all three variants of the BCM50 main module:

- 12 digital station ports supporting the complete line of Business Series Telephones. These ports are accessible through the front pane RJ-21 connector and are enabled through the use of Keycodes;
- 4 Analog Loop Supervised Trunks in versions using North American networking standards. These ports are accessible through the front panel RJ-21 connector and are enabled through the use of Keycodes;
- 4 Analog Station interfaces with message waiting and CLID support in versions using North American networking standards. These ports are accessible through the front panel RJ-21 connector and enabled through the use of Keycodes;
- Page and auxiliary relay output also provided on the front panel RJ-21 connector;
- 3 port 10/100 Ethernet switch with auto sensing and auto polarity. Two of these ports also support connection of optional expansion units;
- 1 10/100 Ethernet port reserved for direct access management of the system;
- BCM50a also has an ADSL WAN port and 3 additional 10/100 Ethernet switch ports
- BCM50e also has an 10/100 Ethernet WAN port and 3 additional 10/100 Ethernet switch ports
- Music on hold input supported either through front panel jack or RJ-21 connector,
- USB port that is used to enhance BCM50 management and connectivity.

Capacity of the system can be extended using the optional BCM50 Expansion unit, as described in a subsequent section.

BCM50 Expansion Unit

The BCM50 Expansion Unit (only available with BCM50) is a cost-effective way to increase the capacity of the BCM50. Connection is via a standard 10/100 Ethernet cable directly from the two expansion ports on the BCM50 main unit. The Expansion Unit supports the following Media Bay Modules:

- GATM 4/8
• DSM 16+/32+
• 4 x 16 (Combo)
• GASM8
• BRIM S/T

Up to two BCM50 Expansion Units are supported and each Expansion Units supports one Media Bay Module.

• One RJ-45 connector for the interface to the BCM50 Main unit
• One 10/100 Ethernet switch port for customer use

A factory-supplied cable is used to connect the Expansion Unit to the BCM. If necessary, any standard 10/100 Ethernet LAN cable may be substituted.

BCM50a/e Integrated Routers

The BCM50a and BCM50e configurations are available with an optional integrated router. This option is best suited for small businesses that might have up to 5 people using external data networking, for such applications as internet access and enterprise branch networking. The variants of the BCM50, with or without router capability, allow for the best value for the customer.

The two variations of the BCM50 main module that provide an optional integrated router vary according to the type of WAN interface desired. The BCM50a includes an Ethernet router, while the BCM50e uses an ADSL WAN. Both routers share the same rich set of data features that make the system attractive to a variety of applications:

• Secure internet access
• Multi-site VoIP trunking using secure VPN tunnels
• Wide area VoIP applications with remote user support
• Remote management and support
Common Features and Capabilities

Both versions of the router share the same rich set of functionality focused on secure internet access and VoIP. The BCM50 equipped with the optional integrated router delivers on the promise of convergence to the small business or enterprise branch.

The BCM50e or BCM50a main unit with optional router provides an additional 3 ports of ethernet LAN switch for a total of 6 LAN ports for local premise use. Because all Ethernet ports are 10/100 Mbps auto sensing, and support auto polarity, no cross-over cable is required to connect data hardware to the unit. An additional port is provided for WAN access, either Ethernet or ADSL depending on the model.

The following features make the variants of the BCM50 with embedded router an attractive package for small sites wishing to become Internet-capable, multisite enterprises with many small sites which formerly could not be part of the corporate WAN due to high cost of traditional WAN connectivity, and managed service scenarios:

**VPN**

- 5 IPSec tunnels
- IKEv1 Main Mode
- IKEv1 Quick Mode
- Diffie-Hellman Group 1,2
- IPSec Tunnel Mode
- ESP
- Support for Dynamically addressed peers – ABOT
- NAT Traversal

**Security Services**

- Cryptographic Services
- 3DES
- DES
- Data authentication SHA-1
- Data authentication MD-5
- Authentication Services
- Pre-shared secrets
• Security Services
• Stateful Firewall
• Intrusion Detection

**NAT**

• Many to one, static, many to many
• Port forwarding
• IPSec Pass through
• H.323 ALG
• NAT support for tunnel mode IPSec tunnels

**Router**

• Clear text routing
• Static – via tunnel
• RIP v1 – via tunnel and clear text
• RIP v2 – via tunnel and clear text

**IP Services**

DHCP client

DHCP server with support for Nortel Networks Internet Telephones

DNS Proxy

DNS w/VPN client

PPPoE

Configurable MAC address

5 Mbps clear text routing with 1500 byte packets

1.5 Mbps 3DES throughput with 1500 byte packets
**Ethernet WAN variant**

BCM50e is intended for those customer premise configurations which have an existing data infrastructure. These versions deliver the VoIP convergence value into the customers existing data network.

In this model, the WAN interface port provides 10/100 Ethernet with auto sensing and autopolarity. Customers with existing or alternative WAN access technology can still benefit from the VoIP features of the integrated router.

**ADSL WAN variant**

BCM50a is targeted at stand-alone “office in a box” applications to businesses served by ADSL. With one product, a customer can be setup for complete voice and internet service with resulting efficiency and convenience. It is very well suited to those partners that wish to offer bundled telephony and internet services. The following features provide a complete integrated ADSL access package for ease of interconnecting with service provider ADSL networks.

- ITU G.992.1 (G.DMT)
- G.992.1 Annex A
- ITU G.992.2 (G.Lite)
- ANSI T1.413 Issue 2
- DSL Forum document TR-042 ATM Transport over ADSL.
- G.hs 994.1
- G.ploam G.997.1
- Auto negotiation rate adaptation.
- RFC 2364 PPP over AAL5
- RFC 2684 Multiprotocol Encapsulation over ATM, both Bridged and Routed encapsulation
- Support for British Telecom SIN 329; BT Broadband IP Products requirements for End User NTE equipment, where the “router” and ADSL “modem” functions are both integrated into a single device.
- RFC 1483 “Multi-protocol over AAL5”
- RFC 2365 “PPP over AAL5”
- RFC 2516 PPPoE
- Traffic shaping UBR , CBR
• ATM forum UNI 3.1 / 4.0 PVC (minimum 5 PVC’s)

BCM50 Data Networking Hardware Components

On the BCM50, the first 10/100 Ethernet ports is reserved for craftspersons access. This port has a fixed IP address (10.10.11.1) and has a DHCP server. The second 10/100 Ethernet port been designated for connection to the local area network (LAN) and the wide area network (WAN). The customer may use the third and forth ports as a 10/100 Ethernet switch ports or to support the Expansion Unit if equipped. The BCM50a has a DSL WAN interface and the BCM50e has a 10/100 Ethernet WAN interface.

BCM50 Modem

The BCM50 has an internal soft modem that allows any line connected to the BCM50 to be used as a modem line, eliminating the need for a separate, dedicated modem line. The internal soft modem supports up to 33.6 Kbps.

BCM50 LAN Interface

The LAN interface is used to connect the BCM system to the LAN. The BCM50 has one dedicated 10/100 Base T Ethernet port for craftperson access and a second 10/100 Base T Ethernet port to interface to the customer’s LAN.

The BCM Ethernet/802.3 interface supports the IBCME 802.3 Ethernet frame format. The Ethernet interface uses Carrier Sense Multiple Access with Collision Detection (CSMA/CD) to manage the access to the physical media.

The BCM Ethernet interface supports the following features:

• 100 BASE -TX with RJ-45 connector
• 10 / 100 Auto Sense
• Half or Full Duplex
• Fast path forwarding in a LAN-LAN routing environment using card drivers(BCM200/400)
• LAN traffic smoothing via interrupt modulation and increased buffer size
• Point-to-Point Protocol over Ethernet (PPPOE)(Optional)
• DiffServ queuing
• Supports IEEE 802.3 format
• Utilizes CSMA/CD for physical media access.
BCM200/400 Components

Connection Ports

Serial Port

The BCM200/400 is equipped with one serial port that supports an asynchronous serial data management interface. The port has a male DB-9 connector and supports all standard baud rates (9600 default).

The serial port connects serial devices, such as a laptop computer. An engineer uses this port to set the initial IP address on the BCM before connection to the customer’s LAN.

A null modem cable is required for this connection. Alternately, a crossover Ethernet cable can be connected directly between a Network Interface Card on the BCM and a Network Interface Card on a laptop. The other ports include a USB port, modem and serial connection point.

Business Communications Manager LEDs

The BCM200/400 is configured with ten LEDs mounted on the front panel. These LEDs are assigned the following functionality:

Table 1.

<table>
<thead>
<tr>
<th>Bezel Indicator</th>
<th>Indicates</th>
<th>Green LED On</th>
<th>Green LED Flash</th>
<th>Red LED On (Only)</th>
<th>Green LED Off</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Power supply(s) good</td>
<td>Good</td>
<td>N/A</td>
<td>At least 1 PS needs attention</td>
<td>N/A</td>
</tr>
<tr>
<td>2</td>
<td>Hard drive activity</td>
<td>Indicates activity only</td>
<td>N/A</td>
<td>N/A</td>
<td>N/A</td>
</tr>
<tr>
<td>3</td>
<td>Applications status</td>
<td>All monitored services are functioning</td>
<td>System may be in startup or shutdown mode</td>
<td>N/A</td>
<td>All monitored services are not functioning</td>
</tr>
<tr>
<td>4</td>
<td>PCI Device / WAN Port # 1 or NIC 2</td>
<td>Device is present and driver is functioning</td>
<td>Device is present but driver is not running</td>
<td>N/A</td>
<td>Device is not present</td>
</tr>
<tr>
<td>5</td>
<td>PCI Device/ WAN Port # 2</td>
<td>Device is present and driver is functioning</td>
<td>Device is present but driver is not running</td>
<td>N/A</td>
<td>Device is not present</td>
</tr>
<tr>
<td>Bezel Indicator</td>
<td>Indicates</td>
<td>Green LED On</td>
<td>Green LED Flash (Only)</td>
<td>Red LED On</td>
<td>Green LED Off</td>
</tr>
<tr>
<td>----------------</td>
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<td>------------------------</td>
<td>------------</td>
<td>--------------------------------</td>
</tr>
<tr>
<td>3</td>
<td>PCI Device Modem</td>
<td>Device is present and driver is functioning</td>
<td>Device is present but driver is not running</td>
<td>N/A</td>
<td>Device is not present</td>
</tr>
<tr>
<td>4</td>
<td>PCI Device MSC</td>
<td>Device is present and driver is functioning</td>
<td>Device is present but driver is not running</td>
<td>N/A</td>
<td>Device is not present</td>
</tr>
<tr>
<td>5</td>
<td>PCI Device NIC1</td>
<td>Device is present and driver is functioning</td>
<td>Device is present but driver is not running</td>
<td>N/A</td>
<td>Device is not present</td>
</tr>
<tr>
<td></td>
<td>Temperature good</td>
<td>Temperature is below threshold</td>
<td>N/A</td>
<td>Temperature is in alarm status</td>
<td>N/A</td>
</tr>
<tr>
<td></td>
<td>Fans good</td>
<td>All installed fans are working</td>
<td>N/A</td>
<td>There is a problem with at least one fan</td>
<td>N/A</td>
</tr>
<tr>
<td></td>
<td>RESET button access</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Telephony Hardware Components**

The telephony components perform call processing and connect the BCM to the public switched telephone network (PSTN) lines and to the BCM telephones. The main telephony hardware components of the BCM system are:

- Media Services Card (MSC) on BCM200/400
- Station Media Bay Modules
- Trunk Media Bay Modules
- Fiber Expansion Media Bay Module (BCM-FEM) BCM200/400 only

**Media Services Card**

The Media Services Card (MSC) on the BCM200/400 is a field replaceable unit (FRU) to allow authorized BCM service providers to replace this card in the field, as opposed to sending the entire system in for repair. The MSC performs call processing and media processing of the voice channels. It also provides the processing power for
telephony functions independent of the Windows NT 4.0 Embedded operating system. This means that if the NTE operating system should malfunction, the MSC can still process telephone calls.

However, if Windows NTE is down for any reason, voice applications such as Messaging, Call Center or IP telephony will not function. The MSC also provides the processing power for the voice channels and compression utilities for Messaging, Call Center and IP telephony.

Installed on the MSC are 2 to 4 Processor Expansion Cards (PECs), which provide digital signal processing (DSP) resource control. The PEC provides DSP resources to translate analog and digital signals and process them into a format useable by the system. The DSPs support voice applications, including Messaging, Call Center and IP telephony. All of these voice applications can share the DSP resources on one BCM platform.

The BCM platform comes equipped with two PEC IIIIs on the BCM400 and one PEC III on the BCM200. Depending on applications requirement, the BCM400 can be optionally equipped with four PEC IIIIs and the BCM200 can be optionally equipped with two PEC IIIIs.

**Figure 6.**

The MSC also provides the following functions:

- Connection between the MSC and the pluggable Media Bay Modules
The MSC has four DS256 interfaces used to connect to the Media Bay Modules installed in the BCM. The DS256 connectors are 2 x 5 pin headers located along the top edge of the MSC. A 10-conductor ribbon cable connects the MSC to the pluggable Media Bay Modules.

- Connection to optional equipment

The BCM400 MSC has one RJ-45 connector located on the faceplate. This allows one Expansion Cabinet to be connected with the base system, providing a means to add from one to six Media Bay Modules, installed in the Expansion Cabinet, to the system. On the BCM200, there is no RJ-45 connector on the MSC and an Expansion Cabinet cannot be connected.

The MSC has four 3.5 mm (1/8 inch) miniature jacks located on the faceplate. These jacks are standard miniature stereo (three-conductor) jacks. All four interfaces are safety extra low voltage (SELV) and the external equipment connected to these interfaces must be SELV. If these interfaces are not SELV, external line isolation units (LIU) must be used.

*Figure 7.*

The miniature jacks are used to connect the following optional equipment:
• Music on Hold input

The BCM 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 customer-supplied radio or music source, provided that it is approved for connection to the network.

The music source connects to the tip and sleeve terminals of the miniature jack. The sleeve terminal of the jack connects to ground. A mono or stereo plug can be used to connect the music source. However, the Music on Hold input only accepts a mono input.

• IP Music on Hold

This feature leverages the Internet and the native capabilities of BCM to provide background music for callers while on hold. By connecting to a data source, audio information is passed directly to BCM. Common formats such as .wav or .ra formats are supported. With this feature, customers now have two ways to provide music while on hold: by playing audio from an external source, whether streaming or finite file, or playing audio from files stored directly on BCM. IP Music on Hold is offered on BCM200/400 only.

• Page Output

The BCM uses Page Output to connect an internally generated voice paging signal to an external paging amplifier (customer supplied). This signal is transformer coupled and is floating with respect to earth ground. The signal has a nominal source impedance of 600 ohms. The output level is 0 dBM with reference to 600 ohms, for a PCM encoded signal at 0 dBM. There is no dc voltage across the page output terminals.

The Page Output uses the tip and ring terminals of the jack. The sleeve terminal of the jack connects to ground. A stereo plug must be used to connect the page signal output.

• Page Relay

When the Page Signal Output jack is used to connect an external paging amplifier, the Page Relay jack is also used. The Page Relay jack connects a floating relay contact pair. The BCM uses this jack to control the external paging amplifier. The contact pair has a switch capacity of 50 mA (noninductive) at 40 V (maximum). Any inductive load on the output must be removed.

The sleeve of the jack connects to ground. The Page Relay contacts connect to the tip and ring terminals of the jack. A stereo plug must be used to connect the Page Relay.

• Auxiliary Ringer

The BCM uses the Auxiliary Ringer jack to control the cadence of an auxiliary ringer (customer supplied). This output must be used in a low current, low voltage application only. This output must not be used for
switching the Auxiliary Ringer directly. The contact pair has a switch capacity of 50 mA (noninductive) at 40 V (maximum). Any inductive load on the output must be removed.

The sleeve of the jack connects to ground. The Auxiliary Ringer connects to the tip and ring terminals of the jack. A stereo plug must be used to connect the Auxiliary Ringer.

**Station Media Bay Modules**

Station Media Bay Modules connect to telephones and analog telecommunication devices. All Station Media Bay Modules are site pluggable in the BCM unit. The BCM portfolio includes the following Station Media Bay Modules:

**16-Port Digital Station Media Bay Module (BCM-DSM 16+)**

The BCM-DSM 16+ connects up to 16 telephones to the BCM. An Amphenol connector on the faceplate attaches to the cross-connect array. The faceplate also has two LEDs labeled as follows:

- Power (indicates operating status)
- Status (indicates hardware status).

*Figure 8.*
32-Port Digital Station Media Bay Module (BCM-DSM 32+)

The BCM-DSM 32+ connects up to 32 telephones to the BCM. Two Amphenol connectors on the faceplate attach to the cross-connect array. The faceplate also has two LEDs:

- Power (indicates operating status)
- Status (indicates hardware status)

*Figure 9.*

Global Analog Station Module 8 (GASM8)

The Global Analog Station Module (GASM8) provides connectivity to eight analog stations, along with additional features. Analog support includes terminals, fax machines, answering machines and modems up to a 28.8 speed. The GASM8 is backwards compatible with earlier versions of BCM. It supports both ASM8 and ASM8+.

Available as of BCM 3.6, the GASM8 includes:

- Message Waiting Indication (MWI)
- Caller ID (CLID)
- Disconnect Supervision (DS) (as of BCM 3.6)
- Downloadable firmware (DF) (as of BCM 3.7)
- Country selectable profiles via dipswitch

The GASM8 has two LEDs on the faceplate labeled as follows:

- Power (indicates working status)
- Status (indicates hardware status).

*Figure 10.*
Trunk Media Bay Modules

Trunk Media Bay Modules connect telecommunications trunks to the BCM system. The following types of trunk Media Bay Modules are available.

**Digital Trunk Media Bay Module (BCM-DTM)**

The BCM-DTM is a trunk module that connects a T-1 or PRI trunk to the BCM system adding up to 24 digital telephone lines. On international BCM systems, the BCM-DTM connects to an E1 or PRI digital line. With an E1 or PRI line, up to 30 digital telephone lines can be added. A maximum of three BCM-DTM modules can be installed on the BCM system. The Digital Trunk Module is supported in the BCM main cabinet only (the DTM is not supported in the expansion chassis). (R2 MFC E-1 is not supported at this time, but is planned for release in 2005 (CALA).)

The front faceplate of the BCM-DTM has a number of LEDs that indicate power status and any ongoing tests and alarms that the module is undergoing. The faceplate also has an RJ-48C connector that connects the BCM-DTM to the service provider’s connection point and a set of loopback connectors used to run loopback tests.

*Figure 11.*

**Digital Drop and Insert Mux (DDIM) Module**

The Digital Drop and Insert Mux (DDIM) Module is a module that fits into the BCM and combines the functionality of the standard BCM Digital Trunk Module with a built-in Drop and Insert CSU. It accepts a T-1
with both voice and data from a service provider and splits off the channels carrying data and routes them to an interface on the front of the module.

**Figure 12.**

The front faceplate of the DDIM module has LEDs that indicate power, status and ongoing test and alarms similar to the Digital Trunk Module. It also has LEDs that indicate the status of the serial data interface, including Transmit, Receive, RTS, CTS, DCD, DSR and TM.

**Figure 13.**
The DDIM is supported in the main BCM200/400 cabinet only (the DDIM is not supported in the Expansion chassis) and supports standard T-1 only (not PRI). The interface is a V.35 in the form of a miniature DB-26 connector. A variety of cables are available that connect to either the BCM WAN card to take advantage of the BCM internal router, or to external Nortel routers and other third party routers. The DDIM cables include:

- DB-26 interface to connect to BCM WAN
- DB-44 interface to connect to Nortel routers
- DB-60 interface to connect to third party routers
- Standard V.35 with M34F interface.

In addition to providing a network service revenue opportunity, the DDIM streamlines implementation costs, as it is easier to install and configure than a separate, external Drop and Insert CSU/DSU. The DDIM also streamlines ongoing management costs as it is managed through the same Unified Manager as BCM.
**Global Analog Trunk Media Bay Module 4 (BCM-CTM 4)**

The BCM-GATM 4 port connects up to four analog CLID PSTN lines to the BCM system. The auxiliary port permits the connection of a 33.6+ Kbps modem, fax machine or single line analog telephone to line 1. When the auxiliary device is using line 1, the BCM system does not allow other telephones to use line 1. When a single line analog telephone is connected to the auxiliary port, it can be used as an emergency telephone. A male Amphenol connector on the faceplate attaches to the cross-connect array.

The BCM-CTM faceplate also has two LEDs:

- Power (indicates operating status)
- Status (indicates hardware status).

*Figure 14.*
Global Analog Trunk Media Bay Module 8 (BCM-CTM 8)

The BCM-GATM 8 connects up to eight analog CLID PSTN lines to the BCM system. One auxiliary port permit the connection of a 33.6+ Kbps modem, fax machine or single line analog telephone to line 1. When the auxiliary device is using line 1, the BCM system does not allow other telephones to use line 1 or 5. When a single-line analog telephone is connected to the auxiliary port, it can be used as an emergency telephone. A male Amphenol connector on the faceplate attaches to the cross-connect array.

The BCM-CTM faceplate also has two LEDs:

- Power (indicates operating status)
- Status (indicates hardware status).

Figure 15.

Basic Rate Interface Media Bay Module (BCM-BRIM S/T)

The BCM-BRIM S/T pluggable module connects up to four BRI S/T ISDN lines to the BCM system. Each BRI S/T ISDN line that is connected adds two telephone lines to the BCM system. Therefore, each BCM-BRIM S/T adds up to eight telephone lines to the system. BRI is often delivered as a U interface by telcos in North America. An external NTI can be used to convert the U interface to an S/T interface that is compatible with the BRIM S/T.
The front faceplate of the BCM-BRIM S/T has four RJ-48C connectors that connect the BCM-BRIM S/T to the service provider’s connection point. On the left side of the RJ-48 connectors are LEDs that show the status of the ISDN lines.

Each BCM-BRIM S/T also has two LEDs on the faceplate labeled as follows:

- Power (indicates working status)
- Status (indicates hardware status).

**Figure 16.**

---

**Fiber Expansion Media Bay Module (BCM-FEM)**

Fiber Expansion Media Bay Modules connect Norstar Fiber Station and Trunk modules to the BCM200/400 system. One Fiber Expansion Module is available for the BCM system. The BCM-FEM connects up to six Norstar Fiber Station or Trunk modules to the BCM system.

Normally, Norstar expansion modules are used to connect PSTN lines and telephones to a Norstar system. With the BCM-FEM, these expansion modules can be connected to the BCM system. The BCM-FEM is very useful when a customer is migrating from an existing Norstar system to BCM.
In new installations, where Direct Inward Dialing (DID), or tie-lines, cannot be provisioned over a T-1 or PRI, the BCM-FEM can be used to support Norstar Analog DID or Analog E&M trunk cartridges to satisfy this requirement.

The front faceplate of the BCM-FEM has six connectors that connect the BCM-FEM to the expansion modules. The connections are made using fiber cables. On the right side of each connector, there is an LED that indicates if the fiber port is enabled. If the LED is on, the fiber port is enabled and it can be used to connect a Fiber Station or Trunk module.

The BCM-FEM has two LEDs on the faceplate labeled as follows:

- Power (indicates working status)
- Status (indicates hardware status).

*Figure 17.*

**4x16 Combo Media Bay Module**

The 4x16 combines a CTM with four analog trunks and a DSM 16 into a single module. Combining the CTM and the DSM 16, a single module provides analog trunk access and digital station interfaces. The 4x16 module provides increased flexibility for the small site, enabling small line and station configurations to be supported without the expansion chassis.

The CTM portion of the 4x16 provides 4 analog trunk interfaces to the telco central office. Each trunk interface supports Bellcore Caller ID and disconnect supervision. In addition, trunk 1 provides an auxiliary port that allows
an analog telephony device, like a modem, a fax machine or an analog telephone, to share this trunk. The operation of this auxiliary port is identical to the auxiliary port in the CTM4 and CTM8.

The DSM16 portion of the 4x16 module de-multiplexes a DS-30 channel into 16 digital phone interfaces. Each digital phone interface supports two bidirectional channels.

The 4x16 module uses one and one-quarter DS30 channels in the DS256 serial bus.

The CTM portion of the 4x16 module requires one quarter of a DS30 channel and its DS30 channel number is selected by the DS30 channel number dip-switches. The DSM 16 portion of the 4x16 module requires an entire DS30 channel. It will use the next adjacent DS30 channel number to which the DS30 dip-switches are set.

The 4x16 is available in North America only.

Figure 18.

The 4x16 Module combines the functionality of:

- 4 port Calling Line ID Trunk Module (CTM4)
- 16 port Digital Station Module (DSM 16).
Business Communications Manager Expansion Cabinet

The Expansion Cabinet (only available with BCM400) is a cost-effective way to increase the capacity of the BCM. Connection is via a DS256 cable directly from the Media Services Card to the Expansion Cabinet. The Expansion Cabinet is backwards-compatible with earlier BCM Releases and supports the following Media Bay Modules:

- GATM 4/8
- DSM 16+/32+
- 4 x 16 (Combo)
- GASM8
- BRIM S/T
- FEM

The BCM Expansion Cabinet houses an additional six bays for Media Bay Modules, excluding the DTI or DDIM. It also contains a cooling fan, a power supply and a hub card.

The hub card is an interface card mounted on the inside of the BCM Expansion Cabinet. The hub card provides a connection between the MSC and the Media Bay Modules. Inside the Expansion Cabinet are the following connectors:

- One RJ-45 connector for the interface to the MSC
- Six DS256 module connectors.

A factory-supplied cable is used to connect the Expansion Cabinet to the BCM. This cable must be exactly five meters (16 feet) long. The hub card has a six-position DIP-switch. Switch position number 1 adjusts the timing on the DS256 bus to manage the cable length between the MSC and the hub. Only the factory-supplied five-meter cable is supported. Do not substitute any other cable. The switch position number 1 is set as 0 for a five-meter cable.
Data Networking Hardware Components

The data networking components connect the BCM200/400 to the local area network (LAN) and the wide area network (WAN). The BCM platform comes complete with two LAN ports.

The on-board LAN port is a fully auto-sensing network interface. The field installable WAN card supports Frame Relay and Point-to-Point protocols.

The factory-installed data networking hardware components of the system include:

- V.90 modem card (North America only)
- LAN interface.

The data networking hardware component is a WAN interface card, which is available as a field installable upgrade.

Modem

The BCM200/400 have a V.90 embedded modem that is used to send and receive data using the public telephone system. This connection can be used to manage the BCM system from a remote location. This on-board interface can also be used as a dial backup for the WAN link. The V.90 modem has the following features:

- RJ-11 connector
- V.90 56 Kbps ITU standard
• V.34 33.6 Kbps ITU standard
• V.42/MNP 2-4 error control
• V.42/MNP 5 data compression.

The modem is capable of receiving data at up to 56 Kbps and sending it at 31.2 Kbps. However, due to FCC regulations, receiving speeds are limited to 53 Kbps. The actual speed may vary depending on the transmission quality of the line.

LAN Interface

The LAN interface is used to connect the BCM system to the LAN. BCM includes two on-board 10/100 Base T Ethernet ports on the BCM200/400.

The BCM Ethernet/802.3 interface supports the IBCME 802.3 Ethernet frame format. The Ethernet interface uses Carrier Sense Multiple Access with Collision Detection (CSMA/CD) to manage the access to the physical media.

The BCM Ethernet interface supports the following features:

• 100 BASE -TX with RJ-45 connector
• 10 / 100 Auto Sense
• Half or Full Duplex
• Fast path forwarding in a LAN-LAN routing environment using card drivers (BCM200/400)
• LAN traffic smoothing via interrupt modulation and increased buffer size
• Point-to-Point Protocol over Ethernet (PPPOE) (Optional)
• DiffServ queuing
• Supports IEEE 802.3 format
• Utilizes CSMA/CD for physical media access.

WAN Interface Card

The BCM200/400 can support an Optional WAN interface card that is used to connect the BCM system to the wide area network. It is a field replaceable unit (FRU), simplifying configuration choices. All customers will have two Ethernet ports available and will be able to add a WAN if desired. Two WAN interface cards are available. One has a T-1 interface port, a built-in CSU and a serial sync port, and the second has two serial sync ports. These
two ports can be independently configured to run Frame Relay or Point-to-Point Protocol (PPP). MultiLink PPP (MLPPP) support for dial-on-demand is supported over the DTM. STAC compression is available.

A special cable is required to connect the serial sync port, which supports a maximum line speed of 8 Mbit/sec over V.35 or X.21 interfaces. The V.35 or X.21 cable is only required if the serial interface on the WAN card is used to connect to an external access device. This may occur if a 56K or 64K digital data service or a managed service that includes an external CSU/DSU (in order for the service provider to do testing and so on) is used.

Normally, a T-1 or fractional T-1 will be terminated on the RJ-45 connector and use the internal CSU on the WAN card. Subscribers should check with the service provider for the required cable type.

**WAN Interface Card Features**

- Two Port PCI card (independently configured)
- Frame Relay (FR)–FRF.9 compression protocol and STAC compression algorithm
- Point-to-Point Protocol (PPP)
- Integrated T-1 w/CSU
- RJ-48C Connector for T-1
- DB-26 female serial connection for V.35.

**Serviceability Improvements**

The hardware platforms have one or two removable trays to improve access to the hardware for service and support. The lower tray, common to both the BCM400 and BCM200, provides access to the motherboard, CPU, RAM, MSC and modem card. The WAN card is also installed in this tray. The upper tray, in the BCM400 only, provides access to the hard drive and RAID card (if so equipped). Access to the power supply is still from the back panel and does not require access through the tray. The Media Services Card is now available as an FRU to allow replacement of this card in the field as opposed to sending the entire system in for repair (BCM200/400 FRU only).

**Upgrade Support for Installed Base**

The BCM upgrade kit provides a CD-ROM with which to upgrade BCM base systems in the field to the latest release. The upgrade allows users with BCM 3.5 and 3.6 to upgrade their systems to release 3.7. User programming and data are preserved during the upgrade. The upgrade kit includes installation instructions and no hardware changes are required to carry out the upgrade. Client operating systems supported for running the
upgrade from desktops include Microsoft Windows 2000, Windows XP and Windows NT 4.0 (Windows 95/98/ME is not supported).

**Business Series Terminals**

Business Series Terminals offer a feature-rich portfolio with enhanced capabilities that provide telephony solutions for a broad landscape of users, from high-volume call positions and executives to low-intensive users and small workgroups.

The Business Series Terminals are flexibly positioned for deployment on two system platforms – Norstar and BCM, providing both investment protection and a migration path between either system. The Business Series Terminals offer full integration with Norstar and BCM features, as well as integration with basic and advanced applications such as Messaging, Call Center, Computer Telephony Integration (CTI) and integrated voice and data solutions.

While the Business Series Terminals boast the industry leadership and strengths of the Norstar telephone portfolio, the portfolio also delivers value-added features, including:

- **Tilt Display** – provides clearer viewing of information or message prompts on the LCD in different lighting environments.

- **Message waiting indication (MWI)/Visual ringing lamp** – alerts the user of incoming messages or that their phone is ringing when they are on another call.

- **Headset interface** – is driven from the Digital Terminal Interface Chip (DTIC). Volume control for the headset is also provided. Operation of the headset is mutually exclusive, with handsfree operation. When a headset is connected, all operations normally associated with handsfree operation affect the headset. This includes on-hook dialing, volume control while active and muting.

- **Handsfree interface** – is programmed through the administration function and is supported by a microphone and loudspeaker.

- **External ringer interface** – receives alerting signals that are routed to the external ringer jack as well as to the speaker in the telephone. This alerting signal can be amplified and connected to external speakers to provide an auxiliary ringer function for the telephone. The external speaker is connected with a two-wire modular telephone cord to pins 3 and 4 of the external ringer jack.

The portfolio also offers tilt display, new aesthetics, a streamlined footprint, new labeling strategy, an audio control center with a headset button and more.
Sets are available in the following colors:

- Platinum
- Charcoal.

Overview of Portfolio

Overview of T7100

*Figure 20.*

The T7100 telephone is part of the Business Series Terminals portfolio. It has one programmable button and a 1 x 16 character alphanumeric display to provide call progress information. There are no display buttons (soft keys) on the T7100 and it does not support headset or Handsfree.

T7100 supports the following features:

- External ringer interface
- Message waiting indicator/visual ringing lamp
- LCD with tilt Display – The one-line by 16-character display on the T7100 telephone provides call progress information. Each character is generated from a matrix of 5 x 7 LCD dots under control of a built-in controller chip on the module.

*Button Matrix*

The T7100 buttons are as follows:

- Twelve dial pad buttons
• Volume control (rocker type)
• Hold button
• Release button
• Feature button
• One programmable button

Loop Limits
• Maximum loop length – 305 m (1000 ft) or 0.5 mm (24AWG) wire
• Maximum loop length – 790 m (2600 ft) with SAPS option
• Bridge taps – not permitted
• Loading coils – not permitted.

Overview of T7208

Figure 21.

The T7208 telephone is part of the Business Series Terminals portfolio. It has eight fully programmable buttons, each with its own LCD indicator and a 1 x 16 character alphanumeric display to provide call progress information. There are no display buttons (soft keys) on the T7208.

The T7208 supports the following features:

• External ringer interface
• Headset interface
• Handsfree interface
• Message waiting indicator/visual ringing lamp
- LCD with tilt Display – The one-line by 16-character display on the T7208 telephone provides call progress information. Each character is generated from a matrix of 5 x 7 LCD dots under control of a built-in controller chip on the module.

**Button Matrix**

The T7208 buttons are as follows:

- Twelve dial pad buttons
- Volume control (rocker type)
- Hold button
- Release button
- Feature button
- Eight programmable buttons with indicators.

**Loop Limits**

- Maximum loop length – 305m (1000 ft) of 0.5 mm (24 AWG) wire
- Maximum loop length – 790 m (2600 ft) with SAPS option
- Bridge taps – not permitted
- Loading coils – not permitted.

**Overview of T7316E**

*Figure 22.*

The T7316E telephone is part of the Business Series Terminals portfolio. It has one programmable button and a 2 x 16 character alphanumeric display that is menu driven and supported by three context-sensitive soft keys. The
T7316E provides access to 24 memory buttons, 16 of which include multi-segment icons for fast and precise decision-making.

The T7316E supports the following features:

- BCM and Norstar compatible
- Multi-line w/multi-segment call appearance icons
- 16 programmable buttons for lines/features/autodials
- 8 additional memory buttons for features/autodials
- Expandable by adding T24 KIM
- Integrated BLF/DSS
- Fixed buttons: feature, hold, release
- Two-line adjustable display w/soft keys
- Built-in speakerphone
- Message Waiting Indication
- Time and date displayed
- Built-in headset
- Audio control center with mute, hands free, headset and volume bar
- Default features based on profiles
- Default line and hunt group assignment
- Retractable quick reference card
- Desk or wall mount
- Hearing aid compatible
- Desktop Assistant label application compatible
- ITU dial pad
- Colors: charcoal and platinum
- English and French keycaps (accessory)
- Extra length 2.7m handset cord (accessory)
- Shoulder rest (accessory)
Overview of BST Central Answering Position (T7316E + T24)

*Figure 23.*

The BST Central Answering Position is an expandable desktop telephone that allows administrative assistants and emergency call centers the ability to centralize and efficiently distribute calls. By attaching the T24 Key Indicator Module (KIM) directly to the T7316E, office administrators and emergency call centers can transform the pace and efficiency of their operations while maintaining an exceptional level of customer service.

**T24 Product features:**

- BCM and Norstar compatible
- Multi-line with multi-segment call appearance icons
- 24 programmable buttons for lines/features/autodials
- Busy Lamp Field/Direct Station Select support
- Multiple appearances of hunt groups
- Multiple line appearances
- 4 T24’s per T7316E without power supply
- 9 T24’s per T7316E with power supply
- Desk or wall mount
- Colors: charcoal and platinum
The following table highlights configuration rules:

<table>
<thead>
<tr>
<th>Max # sets per system</th>
<th>Enhanced KIM (use KIM for multiple appearances of target lines)</th>
<th>Ordinary KIM (use KIM for BLF/DSS)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Max # sets</td>
<td>Max # EKIM on each set</td>
</tr>
<tr>
<td>96</td>
<td>12</td>
<td>4</td>
</tr>
</tbody>
</table>

**BST Doorphone**

This device enables office personnel to talk directly with visitors prior to their entering a business. When a visitor presses the Doorphone’s call button, the BCM rings the designated phone(s) in an office and allows two-way conversation. The optional Door Opening Controller enables any BST to control a latch on a door or gate. The BST Doorphone is supported on the BCM200/400.

**Advanced Features**

**Audio Control Center**

**Headset Button**

The Headset button has the following capabilities:

- Users can leave their Headset plugged in and toggle between Headset, Handset and Handsfree
- Users can press the Headset button to toggle from either Handset or Handsfree to Headset
- Users can press the Handsfree button to toggle from Headset to Handsfree
- Users can lift the Handset to toggle from Headset to Handset
- The Headset LED will be solid when the Headset is activated
- The Headset button does not work until a Headset is plugged in – the set senses the Headset
- Users can answer incoming calls by pressing either the Headset button or the incoming line appearance.

There are three speech paths: Handset, Handsfree or Headset.
The speech path is dictated by the previous call. For example, if a person had used the Headset on the previous call, the speech path would immediately go to the Headset when the user pressed a ringing line to answer the next call. A user could also press Handsfree or pick up the Handset to have the call go to those speech paths.

**Note:** Nortel does not support the connection of Headsets to the T7208 or T7316E telephones, unless Handsfree is enabled within the system programming.

**Mute Button**

The Mute button has the following capabilities:

- Mutes Handset, Handsfree, or Headset
- The Mute LED flashes when on Mute
- Button inquiry (F*1) of the Handsfree Key displays “Handsfree/Mute.” This message is sent from the KSU or BCM, which does not know if the set is a Business Series Terminal or a Norstar telephone set. (This messaging is required to ensure that KSU and BCM compatibility is the same as with the Norstar Telephone sets.)
- The display does not show “Microphone Muted” when the Mute button is on
- A muted call placed on hold is no longer muted when a user retrieves it. This feature is different with the Norstar sets.

If users press Handsfree when a call is on hold, their set will display the message “Microphone Muted.” To unmute the set, users need to press the Handsfree key again. (This messaging is necessary to ensure that KSU and BCM compatibility is the same as with the Norstar telephone sets.)

**Business Series Terminals Feature Comparison**

*Table 2.*

<table>
<thead>
<tr>
<th>Feature List</th>
<th>T7100</th>
<th>T7208</th>
<th>T7316E</th>
</tr>
</thead>
<tbody>
<tr>
<td>LCD display</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Integrated tilt display</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Deployment on Business Communications Manager and Norstar (all releases)</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>• buttons with LCD indicator</td>
<td>0</td>
<td>8</td>
<td>16</td>
</tr>
<tr>
<td>• buttons without LCD indicators</td>
<td>1</td>
<td>0</td>
<td>8</td>
</tr>
<tr>
<td># of line appearances</td>
<td>0</td>
<td>8</td>
<td>10</td>
</tr>
<tr>
<td># of programmable autodial buttons</td>
<td>1</td>
<td>8</td>
<td>24</td>
</tr>
</tbody>
</table>
**Feature List**

<table>
<thead>
<tr>
<th>Feature</th>
<th>T7100</th>
<th>T7208</th>
<th>T7316E</th>
</tr>
</thead>
<tbody>
<tr>
<td># of fixed buttons</td>
<td>5</td>
<td>7</td>
<td>10</td>
</tr>
<tr>
<td># of soft key buttons</td>
<td>0</td>
<td>0</td>
<td>3</td>
</tr>
<tr>
<td>Handsfree</td>
<td></td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Visual ringing indicator</td>
<td>√</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Audio control center (dedicated Headset and Mute buttons)</td>
<td></td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Volume bar</td>
<td>√</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Call log</td>
<td>√</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Intercom</td>
<td>√</td>
<td></td>
<td>√</td>
</tr>
<tr>
<td>Selective ringing tones / Discriminating ringing</td>
<td>√</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Automatic set relocation</td>
<td>√</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Multilingual capability</td>
<td>√</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>FWD / DND</td>
<td>√</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Wall mount capability</td>
<td>√</td>
<td></td>
<td>√</td>
</tr>
<tr>
<td>Support for Central Answering Position (CAP) Module</td>
<td>N/A</td>
<td>N/A</td>
<td>N/A</td>
</tr>
</tbody>
</table>

**Desktop Assistant Button Labeling Application**

The Desktop Assistant is a software application tool developed to support the new button labeling strategy on the Business Series Terminals. The Desktop Assistant tool supports users in quickly and easily labeling their T7100, T7208 and T7316E telephone sets.

This application allows end users to create customized labels for their sets. In the application, users select the set type they wish to label and are presented with an image of the set. Users then enter the text in the button label fields. For each button, users may select from font type, size and color and a background color. When the process is complete, users can print the labels on a black-and-white or color printer and may save the data file (*.ntl) for later modification or for sharing among users.

**Desktop Assistant Pro Administration Edition**

BCM features the Desktop Assistant Pro Administration Edition, a client application available for system administrators that can be downloaded from the BCM and used on a desktop. In addition to defining labels for printing for a set, users can accomplish administration of any set on any BCM system in the network. Only user preferences for the set can be programmed.
Key Features

- Labels T7100, T7208, T7316E only (does not label T7406, or M7XXX Series)
- Available in English, French and Spanish
- Supported on Windows 95/98/2000 and NT4.0.

Customers can purchase the application on CD (NTAB3320) or download it from [http://www.nortelnetworks.com](http://www.nortelnetworks.com) at no charge.

Business Communications Manager Accessories

- Analog Terminal Adapter (ATA-2)
- Nortel Audio Conferencing Unit (NACU)
- Central Answering Position (CAP)
- Station Auxiliary Power Supply (SAPS).

Analog Terminal Adapters

*Figure 24.*

Description

The Analog Terminal Adapter-2 (ATA-2) converts BCM digital interfaces to analog for communication with such analog devices as single line telephones, fax machines, modems and answering machines. Single line sets can interface with BCM system features, including Call Waiting, Call Forward and many more. The ATA-2 provides a means of connecting a single line set to the BCM system in either a long loop or off-premise extension configuration.
The ATA-2 supports data transmission speeds, up to and including 28.8 Kbps. (Note that the maximum data transmission rate is subject to the quality of the end-to-end channel and cannot be guaranteed.)

The ATA-2 is powered by a grounded AC power supply that is packaged with the ATA-2. CMS/CLASS feature interworking is not supported by any version of the Analog Terminal Adapter. There are separate ATA-2s for North America, Europe and Australia/New Zealand.

For high-density analog connectivity with a BCM, customers may make use of the Analog Station Module.

<table>
<thead>
<tr>
<th>Feature</th>
<th>ATA-2</th>
<th>Feature</th>
<th>ATA-2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Separate power supply</td>
<td>√</td>
<td>Priority Call</td>
<td>√</td>
</tr>
<tr>
<td>Alternate Line</td>
<td>√</td>
<td>Reach Through – timed Release</td>
<td>√</td>
</tr>
<tr>
<td>Call Forward</td>
<td>√</td>
<td>Restriction Override</td>
<td>√</td>
</tr>
<tr>
<td>Call Park</td>
<td>√</td>
<td>Ring Again</td>
<td>√</td>
</tr>
<tr>
<td>Call Pickup – Group or Directed</td>
<td>√</td>
<td>Saved Number Redial</td>
<td>√</td>
</tr>
<tr>
<td>Call Queuing/Waiting</td>
<td>√</td>
<td>Send Message</td>
<td>√</td>
</tr>
<tr>
<td>Camp</td>
<td>√</td>
<td>System Speed Dial</td>
<td>√</td>
</tr>
<tr>
<td>Centrex/PBX Reach Through</td>
<td>√</td>
<td>Tones On</td>
<td>√</td>
</tr>
<tr>
<td>Conference</td>
<td>√</td>
<td>Transfer</td>
<td>√</td>
</tr>
<tr>
<td>Hold – Exclusive</td>
<td>√</td>
<td>Trunk Answer</td>
<td>√</td>
</tr>
<tr>
<td>Hold – Public</td>
<td>√</td>
<td>Voice Call</td>
<td>√</td>
</tr>
<tr>
<td>LNR</td>
<td>√</td>
<td>Messaging – Access via DN</td>
<td>√</td>
</tr>
<tr>
<td>Line Pool Selection</td>
<td>√</td>
<td>Messaging – Mailbox access</td>
<td>√</td>
</tr>
<tr>
<td>Link</td>
<td>√</td>
<td>Messaging – Leave message</td>
<td>√</td>
</tr>
<tr>
<td>Page – General</td>
<td>√</td>
<td>SMDR Account Codes</td>
<td>√</td>
</tr>
<tr>
<td>Page – External</td>
<td>√</td>
<td>CDR Account Codes</td>
<td>√</td>
</tr>
<tr>
<td>Page – External/Internal</td>
<td>√</td>
<td>CDR Account Codes</td>
<td>√</td>
</tr>
<tr>
<td>Page – Internal</td>
<td>√</td>
<td>Toll Restriction Improvements</td>
<td>√</td>
</tr>
<tr>
<td>Privacy Control</td>
<td>√</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
NACU

*Figure 25.*

**Description**

Designed in partnership with Polycom, the NACU offers superior teleconferencing by using three microphones to provide 360 degrees of voice coverage. The Conferencing Unit is a full duplex handsfree unit. This feature allows voice to be heard and picked up at the same time, providing faster response time and eliminating conversation “collisions” and losses. The unit has a Feature Key that allows access to many of the same features found on the Business Series Terminals. The NACU connects to the BCM system via a station port. The NACU supports both analog and digital.

**Station Auxiliary Power Supply (SAPS)**

*Figure 26.*
Description

The Station Auxiliary Power Supply (SAPS) extends the loop length between a set or terminal and the BCM system from 1,000 to 2,600 feet. A dedicated cable must be used to connect the two locations. One SAPS powers up to three sets at 2,600 feet or two CAP modules (which do not have to be connected to the same T7316E set) at 1,000 feet. One SAPS is required to power every two CAP modules.

IP Telephones

Now users can enjoy the next-generation features of VoIP, plus all of the carrier-class reliability and ease of use of a traditional telephone. Nortel offers four superior desktop models, along with two innovative software-based solutions that bring VoIP to a user’s computer or PDA. The Nortel Internet telephones include:

- **IP Phone 2001** – offers a desktop solution with a broad range of features. Users immediately feel comfortable with this new phone because it operates like a traditional phone. This shortens the learning curve and reduces the need for training during the transition to VoIP.

- **IP Phone 2002** – is designed for office professionals and technical specialists. This multiline phone offers an integrated LCD display screen and is well-suited for moderate call volumes.

- **IP Phone 2004** – is ideal for managers, executives and office administrators. This multiline phone features a large LCD display screen capable of displaying a maximum amount of information and is well-suited for high call volumes. The Internet telephones operate seamlessly across our entire range of IP-enabled platforms, offering a complete, full-featured VoIP solution unmatched by any other vendor in the industry.

- **IP Softphone 2050** – is suitable for a broad range of workplaces and mobile users. This software-based solution transforms a user’s laptop or desktop PC into a converged voice/data communications platform. The Nortel Internet telephones provide support for a wide range of today’s high-value ebusiness applications, including Unified Messaging and Symposium* Call Center services.

- **Mobile Voice Client 2050** – extends the functionality of the IP Softphone 2050 to a user’s PDA. Voice and data communications are secure for external-to-campus employees with support for IPSec VPN tunneling with Nortel Communication Servers delivering support of up to 650 telephony features. Converges business applications onto a single, mobile device delivering full telephony capabilities to Pocket PC-based PDAs, thus eliminating the need for separate voice and data communication devices.

- **IP Phone 2007** – The IP Phone 2007 is Nortel touch-screen IP color display phone. It offers a fully pixel-based 5.7” diagonal CSTN display offering new and exciting multimedia capabilities including presentation of Web content and video frames from application servers such as the NET6 Application Gateway. Additionally, the phone will offer future support for Bluetooth headsets as well as USB defined peripheral capabilities (keyboard, mouse, thumb drive etc).
This rich, future-proof feature set will evolve to support advanced services such as voice-activated dialing and corporate and personal directory services. In addition, VoIP reduces costs by putting voice signals on standard network cabling, as opposed to having a separate cabling system dedicated to voice. By eliminating the need for separate wiring to support voice and data transmission, customers can capitalize on the cost economies provided by a simplified wiring system within the enterprise.

Desktop Solutions

Nortel IP Phone 2001

*Figure 27.*

The Nortel IP Phone 2001 offers a desktop solution with a broad range of features. Users immediately feel comfortable with this new phone because it operates like a traditional phone. This shortens the learning curve and reduces the need for training during the transition to VoIP.

The IP Phone 2001 provides the following unique features:

- Suite of rich and reliable telephony features
- Small device footprint
- Five fixed keys & navigation keys
• Hold, Goodbye, Volume, Services, Messages
• Up/down navigation keys for LCD display
• Four soft keys
• Visual ringing alerter / Message Waiting Indicator (MWI)
• LAN or AC local power options
• Integrated speaker in “Listen Only” mode


Figure 28.

The IP Phones 2002 and 2004 offer a desktop solution with a broad range of features. Users immediately feel comfortable with the new phones because they operate like traditional phones. This shortens the learning curve and reduces the need for training during the transition to VoIP.
The IP Phone 2007 is a displayed-based desktop telephone that deliver web-based content for an interactive user experience that results in more accurate decision making. The IP Phone 2007 is a touch-screen IP color display phone that offer a fully pixel-based 5.7” diagonal CSTN display with new and exciting multimedia capabilities, including presentation of web content and video frames from application servers such as the NET6 Application Gateway. Additionally, the phone will offer future support for Bluetooth headsets as well as USB-defined peripheral capabilities (keyboard, mouse, thumb drive, and so on).

The Nortel IP Phones feature an internal 10/100 Layer 2 switch that enables the user’s phone and PC to use the same network port for voice/data connectivity.

The Nortel IP Phones provide the following unique features:

- High-fidelity speakerphone facilitates group conference calls and delivers crystal-clear, handsfree communication
- A pixel-based LCD display provides a window into a full range of personal productivity tools
- Self-labeling programmable keys eliminate the need for paper inserts or an additional labeling application and offer improved visibility and clarity
- Adjustable LCD contrast supports viewing in a wide range of user environments
- BCM provides easy access to features on the IP Phones 2001, 2002, 2004, & 2007 by allowing users to scroll through features on the LCD, launch a feature from a programmable feature list and program additional buttons on the sets.
- Integrated 802.3af Power over LAN; virtual LAN (VLAN) tagging
Software Solutions

Nortel IP Softphone 2050

Ideal for mobile users, the Nortel IP Softphone 2050 is a software-based solution that loads directly onto your laptop or desktop PC. Once a Nortel headset is connected to the USB port, the IP Softphone 2050 delivers virtually identical functionality as the IP Phone 2004 desktop phone. The IP Softphone 2050 is also particularly useful for contact center agents who need a handsfree solution to do their jobs more effectively.

Mobile workers can simply plug their laptop into a network port at a shared office location, snap in a USB headset and function as if they were in their own office. And, because the network recognizes them as unique users, all of their phone features will be available to them, including outbound Caller ID. Calls can be placed on both the internal and external network, providing a truly portable and practical solution.

The IP Softphone 2050 provides the following unique features:

- Macro functions transform lengthy operations into a single-digit action
- Nortel USB Audio Kit support offers wireline-quality voice performance
• Local directory support imports or reads Symantec ACT, Microsoft Outlook and LDAP databases, for seamless directory integration.

**IP Softphone 2050 Diagnostic Tool**

In addition to the Internet telephones, BCM includes the IP Softphone 2050 Diagnostic Tool. This diagnostic tool is a client application that can be downloaded from the BCM and used on a desktop in conjunction with an IP Softphone 2050 client on that desktop. It facilitates the quick resolution of any issues with the IP Softphone 2050, providing both IP and multimedia information.

**Internet Telephones Benefits**

• Support connectivity to any Nortel VoIP enabled platform, including BCM and Communication Server 1000
• Transnetwork support increases employee productivity by providing users and network managers with a common set of telephones across the entire network
• Internal voice/data switch prioritizes voice traffic to ensure high-quality speech and reduces costs by conserving wiring closet ports and eliminating the need for separate cable drops to the desktop
• Automatic firmware upgrade ensures top performance and streamlines maintenance, reducing the need for site visits
• DHCP-enabled Internet telephones simplify network administration by providing centralized, automated IP address management

**Multiple Platform Support**

The Nortel IP Phones are supported by multiple Nortel communication systems, including BCM, and Communication Server 1000. This industry-leading platform interoperability facilitates growth and offers seamless migrations across customer premises and carrier-based solutions. All Internet telephones offer a rich suite of business features designed to meet current and emerging user requirements and international icon-based characters. These features, in addition to multilanguage support, make the Internet telephones ideal for worldwide use.

**Integrated Switched Ethernet Connection**

Users can connect their desktop Internet telephone and their PC to the network on a single port. The IP Phones feature a built-in 10/100 Base T Layer 2 switch that splits the network Category 5 cable into separate feeds,
providing an additional RJ-45 port to connect a user’s PC. Giving fixed, hardware-based priority to the voice port, the internal Ethernet switch ensures that high-quality voice service is always available.

**Reliable LAN Power Options**

To ensure continuous phone service in mission-critical environments, the IP Phones can receive their power over the network cabling. The Nortel Ethernet Switch 460-24T-PWR delivers power over the unused pairs of standard Category 5 UTP cables, eliminating the need for the phones to be connected to a power supply at the desktop.

Power is supplied on an as-needed basis, thanks to the sophisticated, software based load and fault-sensing algorithm used by the Ethernet Switch 460-24T-PWR. The unit fits into a standard 19-inch wiring closet rack and provides a cost-effective way to centralize power to the IP Phones. This approach delivers carrier-grade reliability by enabling redundant power resources located at the wiring closets to provide emergency backup power to Nortel Internet telephones located across the network.

**Dynamic Host Configuration Protocol (DHCP) Addressing**

Easy to set up and configure, the IP Phones deliver an innovative solution that enables users to connect anywhere on the network without intervention by a network administrator, enabling management staff to focus on more complex, mission-critical responsibilities.

Whether users relocate their phone down the hall or across the globe, the service comes up in the new location exactly as if they were sitting in their own office, even though it might be thousands of miles away.

**Prioritizing Network Traffic**

Now companies can make their network’s priorities mirror those of their business. By installing the Nortel BayStack 460, a business’ network becomes an intelligent partner that uses Layer 2/3 packet classification to prioritize business-critical traffic, helping them get the most out of their existing infrastructure.

Latency-sensitive applications and mission-critical users will receive platinum-level service, while less urgent traffic is allocated bandwidth on a lower-priority basis. Prioritization is especially important for businesses that need to support mission-critical IP applications, including VoIP, but do not want to incur higher costs by overprovisioning the network to ensure bandwidth availability.
Universal Features

With over 100 years of experience in creating phones designed to improve employee productivity, reduce operational expenses and improve customer service, Nortel is the industry leader in telephony solutions. All of the IP Phones offer the following features:

- Intuitive navigation cluster provides fast menu, sublist and call log scrolling, as well as one-touch dialing and quick access to system features
- Message waiting/visual ring indicator offers visual notification of incoming calls and messages
- Voice compression optimizes bandwidth and audio quality requirements
- Audio control center enables users to toggle quickly between the handset or headset and the speakerphone without audio interference
- Volume bar provides fingertip control of audio and ringer volume settings and LEDs clearly display handset/headset/speakerphone/mute settings
- Local tone generation conserves valuable network bandwidth
- Dynamic IP addressing with a standard DHCP server offers a flexible, simplified solution for handling adds, moves and changes, reducing management costs
- Microsoft TAPI-compliant interface operates seamlessly with Messaging and Personal Call Manager – this offers users onscreen displays of call logs and directories and also provides support for drag-and-drop dialing
- User-friendly design supports the full range of potential users, including disabled users who require hearing aids.

Future-Proof

Nortel actively participates in defining standards-based solutions that support the broad deployment of VoIP across enterprise environments. The IP Phones are designed for flexibility and ease of upgrades and will support firmware upgrades as features and industry standards continue to evolve.

When used in conjunction with a terminal proxy server, the IP Phones behave like standards-compliant MGCP and H.323 devices, enabling Nortel platforms to be distributed across the network while seamlessly interworking with similar standards-based gateways.
IP Set Features and Programming

The large LCD display of the IP Phones has many uses. The display shows a list of telephony features that users can highlight and then activate from the display, so that a user does not have to refer to the feature code list or memorize the feature codes as a means of activating features. BCM provides a default list of the most common features, which can be displayed on the IP Phone LCD by using the Services button:

- Hot desking
- Last Number
- Conference/Transfer
- Do Not Disturb
- Call Forward
- Page
- Background Music

- Call Park
- Call Pickup
- Voice Call
- Speed Dial
- Message Send.

This list can be changed by the user to suit the user’s needs. This feature scrolling is available for IP Phones 2002, 2004, & 2007, along with IP Softphone 2050.

BCM allows the user to program additional buttons on the IP set. On the IP Phone 2004 and IP Softphone 2050 sets, six additional buttons are programmable for a total of 12 programmable buttons. On the IP Phone 2002 set, five extra buttons for a total of 9 buttons are programmable. The IP Phone 2001 has four soft keys.

BCM also allows the hot desking of IP sets, which allows one IP set to adopt the configuration of another set on a BCM system. Once an IP set is programmed for this, all the button programming, line appearances and feature buttons from the original set appear on the hot desk set and calls appear on this set as they would on the original. This password-protected feature can be disabled from either set once it is no longer needed. Hot desking of IP sets supports mobile workers and helps them maintain their productivity while away from their home office.
Introduction

Hardware

> Telephony

Data Capabilities

Messaging

Voice Over IP (VoIP)

Voice Networking

Call Center

Interactive Voice Response (IVR)

Mobility Solutions

Computer Telephony Integration (CTI)

Virtual Private Networks (VPN)

System Management and Software Options

Glossary

Index
Telephony

The BCM supplies feature-rich telephony to the small and medium-sized business, offering the following standard features:

- PBX/Key system functionality
- Fully integrated messaging
- Automated Attendant with Custom Call Routing (CCR)
- Computer Telephony Integration (LAN CTE)

Additional BCM features are available when CLASS/CMS (Custom Local Area Signaling Services/Call Management Services) is supplied on a business’ telephone line.

BCM offers a number of ways to connect to network service providers, including:

- Analog Loop Start
- T-1 (North America); E1 (Europe)
- Basic Rate Interface (BRI)
- Primary Rate Interface (PRI).

The faster call setup and tear-down capability of ISDN can provide significant benefits to businesses, particularly in call center applications. In addition, ISDN network services provide a number of features that can enhance the usefulness of the BCM. These include such features as Calling Name and Number Delivery (CLID).

CLASS/CMS Features

Custom Local Area Signaling Services/Call Management Services (CLASS/CMS) is a unique set of features and protocols. One of the prime features is the delivery of call-related information from a central office switch to a customer’s premises. This information is delivered between the first and second ring. If the call is answered prior to delivery, the data is permanently lost.

One of the most popular CLASS/CMS features is Calling Name and Number Display. Besides the obvious benefit of letting called parties know who is calling before they pick up the phone, there are some significant marketing and sales benefits that can be derived from this information.

The following BCM features are available when CLASS/CMS is supplied on a business’ telephone line:
• Call Information feature
• Calling Name and Number Display
• LOGIT feature (manual logging)
• Long-distance Indicator
• Caller Log
• Auto Bump On/Off
• Automatic Redial
• Caller’s Name/Number
• Logging Options
• Long-distance Indicator
• Optional Password Protection
• Repeat Call Counter
• Automatic Redial
• Central office-based messaging visual message notification.

The following CO-based CLASS/CMS features also interact with BCM for enhanced operation:

• Automatic Callback (AC)
• Automatic Recall (AR)
• Calling Number Delivery Blocking
• Customer Originated Trace
• Distinctive Ringing/Call Waiting
• Selective Call Acceptance.

Note: Feature names and availability will vary from region to region.

Call Detail Recording (CDR)

BCM Call Detail Recording (CDR) records and reports call activity. Call records can be displayed in real-time or copied to a local PC desktop and processed via third party call accounting packages for billing or analysis of long-distance costs.
CDR allows companies to record information about all incoming and outgoing calls. Further, it allows businesses to print recorded information in reports. CDR can also provide information on incoming calls as the events occur and record this information in a real-time call record.

Call Detail Recording provides the following information:

- Date and time of the call and digits dialed
- The originating and terminating line or station set
- Whether an incoming call was answered
- Elapsed time between origin of a call and when it was answered
- Whether a call was transferred or put on hold
- Call duration
- Calls associated with account codes
- Incoming call Calling Line Identification (CLID) information
- Bearer capability of the line in the call
- Hospitality records for room occupancy status
- Real-time records for ringing, DNIS, answered, unanswered, transferred and released events for incoming calls with CLID information and hospitality room occupancy status.

BCM CDR delivers the following information in the form of CLID reports:

- Custom Local Area Signaling Services (CLASS)
- Call Management Services (CMS)
- Automatic Number Identification
- Dialed Number Identification Services (DNIS).

This information is only available if the appropriate BCM hardware is installed and the service is available from the company’s public telephone company.

**CDR Enhancements**

The BCM supports a CDR Push feature and includes a file transfer mechanism that can send CDR data files (stored on the BCM) to a central server on a predefined schedule. The BCM also allows a central server to pull CDR files on a demand, rather than scheduled, basis. These capabilities are particularly valuable in a large network of BCMs, where sending significant amounts of data in real-time would not be practical or effective. By
using the CDR Push or Pull features, the central server can receive and process the CDR data in a more manageable fashion.

The BCM provides call detail records that can be used as an input to billing systems by accounting departments, to judge traffic loads by system administrators and for other purposes. These records are processed remotely from the BCM.

Customers can use information collected by CDR to:

- Allocate telephone costs to departments or individuals
- Charge back telephone costs to billable clients through account codes
- Determine whether the telephone system is being used efficiently
- Guard against abuse of the telephone system
- Provide immediate call information to database applications through real-time call records
- Store and clip
- Provide real-time via IP
- Track changes in room occupancy status.

BCM provides a “CDR Pull” mechanism that allows a central client machine, on its own schedule and up to its own capacity, to contact concurrently (or sequentially) any number of BCMs and fetch call data. BCM allows the use of FTP and other TCP/IP-based standard tools to set up and transfer the CDR files and maintains platform independence.

**Optional capture of CDR dialed digits**

This feature provides a pull-down menu option within Unified Manager (CDR tree) to enable/disable the recording of dialed digits after the far-end has answered the call. This option will only be available on outgoing calls with “Answer Supervision”.

**Report Information**

CDR includes a number of parameters that businesses set based on the type of information they require in their reports.

BCM Call Detail Recording generates both Norstar and SL-1 report types. SL-1 report types are used when supplying reports to legacy commercial call accounting packages or equipment. Norstar reports are used for more detailed and concise call reports.
**Types of Calls Collected**

Customers can configure Call Detail Recording to collect:

- All calls (incoming/outgoing)
- Outgoing calls only
- Long-distance calls only
- All calls associated with account codes.

**Report Languages**

The default report language is English; however, CDR also supports other languages depending on the business’ market profile.

**Account Codes**

Account codes create a reference for tracking calls. For example, to track calls to a billable client, an assigned code is entered each time a call is placed to that client. Businesses can also associate account codes with a particular employee. Table 4 is an example of an account code list.

**Table 4.**

<table>
<thead>
<tr>
<th>Account code</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>11127</td>
<td>Patricia</td>
</tr>
<tr>
<td>37</td>
<td>Field Support</td>
</tr>
<tr>
<td>239</td>
<td>Joe</td>
</tr>
<tr>
<td>45</td>
<td>Modern Ways Limited</td>
</tr>
<tr>
<td>100</td>
<td>Long-distance</td>
</tr>
</tbody>
</table>

**Call Detail Recording Display**

The CDR client allows businesses to remotely monitor records as calls occur. To use Call Detail Recording Display, the business must connect to the BCM server and start the recording.
Call Detail Recording Record Security

The records from BCM Call Detail Recording are sensitive. Information such as communication among top executives and external companies, telephone banking passwords and long-distance PIN codes require protection from unauthorized access.

The BCM system administrator provides only authorized users with launch permission to the records.

Telephony Features

This section lists BCM telephony features and describes them.

An asterisk (*) denotes the described feature as a CLASS/CMS line feature and not a BCM feature.

Table 5.

<table>
<thead>
<tr>
<th>Features</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Access Control to Link, LNR (Last Number Redial), SNR (Stored Number Redial)</td>
<td>System security enhancement that allows system administrator the option to remove access to Link, LNR and/or SNR on a set-by-set basis</td>
</tr>
<tr>
<td>Accidental Disconnect Protection</td>
<td>If the receiver is accidentally dropped back into the cradle when answering a call, it can be retrieved within one second</td>
</tr>
<tr>
<td>Administration &amp; Configuration Tree</td>
<td></td>
</tr>
<tr>
<td>Alpha Tagging</td>
<td>Displays the programmed system speed dial name for an incoming call if the CLID number matches.</td>
</tr>
<tr>
<td>Alternate Restrictions</td>
<td>Alters which calls can be made by changing dialing restrictions according to both time of day and day of week</td>
</tr>
<tr>
<td>Analog Station Module Recognition</td>
<td></td>
</tr>
</tbody>
</table>
| Answer Groups (also called Answer Button or Answer DN) | A telephone button with an indicator that is used to monitor ringing calls at another set  
Let users answer calls at the monitoring set by pressing the active button  
Also functions as a non-user-programmable DSS key for the relevant DN. Pressing that set key when the key is idle results in a DSS intercom call being made to the assigned DN. |
<p>| Auto-Answer                                   | Used with DID, DISA and E&amp;M trunks, calls are automatically answered by the BCM, bypassing the attendant; a caller enters the digits for routing to a specific set or line pool access |</p>
<table>
<thead>
<tr>
<th>Features</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Auto Bump On/Off</td>
<td>When a Call Log becomes full, Auto Bumping – “On,” will cause the oldest entry previously viewed to be deleted and the new call to be logged&lt;br&gt;When “Off,” the Call Log will not log new calls</td>
</tr>
<tr>
<td>Auto Dial (Internal and External)</td>
<td>Allows users to program internal or external numbers onto memory buttons for one-button dialing access</td>
</tr>
<tr>
<td>Automatic Callback (AC)*</td>
<td>Automatically redials the last outgoing number dialed&lt;br&gt;If the number is busy, the central office will use “Ring Again” to monitor the line. When it is free, the caller will hear special ringing and the number and/or name of the called party will be delivered to the LCD. Multiple busy lines can be monitored for up to 30 minutes&lt;br&gt;Can be programmed as an external autodial for one button convenience</td>
</tr>
<tr>
<td>Automatic Daylight Savings time</td>
<td>The system clock automatically falls back one hour on the last Sunday of October at 2:00 a.m. and automatically advances one hour on the first Sunday of April at 2:00 a.m.&lt;br&gt;Can be deactivated where not applicable</td>
</tr>
<tr>
<td>Automatic Hold During Incoming Page</td>
<td>Each set can automatically and immediately hold its active call when an incoming page is being presented to that set, and return the call to active when the page terminates.</td>
</tr>
<tr>
<td>Automatic Line Selection</td>
<td>When answering incoming ringing calls, BCM automatically selects the longest ringing line first&lt;br&gt;Ringing incoming calls are automatically connected by lifting the receiver, pressing Handsfree, or using Call Queuing</td>
</tr>
<tr>
<td>Automatic Number Identification (ANI)</td>
<td>Delivers the calling line number (T-1 Specific)</td>
</tr>
<tr>
<td>Automatic Recall (AR) *</td>
<td>Works the same as Automatic Callback (AC)&lt;br&gt;Applies to the last incoming call received</td>
</tr>
<tr>
<td>Automatic Route Selection (ARS)</td>
<td>Automatically selects the preprogrammed long-distance carrier based on the dialed digits, time of day and day of week</td>
</tr>
<tr>
<td>Automatic Set Relocation</td>
<td>Sets moved to a different location retain all custom programming</td>
</tr>
<tr>
<td>Auxiliary Ringing</td>
<td>A set’s headset jack can send ringing tones via an amplifier to an external loud ringer connected to the set</td>
</tr>
<tr>
<td>Background Music</td>
<td>Users can listen to music (customer supplied) through the set’s speaker when the set is idle</td>
</tr>
<tr>
<td>Brazil Localization</td>
<td>Provides support for a Brazil profile, local ISDN protocol and support for Portuguese language on voice messaging prompts and end-user documentation.</td>
</tr>
<tr>
<td>Busy Lamp Indication</td>
<td>Indicates that a user’s set is busy</td>
</tr>
</tbody>
</table>
## Features

<table>
<thead>
<tr>
<th>Features</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Button (Key) Inquiry</td>
<td>Allows users to check the programming on memory buttons</td>
</tr>
<tr>
<td>Call Display when Busy</td>
<td>Call display shows the name of a calling party while the user is on a call</td>
</tr>
<tr>
<td>Call Duration timer</td>
<td>Temporarily displays the length of the last or current call so a user can record it.</td>
</tr>
<tr>
<td>Call Forward All Calls</td>
<td>Sends all calls to another set</td>
</tr>
<tr>
<td>Call Forward No Answer</td>
<td>Transfers an unanswered call to another DN after a preset number of rings</td>
</tr>
<tr>
<td>Call Forward On Busy</td>
<td>If a set is busy, sends calls immediately to another DN</td>
</tr>
<tr>
<td>Call Forward Override</td>
<td>When a set is on Call Forward, the &quot;Forward To&quot; set can still call the &quot;Forwarded&quot; set to relay important messages</td>
</tr>
<tr>
<td>Call Forward – Selective</td>
<td>Allows users to transfer a call to the Prime set by pressing the &quot;Do Not Disturb&quot; button when a central office line is ringing</td>
</tr>
<tr>
<td>Call Information</td>
<td>Displays information about incoming calls</td>
</tr>
<tr>
<td></td>
<td>For external calls, CMS is required and it displays the caller's name, telephone number and the line name</td>
</tr>
<tr>
<td></td>
<td>For internal calls, it displays the name and the internal number</td>
</tr>
<tr>
<td>Call Log</td>
<td>Allows users to enter Call Log to view stored information including:</td>
</tr>
<tr>
<td></td>
<td>• Caller’s name and/or number (if delivered from the central office)</td>
</tr>
<tr>
<td></td>
<td>• Date and time</td>
</tr>
<tr>
<td></td>
<td>• Answered Call Indication</td>
</tr>
<tr>
<td></td>
<td>• Repeat Call Counter</td>
</tr>
<tr>
<td>Call Log – Optional Password</td>
<td>Allows users to enable password protection of their call log</td>
</tr>
<tr>
<td>Caller Log: 600</td>
<td></td>
</tr>
<tr>
<td>Call Log – Passive</td>
<td>Call logging is enabled once the call log space is assigned. In case of re-routed calls, the logging is carried out for explicitly transferred calls only.</td>
</tr>
<tr>
<td>Calling Name &amp; Number Display</td>
<td>Allows users to view the name and number of the incoming caller before answering and during the call</td>
</tr>
<tr>
<td></td>
<td>The calling number is also stored in the Call Log (Requires CLASS/CMS, ISDN-BRI or ISDN-PRI)</td>
</tr>
<tr>
<td>Features</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
</tbody>
</table>
| Calling Name & Number          | Allows users to prevent delivery of calling name and/or number when placing a call                                                                                      
| Display Blocking               | For outgoing calls, BCM supports one button convenience when the user wishes to “block” their number and/or name (requires CLASS/CMS, ISDN-BRI or ISDN-PRI)  
|                                | On incoming calls, presents users with “Private Name” and/or “Private Number” when receiving a “blocked” call                                                                                                             |
| Call Park – Linear/Round Robin | Allows system administrators to choose linear or round robin call park codes                                                                                      
|                                | With linear codes, the system assigns the first free call park code to the call. This means that the first few call park codes will be used most frequently. In busy environments this can cause confusion when calls have been hung up or callback and within moments a new call is parked using the same code  
|                                | Round robin call park codes are assigned sequentially until the maximum number of codes is reached before starting again at the first code  
<p>|                                | Number of Call Park Codes: 25                                                                                                                                         |
| Call Park (with Callback)      | Automatically places an active call on hold and assigns it another code so it can be retrieved from another set                                                                                                           |
| Call Pickup Directed           | Allows users to answer a ringing call at any other set by dialing the ringing set’s intercom number                                                                                                           |
| Call Pickup Group              | Allows users to answer any call ringing at another set within the pickup group                                                                                                                  |
| Call Pickup Trunk Answer       | Allows users to answer a ringing external call at any other set                                                                                                                  |
| from any station               |                                                                                                                                                                                                                                                                           |
| Call Queuing                   | Answers the next available call, but gives priority to the longest waiting external call                                                                                                                   |
| Callback                       | Automatically returns unanswered parked or transferred calls to the originating set after a preset number of seconds                                                                                       |
| Calling Line ID on Multiple    | Calling Line ID (CLID) is displayed on up to 30 sets have line appearance for an incoming external call on a physical or target line.                                                                                     |
| Line Appearances               |                                                                                                                                                                                                                                                                           |
| Camp On (Call Waiting)         | An external call waits at a busy set, making alerting tones, until answered or Callback returns it to the originating set                                                                                          |
| Class of Service (COS)         | Controls the BCM features and lines available when a call is placed within the system or remotely; it can be associated with a line, a set or a Class of Service (COS) password                                                                                           |</p>
<table>
<thead>
<tr>
<th>Features</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Class of Service (COS) Password</td>
<td>A six-digit code that lets users switch from their current class of service to one that lets them dial a number prohibited by their current class of service. Is used when DISA access is controlled with passwords. Includes an access package that defines the set of line pools a user may access and provides access to remote paging capabilities. BCM can retain up to 100 six-digit COS passwords. In addition, a remote caller can change the Class of Service of an incoming call by dialing the DISA DN and entering a COS password. <em>Number of COS Passwords: 100</em></td>
</tr>
<tr>
<td>Compression of Feature Codes</td>
<td>Reach-through codes (run/stop, programmed release, pause) can be compressed to use less digit space in Autodial or Speed Dial programming modes.</td>
</tr>
<tr>
<td>Conference (Three-Person)</td>
<td>Creates a three-person call with two other internal or external parties. Is easily set up with LCD prompts; Automatic Hold protects the first call from being accidentally cut off. Flash hook (switch hook flash) during a conference call. TAPI application dialing and flash hook during a conference.</td>
</tr>
<tr>
<td>Consultation Hold</td>
<td>Allows users to put a call or conference on hold to consult with others on another line; held parties can still talk to each other.</td>
</tr>
<tr>
<td>Coordinated Dialing Plan</td>
<td>Allows administrator to program calls to route over a network based on destination codes.</td>
</tr>
<tr>
<td>Customer Originated Trace *</td>
<td>Allows users to send the number of the last incoming call to the telephone company. This includes calls where the name and/or number have been blocked. Note: The user does not receive the number of the caller. Security procedures will vary with different telephone companies.</td>
</tr>
<tr>
<td>Delayed Ring Transfer</td>
<td>Automatically transfers incoming calls to a “prime set” after a preset number of rings.</td>
</tr>
<tr>
<td>Dial “0” Station (Dial “X”)</td>
<td>Designated receptionist set that can be reached from any other set in the system by pressing the intercom key followed by the designated digit.</td>
</tr>
<tr>
<td>Dial Intercom</td>
<td>Allows users to quickly call co-workers internally by pressing the Intercom button and dialing the intercom number.</td>
</tr>
<tr>
<td>Dialing Filters</td>
<td>A maximum of 100 line, set or line/set filters provide virtually unlimited flexibility in programming dialing restrictions and exceptions.</td>
</tr>
<tr>
<td>Features</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------------------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Dialing Modes</td>
<td>A user can select from the following dialing mode options:</td>
</tr>
<tr>
<td></td>
<td>Standard: Lets a user choose a line and dial a call using either the receiver or handsfree</td>
</tr>
<tr>
<td></td>
<td>Automatic: Pressing a dial pad button will automatically select the set’s prime line, thus saving time</td>
</tr>
<tr>
<td>Dialing Mode: Pre-Dial</td>
<td>A person enters, checks and edits a number before selecting a line</td>
</tr>
<tr>
<td>Dial Mode for Lines</td>
<td>Temporarily changes set from pulse to tone mode to signal external systems and devices</td>
</tr>
<tr>
<td>DID Template</td>
<td>At System Start, template choices include a DID Template which automatically assigns target lines and received numbers as the set DN</td>
</tr>
<tr>
<td></td>
<td>When a system is expanded these assignments are preserved (Typically programmed by the installer)</td>
</tr>
<tr>
<td>Direct Dial – Flexible Digits</td>
<td>A systemwide digit used to call a direct dial set can be any digit from 0-9</td>
</tr>
<tr>
<td>Direct Dial – Multiple Attendants</td>
<td>Single digit access to an attendant. There may be up to five direct dial sets in the system, but each extension is assigned to a single direct dial set</td>
</tr>
<tr>
<td>Direct Inward System Access (DISA)</td>
<td>Allows remote users to dial directly into the BCM to access features. Users hear a stuttered dial tone and must enter a Class of Service (COS) password to gain access to the system</td>
</tr>
<tr>
<td>Disconnect Supervision</td>
<td>After an external call disconnects, drops the line immediately</td>
</tr>
<tr>
<td>Discriminating Ringing at Set</td>
<td>Different rings for internal and external calls allow users to easily distinguish between call types</td>
</tr>
<tr>
<td>Distinctive Ringing/Call Waiting*</td>
<td>A user hears special ringing or call waiting tones if the caller is included in a user-specified list of numbers</td>
</tr>
<tr>
<td>Do Not Disturb (DND)</td>
<td>Incoming calls will not ring at a set, but the LCD line indicator will continue to flash as calls are forwarded to the prime set</td>
</tr>
<tr>
<td>Do Not Disturb (On Busy)</td>
<td>Internal and private network callers hear a busy tone instead of ringing while the user is on a call</td>
</tr>
<tr>
<td></td>
<td>Transfers external callers to the Prime set for answering</td>
</tr>
<tr>
<td></td>
<td>The line indicator for an external incoming call flashes, but the phone does not ring</td>
</tr>
<tr>
<td>Enhanced Call Restrictions and Overrides</td>
<td>Maximum Number of Dialing Filers: 100</td>
</tr>
<tr>
<td>Enhanced Trunking Connectivity Private Network – E&amp;M (tie) Trunk Connectivity</td>
<td>E&amp;M Type II trunks can be connected to a BCM via Norstar Trunk Modules or via T-1 channels interfacing the FEM MBM to create a private network between locations</td>
</tr>
<tr>
<td></td>
<td>For each system within the network, the length of directory numbers (DNs), line pools and line pool access codes are the same</td>
</tr>
<tr>
<td>Features</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Enhanced Trunking Connectivity Public Network</td>
<td>DID (Direct Inward Dialing) trunks let incoming callers bypass the attendant and be directly routed to a target line.</td>
</tr>
<tr>
<td>– DID Trunk Connectivity</td>
<td>夾oring                                                                atched</td>
</tr>
<tr>
<td>Enhanced Trunking Connectivity Public Network</td>
<td>Remote access (with or without DISA) provides off-site remote access to BCM private or public network facilities, which avoids public network toll costs.</td>
</tr>
<tr>
<td>– Remote Access</td>
<td>夾oring                                                                atched</td>
</tr>
<tr>
<td>Executive Busy Override (Priority Calls)</td>
<td>Allows users within a BCM to force a voice connection to busy set or one on &quot;Do Not Disturb&quot; anywhere in the system.</td>
</tr>
<tr>
<td>External Calls on Intercom Keys</td>
<td>Program lines to ring on an intercom key.</td>
</tr>
<tr>
<td>External/Network Transfer</td>
<td>Allows the user to transfer calls over the public or a private network.</td>
</tr>
<tr>
<td>External Line Access</td>
<td>Allows users to directly access outside lines by buttons on individual phones or indirectly by a line pool.</td>
</tr>
<tr>
<td>Feature Access Key</td>
<td>Allows users to program any feature code onto a memory button.</td>
</tr>
<tr>
<td>Flexible Call Restrictions and Overrides</td>
<td>Call restrictions and overrides can be applied to individual lines and/or sets, but can be overridden with passwords.</td>
</tr>
<tr>
<td>Flexible Numbering Plan – Changing DN Length</td>
<td>The length of the Directory Number (internal number) can be from two to seven digits. All DNs in a system must be of the same length.</td>
</tr>
<tr>
<td>Global Analog Trunk Module</td>
<td>This redesigned module (4 and 8 port versions) replaces the current CTM4 and CTM8 and supports downloadable global analog profiles in North America, UK and Australia. With this feature, multi-national customers can standardize on a single platform and PSTN interface.</td>
</tr>
<tr>
<td>Group Listening</td>
<td>Allows users to hear an incoming voice on both handset and speaker, while an outgoing voice occurs only through the handset.</td>
</tr>
<tr>
<td>Group Set Copy</td>
<td>Allows the system programmer to copy data from one set to a range of DNs Two options are provided: copy from a set to all like sets, or copy from a set to all like sets within a specified range (e.g. copy data from a T7316E to all T7316Es within the range) Copying can be done for a particular subheading of programming or to duplicate all or a portion of programming for a set</td>
</tr>
<tr>
<td>Handsfree Answerback</td>
<td>Internal voice calls automatically turn on the set microphone so users can reply without touching the set. (Not available with the T7100)</td>
</tr>
<tr>
<td>Handsfree – Automatic</td>
<td>Allows users to program the set microphone and speaker to automatically turn on every time a call is answered or placed (Not available with the T7100)</td>
</tr>
<tr>
<td>Hold – Automatic</td>
<td>Automatically places an active line on hold if the user forgets to press the Hold button before selecting a second line, an intercom or a Transfer button</td>
</tr>
<tr>
<td>Features</td>
<td>Description</td>
</tr>
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</tr>
<tr>
<td>Hold – Exclusive</td>
<td>Your call can only be retrieved at the set where it was placed on hold</td>
</tr>
<tr>
<td>Held Line Reminder</td>
<td>External calls on hold play periodic reminder tones over the set speaker until the call is retrieved</td>
</tr>
<tr>
<td>I-Hold/U-Hold Indication</td>
<td>LCD line indicators will flash faster for held calls at the user’s own set than for calls on hold at other sets</td>
</tr>
<tr>
<td>Hospitality Feature set</td>
<td>BCM includes a set of three features that are applicable to the hospitality industry: Alarm Feature – Alarm clock operation on Business Series Terminals Room Occupancy -This feature allows the administrator to set dialing restrictions to a room, so that various levels (vacant, basic, mid and full) of call access are available to that room. Room Condition – This feature provides setting and querying the serviced condition (service done or service required) for the room</td>
</tr>
<tr>
<td>Hot Line</td>
<td>Allows users to program a set to call a specific internal or external number whenever they lift it or press a handsfree button</td>
</tr>
<tr>
<td>Hunt Groups</td>
<td>Enable single DN to call a group of sets Three hunting modes are available: broadcast, sequential and rotary. All Business Series Terminals, Digital Mobility sets, Attendant Consoles and 2500 analog sets can be assigned to a hunt group Silent Monitor for Hunt Groups. Maximum Number of Groups: 24 Number of Members per Group: 40</td>
</tr>
<tr>
<td>Language Choice</td>
<td>Allows users to select an alternate language for their Business Series Terminals: English, French or Spanish (The same system may have multiple languages active simultaneously)</td>
</tr>
<tr>
<td>Last Number Redial</td>
<td>Allows users to redial the last externally dialed number Number of Digits: 24</td>
</tr>
<tr>
<td>Line Assignment (Set)</td>
<td>A maximum of eight lines can be assigned to any of the sets in the system</td>
</tr>
<tr>
<td>Line Names</td>
<td>Names can be programmed for incoming and outgoing lines</td>
</tr>
<tr>
<td>Line Pool(s)</td>
<td>Allows users to select a line from a pool of lines using an access code when several external lines are shared by a group of telephones</td>
</tr>
<tr>
<td>Line Pool(s) – Busy Status</td>
<td>When all lines in a Line Pool are busy, the associated LCD set indicator will turn on</td>
</tr>
<tr>
<td>Line Profile</td>
<td>Line settings programmed in Configuration and Administration will appear on the T7316E set display</td>
</tr>
<tr>
<td>Features</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
</tbody>
</table>
| Line Redirection    | Often referred to as Selective Line Redirection  
Incoming calls on one or more lines can be redirected on a BCM telephone to one or more locations outside the system complying with associated dialing filters  
Redirected calls cannot be answered from another set (Not available with T7100) |
| Line Selection      | Users can press idle or ringing lines manually to override the automatic line selection feature                                               |
| Link/Flash (Recall) | If the BCM is connected to a PBX or Centrex, a link signal can be used to access special features                                           |
| Listen On Hold      | Users on hold may work handsfree while waiting by pressing the hold button, replacing the handset and then reselecting the held line  
The call can now be monitored through the speaker                                           |
| Log Space (CLID)    | BCM provides a maximum of 600 Call Log spaces                                                                                                |
| Logging Options     | Allows users to determine which type of calls will be logged at a set (i.e. no one answered, unanswered by me, log all calls and no auto-logging) |
| LOGiT (Manual Logging)| If calls are not automatically logged, lets users manually log an incoming call after they answer it                                           |
| Long Tones          | Sends long DTMF tones to access devices                                                                                                      |
| Loss Package        | Compensates for Loop Start (analog) trunk quality  
Allows selection of appropriate loss/gain and impedance settings for each line  
The settings are based on the distance between location of the BCM and the service providing central office |
<p>| Message Leave (List)| Display messages (“Message for you”) are sent to other set displays requesting a callback                                                  |
| Message Waiting (List)| Allows users to automatically call back the person who sent “Message for you” to their display; they can cancel the message           |
| Move Lines          | Assigned lines are moved to different LCD memory buttons on the set (except Handsfree, Intercom or Answer buttons) or Enhanced CAP (Not available with the T7100) |
| Music/Tone/Silence On Hold | Allows external callers on hold within the system to listen to music (customer supplied), a periodic tone or silence, as preset by the System Administrator |
| Network Direct Dial | Allows users to dial one digit to reach a specific destination on either a public or a private network                                     |
| Night Service       | Outside calls that normally ring at the prime set can also ring at additional, preselected sets during preset times                         |
| Numbering Plan – Flexible | The length and sequence of digits needed to access other sets or outside lines can be controlled                         |</p>
<table>
<thead>
<tr>
<th>Features</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>On-Hook Dialing</td>
<td>Allows users to dial directly from the dial pad and speak using the handset or Handsfree button</td>
</tr>
<tr>
<td>Paging – Internal</td>
<td>Allows users to initiate a page or be paged internally through the set speakers</td>
</tr>
<tr>
<td></td>
<td>It is also easy to make announcements through the telephone speakers to a select group of users or to all sets</td>
</tr>
<tr>
<td>Paging Feature Enhancement</td>
<td>Page time-out is now programmable and a system-wide parameter can now administer the paging tone to be “on” or “off”</td>
</tr>
<tr>
<td>Paging – External</td>
<td>Allows users to make external paging announcements when BCM is connected to a user-supplied amplifier and speaker</td>
</tr>
<tr>
<td>Paging – External and Internal</td>
<td>Allows users to make announcements using both the telephone speakers and the office’s loudspeaker system</td>
</tr>
<tr>
<td>Paging Set Access</td>
<td>Individual sets can be denied the ability to perform paging</td>
</tr>
<tr>
<td>Park and Page from Mailbox</td>
<td>Allows an external caller to page a mailbox owner while listening to the owner’s personal greeting or while recording a message</td>
</tr>
<tr>
<td>Password Protection</td>
<td>Allows the coordinator to change the system administration password</td>
</tr>
<tr>
<td>Preselection/Call Screening</td>
<td>The assigned name of the caller’s set or line will appear on the set display</td>
</tr>
<tr>
<td>Prime Line</td>
<td>A line (CO, Intercom or Line Pool) can be assigned to a set as its primary line of use for automatic outgoing line selection</td>
</tr>
<tr>
<td>Prime Set</td>
<td>A set can be designated as prime or backup to receive unanswered calls via Delayed Ring Transfers, Held Line Reminders and Do Not Disturb transfers and overflow call routing</td>
</tr>
<tr>
<td>Priority Call</td>
<td>Can interrupt a conversation on a busy set or override Do Not Disturb (DND)</td>
</tr>
<tr>
<td></td>
<td>Users have the option to block a Priority Call, but it cannot be ignored</td>
</tr>
<tr>
<td>Privacy – On Lines</td>
<td>Automatically prevents another telephone, which shares a user’s line, to access or join the call</td>
</tr>
<tr>
<td>Privacy Control</td>
<td>The Privacy ON/OFF switch lets a third person join the call</td>
</tr>
<tr>
<td>Receiver Volume</td>
<td>Allows users to program volume</td>
</tr>
<tr>
<td>Features</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------------</td>
<td>-----------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Remote System Access</td>
<td>Allows callers on the public network to access the system directly, without going through the attendant. Once in the system, a caller can access some of the system's resources (dialing capabilities, line pool access, feature access)</td>
</tr>
<tr>
<td></td>
<td>Auto Answer DN – when a user dials into the system on an auto-answer loop-start trunk that is not configured to answer with DISA, no password is required to access the BCM. The Class of Service (COS) that applies to the call is determined by the COS for the trunk on which the user is calling</td>
</tr>
<tr>
<td></td>
<td>DISA DN – when a user dials in on a trunk that has auto-answer with DISA, the system presents a stuttered dial tone to prompt the user to enter their Class of Service password. The Class of Service that applies to the call is determined by this COS password</td>
</tr>
<tr>
<td>Restriction Override</td>
<td>Allows users to bypass any call restrictions applied to any set or line</td>
</tr>
<tr>
<td>Password</td>
<td></td>
</tr>
<tr>
<td>Ring Again on Busy Set</td>
<td>Alerts a user when a previously busy set becomes available</td>
</tr>
<tr>
<td>Ring Again on Busy Line Pool</td>
<td></td>
</tr>
<tr>
<td>Ring Again on No Answer</td>
<td>Notifies a user when a set that was not answered is used</td>
</tr>
<tr>
<td>Ringing Line Preference</td>
<td>Automatically places the longest ringing call to the head of the queue when several lines are ringing</td>
</tr>
<tr>
<td>Ringing Service</td>
<td>Now alternate ringing can be programmed for day of week as well as time of day</td>
</tr>
<tr>
<td>Features</td>
<td>Description</td>
</tr>
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<td>----------------------------------------------</td>
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</tr>
<tr>
<td>Routing Service/Destination Codes</td>
<td>A programming section that allows outgoing calls to be directed automatically, based on the numbers a caller dials (also called Automatic Route Selection – ARS)</td>
</tr>
<tr>
<td></td>
<td>For systems linked in a network, routing can create a transparent or coordinated dialing plan. It can also be used to direct calls to the least expensive lines according to a Services schedule (sometimes called least cost routing).</td>
</tr>
<tr>
<td></td>
<td>BCM provides multiple alternate routes (more than one route) for each service mode. The Least Cost Routing (LCR) feature can be more effectively used by allowing an attempt to route the call through multiple alternate routes before the call is directed back to the normal mode. Two alternate routes are provided in each mode so that there are a total of three routes available for each mode. If all three routes are programmed in order of priority for a service mode, then all these routes will be tried in order of priority before the call falls back to normal mode.</td>
</tr>
<tr>
<td></td>
<td>The digit absorption feature makes programming route easier. Digit absorption selects the portion of the destination code that is always absorbed by the system and not used in the dialing sequence.</td>
</tr>
</tbody>
</table>
|                                              | *Number of Destination Codes: 500  
*Number of Destination Routes: 999  
*Number of Dialed Digits: 12, allowing for more distinct destination codes*                                                                                                                                                                                                 |
| Saved Number Redial                          | It saves and later recalls the external telephone number appearing on the display                                                                                                                                                                                                                                                      |
| Selective Call Rejection *                   | When activated, screens incoming calls against a user-specified list of numbers to be rejected. If rejected, the caller hears a message informing them the called party does not wish to receive their call; the last incoming number can be added to the selective call rejection list even if the number is “Private”                                                                                     |
| Selective Line Redirection                   | See Line Redirection                                                                                                                                                                                                                                                                                                                      |
| Set Names                                    | Names can be programmed for internal sets                                                                                                                                                                                                                                                                                               |
| Set Profile                                  | With the Unified Manager, allows the administrator to view system data for each set in the BCM system                                                                                                                                                                                                                                   |
| Service Modes                                | Three different service modes can be programmed (i.e., lunch, evening, night) with their own ringing arrangements for automatic or manual activation                                                                                                                                                                                     |
| Speed Dial Access – Personal and System      | Allows users to access both Personal and System Speed Dial Codes  
*Number of Digits: 24  
*Number of Entries: 70*                                                                                                                                                                                                                                                                                                               |
<p>| Speed Dial Line Selection                    | For speed dialing, the system will use a specific line as determined in administration                                                                                                                                                                                                                                                                                                            |</p>
<table>
<thead>
<tr>
<th>Features</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Speed Dial – Personal Programming</td>
<td>Allows a user to add or change a Personal Speed Dial number on their set</td>
</tr>
<tr>
<td>Speed Dial System Bypass Restrictions</td>
<td>Allows users to program speed dial numbers to override set and line restrictions</td>
</tr>
<tr>
<td>Speed Dial System Names</td>
<td>Allows users to program names instead of numbers on the set display when they access the speed dial code</td>
</tr>
<tr>
<td>Start DN Option</td>
<td>Allows user to choose the start DN and DN length, rather than the previous mandatory 221. Is typically programmed by the installer under Configuration.</td>
</tr>
<tr>
<td>Station Set Test</td>
<td>Allows users to determine if there is a physical problem with their Business Series Terminal before returning it to the distributor or factory for repair. Pressing Feature 805 on the set activates Station Set Test. The LCD prompts the user through the testing procedure. All tests are available for all sets, with the exception of the Headset Speaker Test, which is not available on the T7100, because it does not have a headset option.</td>
</tr>
<tr>
<td>System Speed Dial Increase</td>
<td>Allows users to increase the number of System Speed Dials from 70 to 255. Configurable in Unified Manager, this option increases productivity by allowing users to dial frequently used numbers at a faster rate.</td>
</tr>
<tr>
<td>System Wide Call Appearance (SWCA)</td>
<td></td>
</tr>
<tr>
<td>Target Lines</td>
<td>A virtual line dedicated to receiving and routing incoming calls on DID or auto-answer trunks to a specific destination; BCM supports up to 104 target lines which offer attendant bypass and line concentration.</td>
</tr>
<tr>
<td>Telephone Administration Lock</td>
<td>Three settings (Full, Partial, None) can be programmed in Administration</td>
</tr>
<tr>
<td>Time and Date Display</td>
<td>The time and date appear on the LCD display of an idle set</td>
</tr>
<tr>
<td>Time and Date – Show Time</td>
<td>Temporarily displays (for three seconds) the time and date while on a call</td>
</tr>
<tr>
<td>Timed Release</td>
<td>Signal releases a call from the line, but keeps the line for another call</td>
</tr>
<tr>
<td>Transfer Immediate (with Callback)</td>
<td>Allows users to transfer calls directly to another set; if unanswered, callback occurs after a preset number of rings</td>
</tr>
<tr>
<td>Transfer Using Conference</td>
<td>Allows users to transfer a call to an internal number</td>
</tr>
<tr>
<td>Transfer Using Hold</td>
<td>Allows users to transfer calls using the Hold button</td>
</tr>
<tr>
<td>Transfer with Announce</td>
<td>Allows users to announce an internal or external call to the designated party before transferring it, by simply staying on the line. To do an immediate transfer, press “OK” soft key or the release key.</td>
</tr>
<tr>
<td>Features</td>
<td>Description</td>
</tr>
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</tr>
<tr>
<td>Unsupervised Conference</td>
<td>Allows a user to establish a conference call with two outside parties and then exit from the call without disconnecting the remaining two people, provided one of those callers was incoming and the incoming line has Disconnect Supervision</td>
</tr>
<tr>
<td>Voice Call</td>
<td>Allows users to make a voice announcement or begin a conversation through the speaker of another telephone</td>
</tr>
<tr>
<td>Voice Call Deny</td>
<td>Prevents a set from receiving Voice Calls</td>
</tr>
<tr>
<td>Wait for Dial Tone</td>
<td>Causes a sequence of numbers to pause until dial tone is present on the line before continuing to dial</td>
</tr>
</tbody>
</table>
Data Capabilities

BCM200/400 can be equipped with an Optional WAN card to provides a router, channel service unit (CSU) and system management interface, all running on one platform. It is a cost-effective solution that supports Voice over IP (VoIP) as well as the following data capabilities:

- TCP/IP, the protocol used in over 60% of LANs and growing due to the Internet
- 10 or 100 Mbps Ethernet, used in over 70% of LANs due to its simplicity and low cost
- Static Route, RIP (Routing Information Protocol), RIP 2 and open shortest path first (OSPF) routing protocols
- Point-to-Point Protocol (PPP) or Frame Relay for permanent WAN connections
- Network Address Translation (NAT) and Port Address Translation (PAT)
- MultiLink PPP (MLPPP) support for dial-on-demand ISDN WAN interfaces
- A choice of permanent (T-1, V.35, X.21) or dialup (ISDN dial-on-demand or persistent) WAN connections as well as ISDN or V.90 analog dial backup
- T.38 Fax over IP
- VoIP QoS enhancement and Universal T1
- Dual V.35 WAN

VoIP minimizes costs with a single IP address to a public WAN interface and increases security by hiding internal IP addresses utilizing IPSec VPN support as part of the Nortel Secure Routing technology. This technology stems from the industry leading Contivity line of products and is an excellent means of combining VPN support with security in one device.

Routing Services

Routing in the BCM is accomplished with a combination of services. IP routing is provided via static route, RIP, RIP 2 and/or OSPF. Basic and stateful packet filtering provide a layer of security for all information entering and leaving the BCM.

IP

IP (Internet Protocol) is the protocol used on the Internet to send data from one computer to another. Each computer on the Internet, called a “host,” has at least one address that distinguishes it from all other computers on
the Internet. When computer users send or receive data (emails or Web pages), IP divides the message into units called packets. Each packet contains both the sender’s and the receiver’s Internet address.

IP sends a packet first to a router that reads the destination address, and then forwards the packet to an adjacent router that reads the destination address. This routing process continues across the Internet until one router recognizes the packet as belonging to a computer in its immediate neighborhood, or domain. That router forwards the packet to the computer whose address is specified.

Because a message is divided into a number of packets, IP can send each packet by a different route across the Internet. Packets can arrive in a different order than the order in which they were sent. Another protocol, the Transmission Control Protocol (TCP), puts the packets in the right order.

IP is a connectionless protocol, which means that the endpoints that communicate do not have an established connection. Each packet that travels through the Internet is treated as an independent unit of data without any relation to another unit of data. The packets get put in the right order because TCP, the connection-oriented protocol, keeps track of the packet sequence in a message. Both servers and workstations on a network must have IP addresses.

There are two ways of assigning IP addresses:

- **Dynamic**: A dynamic IP address changes. An IP address server assigns these addresses to computers as they need them. With dynamic IP addressing, a computer can have a different IP address every time it connects to the network.

- **Static**: A static or fixed IP address never changes. It is assigned to a computer permanently. The computer has the same IP address every time it connects to the network and is known to other devices on the network by that IP address.

**Static Routing**

In static routing, system administrators manually add static routes to the IP routing table. These static routes take precedence over those chosen by routing protocols, such as Routing Information Protocol (RIP). Static routing provides more security than RIP because it is possible for a hacker to attach a purposely mis-configured RIP router to a network.

However, static routing limits inter-networking to a fairly small scale because in order to perform multihop routing, each static router must be configured with entries for all the other networks.
RIP

BCM uses the Routing Information Protocol (RIP) to manage routing information in a self-contained network, such as a corporate intranet. Every 30 seconds, an RIP router sends full updates to its closest neighbor host. These updates list all the other hosts it knows about. The neighbor host sends the information to its next neighbor until all the hosts in the network know the routing paths. This state is called network convergence. Each host with a router in the network uses the routing table information to determine the next host for the packet, until a specified destination is reached.

RIP is a time-saver because it allows automation of the process of learning the routes between routers instead of manually adding static routes.

RIP v2

RIP version 2 is very similar to the original version, but was developed to address some of the shortcomings of RIP v1. All of the original limitations of RIP apply to RIP v2; however, the critical difference is that RIP v2 can be used in networks that require either support for authentication and/or variable length subnet masks.

RIP v2 has the following limitations:

- Remains unsuited for network environments that require routes to be selected in real-time based on either delay, traffic loads or any other dynamic network performance
- 15 hops
- Counting to infinity
- Static distance-vector metrics.
Open shortest path first (OSPF) is the most well-known and deployed link state routing protocol today. OSPF is an interior intra-domain routing protocol and is supported on most, if not all, routers on the market. OSPF has the following primary functional attributes:

- Includes link state routing algorithm, also referred to as shortest path first
- Supports multiple equal-cost paths to the same destination
- Has two-way hierarchy
- Generates link state advertisements only as a result of changes in network topology
- Is extensible.

OSPF offers the following advantages:
- Scales much better than RIP
- Has the ability to incrementally extend and enhance the function it provides by simply defining and adding new fields to carry new information.

**Note:** The implementation on the BCM is designed to operate as an edge router (“OSPF-Other”) in an OSPF intranet, or to be a backup router (BDR) in a small network. The BCM should not be configured for multiple OSPF areas.

OSPF is a “link state” protocol, meaning that it can become unstable with frequent link state changes which could arise from constant dialing and disconnecting. Therefore, the BCM does not support OSPF configuration on dialup interfaces (i.e. V.90 and ISDN). Likewise, OSPF and RIP cannot be used for RIP or OSPF redistribution with BCM LAN and WAN interfaces running different protocols.

*Figure 32.*
Packet Filtering

Packet filtering provides security for all information that enters and leaves the BCM. A packet is a unit of data routed between an origin and a destination on the Internet or on any other packet-switched network. When a user sends any file, such as an email message, HTML file, GIF file or URL request on the Internet, the IP layer divides the file into packets of an efficient size for routing. Each packet is numbered and includes the destination’s Internet address.

BCM supports basic and stateful packet filtering for IP. Basic filtering means the filter can be configured to pass only the packets from the routes they list, or to pass everything except the packets for the routes they list.

Stateful packet filters maintain state information for each flow (TCP, UDP or ICMP) and for the following protocols:

- H.323
- FTP
- Telnet
- SMTP and SNMP Traps
- DNS
- TFTP
- Gopher
- Finger
- HTTP
- POP3
- NNTP
- RPC
- SUNNFS

Using this state information, the system can determine if a packet should be allowed to pass through the BCM system based on whether the original flow was initiated from inside or outside the BCM and its LAN environment.
IP Services

BCM provides a number of services that enhance IP Routing. Network Address Translation (NAT) allows address allocation that provides routing stability and network scalability. DHCP provides automatic assignment of IP addresses, DNS maps easy-to-remember names to IP addresses and Web caching allows multiple users to share information downloaded from the Internet.

Network Address Translation (NAT)

Network Address Translation (NAT) allows a network administrator to translate one set of IP addresses into another. For example, NAT allows a host with a private address to appear on the Internet with a registered address. It can be used to balance loads between servers, provide server redundancy and connect companies that use the same address space.

BCM NAT includes static and dynamic address translations for TCP, UDP and ICMP packets.

BCM also provides NAT support for the following protocols:

- H.323
- FTP
- Telnet
- SMTP
- SNMP, SNMP Traps
- DNS
- TFTP
- Gopher
- Finger
- HTTP
- POP3
- NNTP
- RPC
- SUNNFS.
Port Address Translation (PAT)

Static translation occurs when a one-to-one mapping is created between an inside and outside address. Dynamic translation creates either one-to-many, many-to-many, or many-to-one mapping between inside and outside addresses. The benefit of Port Address Translation (PAT) is sharing one address to many ports.

DHCP

Dynamic Host Configuration Protocol (DHCP) lets network administrators manage and automate the assignment of IP addresses in an organization's network. If an organization sets up its computer users with a connection to the Internet, an IP address must be assigned to each machine.

Without DHCP, a user at each computer must manually enter the IP address. If computers move to another location in another part of the network, users must enter a new IP address. DHCP allows a network administrator to supervise and distribute IP addresses from a central point and automatically sends a new IP address if a computer connects to a different place in the network.

BCM can be set up to be the LAN’s DHCP server and to let it assign IP addresses dynamically to the workstations on the LAN as necessary. When deployed in a network where there is a separate DHCP server, a built-in DHCP Relay Agent on the BCM allows the pass through of DHCP traffic to and from LAN connected devices.

DHCP supports the following attributes:

- DNS Servers (Primary and Secondary) (06)
- IP Domain Name
- WINS Server (044)
- WINS Node Type
- Default Gateway (03)
- Lease time In Seconds (051).

The BCM DHCP Server supports the following features:

- **DHCP Remote Scope** – provides the ability to create additional scopes other than LAN 1 and LAN 2. This allows devices connected across a WAN interface, such as remote IP Softphone 2050 clients, to be assigned IP addresses by the BCM.
- **Reserved Addresses** – allows the administrator to reserve IP addresses for specific moves, adds and changes (MAC) addresses.
- **Support for Nortel IP Clients** (128) – automatically provides IP Telephony clients with the address of the telephony server during the DHCP assignment. This allows IP telephones served by the BCM to operate in a plug-and-play mode rather than having to be manually configured with the telephony server address.

*Figure 33.*

**DNS**

The Domain Name Service (DNS) is the system within the Internet that maps names of objects, usually host names, into IP numbers or other resource record values. The name space of the Internet is divided into domains. The responsibility for managing names in each domain is usually delegated to systems in each domain.

BCM functions as both a gateway to the Internet and as a DNS proxy:

- **Gateway** – is a system that links two different types of networks and enables them to communicate with each other. BCM is the gateway that links a company’s network to an intranet or to the Internet.
- **DNS Proxy** – translates alphabetic domain names into computer-readable IP addresses. For example, the domain name www.nortelnetworks.com for the Nortel Website can translate to the IP address 192.177.5.18. After a domain name is translated into an IP address, the workstations on a network can communicate with the Website.

**IPSec**

IPSec is a developing standard for security at the network or packet-processing layer of network communication. IPSec is especially useful for implementing a VPN. One of the main advantages of IPSec is that businesses can handle security arrangements without having to make changes to individual users’ computers. IPSec offering on BCM provides privacy, integrity and authenticity for networked commerce-crucial requirements for transmission of sensitive information over the Internet.

The level of encryption created is based on the choice of protocol, encryption method and the authentication method. The implementation of IPSec on the BCM supports the Encapsulating Security Payload (ESP) and Authentication Header (AH) protocols. ESP provides confidentiality for IP datagrams by encrypting the payload data to be protected. ESP uses the Data Encryption Standard (DES) algorithm. AH protocol provides data integrity and source authentication but does not encrypt data.

The encryption method on the BCM can be set for 128-bit Triple DES, 56-bit DES or 40-bit DES, with Triple DES being the strongest and 40-bit DES being the weakest level of encryption.

The authentication method can be either Secure Hash Algorithm (SHA1) or Message Digest 5 (MD5) Algorithm. SHA1 produces a 160-bit hash, but does not encrypt data. MD5 produces a 128-bit hash. It is used to confirm the authenticity of a packet but also does not encrypt data. MD5 also provides integrity that detects packet modifications. Both SHA1 and MD5 use Hashed Message Authentication Code (HMAC) to improve authentication. HMAC is a technique that uses a secret key and a message digest function to create a secret message authentication code. Cryptographers regard SHA1 as being more resistant to attacks than MD5.

The BCM NATs, firewall and firewall filters are supported in an IPSec environment.

The BCM IPSec capability is based on the Contivity client capabilities. IPSec on BCM allows up to 16 secure tunnels to be established between BCM and Contivity and/or BCM to BCM. BCM also supports 16 SOHO clients.

**PPTP**

Point-to-Point Tunneling Protocol (PPTP) is a proposed standard sponsored by Microsoft as an extension of the Internet’s Point-to-Point Protocol. Any user of a PC with PPP client support is able to use an Internet service
provider to connect securely to a server elsewhere in the user's company. The PPTP implementation on the BCM is designed for router-to-router configurations only; it does not support personal clients. In order for this facility to work, the username and password for each remote router must be set up on BCM.

The BCM implementation of PPTP offers the following features:

- Support for multiple authentication schemes: MS-CHAP, CHAP or PAP
- Support for IP address translation via encapsulation
- Support for RC4 encryption (either 56-bit or 128-bit, within the limits of US export law)
- Support for compression of data packets

A total of ten PPTP tunnels can be configured on the BCM.

**Note:** While BCM supports both IPSec and PPTP, they may not be used at the same time.

### Web Caching

When BCM is used as a Web proxy, it can store, or cache, information downloaded from the Internet. A proxy is a server that acts on behalf of another. Web caching allows LAN workstations to share common information downloaded from the Internet.

Data is usually cached on individual workstations. Each time a workstation on the LAN requests information from the Internet, the individual’s request is sent to the Internet and the information is returned to their workstation. If multiple LAN workstations request common data, a Web cache on the network reduces download time from the Internet. The BCM defaults to a 20 MB cache size, but can be configured with up to 100 MB.

When BCM is configured as a Web proxy with Web caching:

- LAN workstations have shorter download times
- Previously downloaded information is stored for future use by all workstations on the LAN
- BCM retrieves information from the Internet only if it is not already cached or if the cached file is out of date compared to the information on the Internet
- Cookie blocking protects users’ privacy.

The Web proxy also provides security features similarly to the DNS proxy because it hides all of the internal browsers’ IP addresses from external Web servers. External Web servers see only the BCM IP address.
T.38 Fax over IP

This feature supports fax communications over the IP network to other BCMs or to the following systems:

- Communication Server 1000
- Communication Server 2100 Release SN08

Fax over IP can offer significant cost savings versus PSTN connections. With this feature, the BCM becomes a T.38 gateway. T.38 support allows the BCM to support the T.30 session with the locally attached fax machine and translate the T.30 session into T.38 message procedures for transmission across the IP connection.

Communication Server interoperability via H.323

BCM systems are often networked to other Nortel call servers. For this reason, all Nortel Enterprise Call Servers are tested, and if necessary, modified to ensure the highest level of seamless networking. The following Nortel Call Servers will interoperate with BCM via the H.323 protocol:

- Communication Server 1000 Release 3.0 and Release 4.0 (formerly Succession 1000)
- Communication Server 2100 Release SN08
- Multimedia Communication Server 5100 and 5200 Release 3.0
- Norstar VoIP Gateway – all Releases

SIP Trunk & Data Services Support

BCM supports SIP trunk connections, with functionality similar to H.323. Trunk connections (enabled with VoIP gateway Keycodes) via SIP protocol will be supported BCM to BCM. (MCDN, T38 fax and SIP clients are not supported.)

BCM also includes data services support that defines an application level gateway (ALG) solution for the following BCM data services:

- Network Address Translation (NAT)
- Firewall
- Quality of Service (QoS)

These features examine each SIP packet and modifies the IP addressing information in both signalling and payload as appropriate for NAT. It opens pinholes to let SIP packets through the Firewall. It also prioritizes and marks the packet to achieve the QoS requirement.
LAN Connections

The BCM Ethernet / 802.3 interface supports the IEEE 802.3 Ethernet frame format. The Ethernet interface uses Carrier Sense Multiple Access with Collision Detection (CSMA/CD) to manage the access to the physical media.

The BCM Ethernet interface supports the following features:

- 100 BASE – TX with RJ-45 connector
- 10 / 100 Auto Sense
- Full Duplex support
- Fast LAN-LAN routing
- LAN traffic smoothing.

LAN-to-LAN Fast Path Routing

BCM provides an optional second 10/100 LAN (Local Area Network) interface. If the second LAN interface is used to send data on from the first LAN interface, BCM provides optimized software for high performance routing. This includes an innovative design that speeds up the performance for LAN-to-LAN routing by over three times the rate that is normally achieved with traditional software architecture.

WAN Connections

A wide area network (WAN) is a geographically dispersed data communication network. The term WAN distinguishes a broader data communication structure from a local area network (LAN). A WAN can be privately owned or rented, but usually means the inclusion of public (shared user) networks.

- BCM provides a choice of permanent (T-1, V.35, X.21) or dialup (ISDN dial-on-demand or persistent) WAN connections as well as ISDN or V.90 analog dial backup.

- The primary WAN link is always a permanent link and it is a dedicated network adapter. It runs either Frame Relay or the PPP protocol at the link layer. The BCM primary WAN connection is through a two-port card. These two ports can be independently configured to run Frame Relay or PPP. For North America, the card includes one serial sync port (V.35) and one T-1 port, while the international version includes both a V.35 and X.21 serial port.

- The backup WAN link is always configured as a dial-on-demand network adapter by the router. The backup WAN link runs PPP only. BCM provides backup WAN connection through a V.90 modem or ISDN B-channels. The primary and backup link management is performed from BCM. A NetLink Manager runs in
BCM and monitors the primary link status and starts the backup link when a break in the primary link is detected. Similarly, the backup link is automatically terminated when the primary link becomes active and stable. This process can take between 90 and 180 seconds. A feature selectable in the Unified Manager allows administrators to select a fast fallback method that should take 20 seconds or less.

- NetLink Manager manages the default route in BCM. If a link breaks, NetLink Manager removes all the default routes on the broken link and adds the default route to the new link. This happens during switch over from primary to secondary links and vice versa.

- BCM provides a dual WAN card as an orderable option. It has the same V.35 protocol supported on both ports. This allows a customer to connect two different CSU/DSUs to the WAN card in the BCM.

**Frame Relay**

BCM supports direct mode operation on the WAN interface. This operation allows each WAN interface to be assigned more than one IP address, which is useful for using a single WAN physical link to connect to both an intranet and the Internet using separate addressing schemes. Likewise, a network service provider may create a separate IP address for management functions over the WAN interface. In either case, broadcast traffic destined for one IP address would not be transmitted on links associated with the other IP address. Up to five IP addresses can be assigned to each WAN interface. Static routes and RIP/RIPv2 routing is supported when multiple IP addresses are configured.

Direct mode enables a single T-1 to support Intranet or corporate WAN access and Internet access, dynamically sharing bandwidth. This eliminates the need for a second T-1, which significantly reduces the recurring monthly cost of network access.

*Figure 34.*
Point-to-Point Protocol

Point-to-Point Protocol (PPP) is a protocol for communication between two computers using a serial interface, typically a personal computer connected by phone line to a server. For example, an Internet server provider can provide a PPP connection so that the provider's server can respond to requests, pass them on to the Internet and forward the requested Internet responses back to the requester. PPP uses the Internet Protocol (IP).

PPP is a full-duplex protocol that can be used on various physical media, including twisted pair or fiber optic lines or satellite transmission. It uses a variation of High Speed Data Link Control (HDLC) for packet encapsulation. PPP can process synchronous as well as asynchronous communication. PPP can share a line with other users and it has error detection.

PPP on the BCM’s primary WAN link uses synchronous point-to-point communication. Because the physical media is point-to-point, authentication attributes are not supported in this mode.
BCM supports PPP Compression Control Protocol (CCP) (RFC 1962) with STAC compression algorithm. This compression can be enabled or disabled by using a parameter in PPP configurations. MultiLink Point-to-Point Protocol (MLPPP) is supported for combining up to 16 ISDN B-channels into a single data connection.

**Point-to-Point Protocol over Ethernet (PPPoE)**

The customer interface on most DSL modems is an Ethernet connection, requiring the customer Internet access device to support the PPP protocol over Ethernet link, referred to as PPPoE. BCM 2.5 (FP-1) supports this with an optional Keycode to enable this ISP connection.

**Quality of Service (QoS)**

In a network using Internet Protocol (IP), Quality of Service (QoS) is the method by which transmission rates, error rates and other characteristics can be measured and improved. QoS is a concern for the continuous transmission of high-bandwidth voice and video multimedia information.

Real-time applications that include voice and video are time sensitive. Delivering voice and video over the Internet requires control of packet delay and jitter. Differentiated Services (DiffServ) is a QoS framework standard that focuses on DiffServ standards for real-time and mission critical applications. The DiffServ standards are evolving and vendors are starting to develop network devices that support DiffServ.

The purpose of the BCM QoS module is to prioritize IP traffic and to provide an acceptable quality of service to delay and jitter sensitive applications such as audio and video as well as mission critical applications.

The BCM QoS module serves two primary purposes:

- In a DiffServ network, it performs the packet classification, marking and prioritization
- In a non-DiffServ or legacy network, it manages the WAN link to make sure premium voice (and optional video) packets get high priority when crossing the slow WAN link in both directions.

**QoS Module and VoIP QoS Monitor**

The Internet Telephony gateway in BCM includes a Quality of Service Monitor (QMON) that periodically monitors the delay and jitter of IP networks between two peer gateways by using a proprietary protocol. These monitoring packets are delivered at UDP port 5000.

The main objective of the QMON is to allow new VoIP calls to fall back to the PSTN if the IP network is detected as “bad.”
The QoS module complements QMON. While QMON passively monitors the IP network, the QoS module actively improves the IP network by giving VoIP packets higher priority to travel so that the chance for QMON to detect “bad” is reduced.

**Note:** For a VoIP call, if a packet passes QMON but fails the QoS admission control, it is delivered over IP, but only as a best-effort flow. There is no fallback to PSTN if a packet has passed QMON checking.

BCM provides a set of options for improving QoS over low speed fractional T1 data connections. These options include Layer 2 Fragmentation and IP/UDP/RTP Header Compression. BCM supports both PPP Fragmentation and Frame Relay Fragmentation, both of which will temporarily interrupt the transmission of the larger non-VoIP packet in order to send the higher priority VoIP packet. Once the VoIP packet has been processed, the remaining portion of the interrupted packet can be sent. To take advantage of these QoS features, the BCM routing WAN interface must use the Digital Trunk MBM interface.

**Common Open Policy Service (COPS)**

BCM provides the capability of Common Open Policy Service (COPS), a key ingredient in Nortel high-performance networking solution. Support for COPS-PR protocol on the BCM allows network policies to be established centrally and be “pushed” to the BCM from an Optivity Policy Server. This allows a network administrator to establish different network policies for time of day or in reaction to other network events like system outages and have the changes automatically propagated to all policy enabled devices.

While Optivity Policy Server is designed for managing policy administration in large networks, policies on BCM can also be set through SNMP messages from other platforms, or they can be set locally as static policies. BCM can have a static policy assigned and still receive a dynamic policy from a policy server. In this case, the dynamic policy will be used until a new dynamic policy is sent or until the existing dynamic policy expires or times out. If it expires, BCM will default to using any existing static policies.
Data Capabilities of the BCM50

For the small site, BCM50 offers maximum flexibility for data management. The BCM50 Main Unit is available in three versions:

- A BCM50 with no router
- A BCM50 with Ethernet Router (BCM50e)
- A BCM50 with ADSL Router (BCM50a)

The option of an integrated router is tailored to the needs of small businesses which have a small number of people using external data networking, such as internet access or for enterprise branch networking.

Customers can choose a four-port integrated router that is either Ethernet or ADSL, depending on what type of data service they use now. If they want to adopt more advanced data options, they can select a BCM50 without an integrated router and combine that with a more advanced Nortel routing solution, such as Contivity Secure Routing.

Ethernet or ADSL routing on BCM50 provides generic Internet access with all typical router features, including firewall, security, Network Address Translation, and more, for secure access to the internet.

BCM50 can also be used as the basis for Virtual Private Networks, or VPNs, to support a small mesh network of BCM50s. Typically, they would be interconnected on secure tunnels for data and VoIP virtual network applications. When using BCM50 with other larger sites, customers can connect their branch BCM50 to a central office equipped with a centralized VPN server. To enable work-at-home users, they can support VoIP and data connectivity with a Nortel Contivity VPN device.

Applications such as voice messaging and attendant capabilities can be centralized, thus reducing overall operating costs, and BCM Network Configuration Manager (NCM) can provide fast and effective standardized roll out of programming to either new or existing sites, regardless of location.

The BCM50e or BCM50a main unit with optional router provides an additional 3 ports of ethernet LAN switch for a total of 6 LAN ports for local premise use. All Ethernet ports are 10/100 Mbps auto-sensing, and support auto-polarity, so no cross-over cable is required to connect data hardware to the unit. An additional port is provided for WAN access, either Ethernet or ADSL depending on the model.
The following features make the variants of the BCM50 with embedded router an attractive package for small sites wishing to become Internet-capable, multisite enterprises with many small sites which formerly could not be part of the corporate WAN due to high cost of traditional WAN connectivity, and managed service scenarios:

### VPN
- 5 IPSec tunnels
- IKEv1 Main Mode
- IKEv1 Quick Mode
- Diffie-Hellman Group 1,2
- IPSec Tunnel Mode
- ESP
- Support for Dynamically addressed peers – ABOT
- NAT Traversal

### Security Services
- Cryptographic Services
- 3DES
- DES
- Data authentication SHA-1
- Data authentication MD-5
- Authentication Services
- Pre-shared secrets
- Security Services
- Stateful Firewall
- Intrusion Detection

### NAT
- Many to one, static, many to many
- Port forwarding
- IPSec Pass through
- H.323 ALG
- NAT support for tunnel mode IPSec tunnels

### Router
- Clear text routing
- Static – via tunnel
- RIP v1 – via tunnel and clear text
- RIP v2 – via tunnel and clear text

### IP Services
- DHCP client
- DHCP server with support for Nortel Networks Internet Telephones
- DNS Proxy
- DNS w/ VPN client
- PPPoE
- Configurable MAC address
- 5 Mbps clear text routing with 1500 byte packets
- 1.5 Mbps 3DES throughput w/ 1500 byte packets
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Messaging

The messaging industry has experienced explosive growth over recent years, as more and more decision-makers have become aware of messaging products and the benefits these provide for their businesses. In fact, most businesses and organizations today use some form of messaging, whether it is equipment on their premises or Messaging mailboxes that are provided by a service bureau or telephone company.

BCM incorporates Messaging features that address the needs of small- and medium-sized customers. These include:

- Automated Attendant, which allows callers to reach features and extensions by browsing through menu options
- Voice Messaging, which allows callers to leave messages for specific people
- Custom Call Routing, which allows customers to route incoming calls depending upon specific user entry
- Unified Messaging, which allows users to manage voice, fax and email messages through a single, PC-based interface, giving them a single point-and-click inbox.
- Fax Messaging, which allows the user to receive, send and forward faxes in the same fashion as voice messages

BCM supports Unified Messaging (V 1.07 & 2.01), which supports a variety of email clients including:

**Groupware email clients:**
- Microsoft Exchange 4.0 and 5.0
- Microsoft Outlook 98 (Corporate Mode)
- Lotus Notes 4.6x/5.x/6.x
- Groupwise 6.x

**Internet IMAP email Clients:**
- Microsoft Outlook Express 5.x/6.x
- Microsoft Outlook 98/2000/2002/XP (Internet Mail Mode)
Email client compatibility includes:

- Windows NT 4.0 SP6a
- Windows 95B
- Windows 98SE
- Windows 2000 Professional
- Windows XP

Unified Messaging improves employee productivity because users can more quickly and easily manage their voice and email messages from their PC or laptop, locally or remotely. And because message management is performed over the Local Area Network (LAN), voice messaging channels are more available to handle incoming customer calls.

Networking capability is also emerging as a technology solution customers require to run their businesses. Using Voice Profile for Internet Mail (VPIM) to message over an existing data network between sites can improve overall intra-company communications and maximize the investment in the existing data network.

Remote administration from the PC desktop is another customer demand that is addressed by the three software utilities:

- Voice Messaging
- Personal Mailbox Manager (BCM200/400 Only)
- Operator Manager (BCM200/400 Only)

These software utilities can be used to manage various aspects of the system, a Personal Mailbox, or the Automated Attendant function and can be used on any computer that runs Windows 95/98/2000/NT or XP operating systems.

The total number of channels available to be shared between Messaging and Call Center is 32 channels on BCM200/400 and 10 channels on BCM50, allowing for improved application (Messaging and Call Center) performance and simultaneous access.

The BCM200/400 allow for 200 hours of voice messaging storage, while the BCM50 allows 100 hours.
Messaging Overview

Messaging Components

Messaging has the following components: main and additional.

**Main Components**

- Automated Attendant
- Mailboxes (voice messaging)
- Custom Call Routing (CCR).

**Additional Components**

- Networking
- Unified Messaging
- Fax Messaging.

Automated Attendant

The Automated Attendant works as a receptionist would when answering incoming calls. Using a voice prompt, it offers callers a list of options. If callers know which option they want, they can interrupt the Automated Attendant by pressing their selection on the dial pad of any touchtone telephone. The Automated Attendant responds to the command by either routing the call to an extension or mailbox within the company or directing the caller to the company directory or designated operator.

Mailboxes (Voice Messaging)

The system coordinator adds mailboxes, which are then initialized by the Mailbox owners. Any caller can leave a message after a mailbox is initialized.

Custom Call Routing (CCR)

Custom Call Routing (CCR) is a single-digit access application, providing call routing paths that direct incoming calls based on the caller’s choices of recorded voice prompt options.
Networking

Networking links BCM Voice Messaging and other BCM voice messaging systems at different locations. It allows the exchange of voice messages between users at different sites on a network. Customers have two network messaging options: Audio Messaging Interchange Specification (AMIS) and Voice Profile for Internet Mail (VPIM) using TCP/IP.

Unified Messaging

Unified Messaging provides access to voice, fax and email messages from a PC through supported email applications. As the name implies all messaging is unified into one access point. All voice, fax and email messages can be managed from one graphical interface. It provides highly graphical user interfaces, making system management easy and effective. Also, Unified Messaging integrates with more email systems (including Lotus Notes, Microsoft Exchange/Outlook and Novell GroupWise, as well as with IMAP4-compliant email clients such as Netscape Mail, Eudora and Microsoft Outlook Express) than any other messaging product.

Fax Messaging

BCM Fax Messaging allows the user to receive, send and forward faxes in the same fashion as voice messages.

DTMF User Choice

Nortel Messaging (previously known as CallPilot) features two user inputs or interfaces into the messaging application.

Interface Support

BCM supports two different DTMF interfaces for users and callers:

- Nortel Messaging on BCM (previously known as Norstar Voice Mail DTMF)
- Nortel Messaging on Meridian and Communication Server 1000 (previously known as Meridian Mail/DTMF).

The Voice Messaging interface includes features such as “86” for “Go to Message” and “9” for “Call Sender.” It also includes Nortel Messaging features such as soft key control on sets and transfer to external number. The introduction of the Voice Messaging user interface provides both choice and simplicity at the same time.
Interface Consistency

BCM provides a consistent Messaging interface for both BCM and Communication Server 1000.

- Class of Service determines UI style for message retrieval
- System setting determines UI style for call answering and login
- Messaging Express is indistinguishable from Messaging in everyday use:
  - No internal greeting, speed controls, delivery to non-user, personal distribution lists
- Adds Outbound Transfer, personalized CLID greetings from Messaging on BCM
- Personal mailbox administration uses BCM-style menus.

Standard Messaging Software

BCM Messaging is standard and comes installed on every system. The mailbox seat licenses must be purchased and the Keycodes installed to activate voice messaging. The Voice Messaging capabilities include:

- Automated Attendant
- Custom Call Routing (CCR)
- Fax Answering.

Optional Voice Messaging Software Applications

The optional Voice Messaging software applications include the following components to enhance business communications:

- Message Networking
  - Audio Messaging Interchange Specification (AMIS)
  - Voice Profile for Internet Mail (VPIM)
- Unified Messaging
- Messaging mailboxes orderable in seat license increments of 1, 4, 8, 16, and 32
- Fax Suite: provides a packaged software option that includes:
  - Fax Messaging: ability to receive fax messages in the Messaging mailbox
  - Fax on Demand: provides printed information to callers without human intervention, such as brochures
• Fax Overflow: Fax Overflow Mailbox acts as virtual fax machine to receive faxes when the fax machine is out of service

For more information on Call Center, see the chapter entitled “Call Center”.

Automated Attendant

The Automated Attendant is the automated message application that answers and routes incoming calls – 24 hours a day, 7 days a week. Always on, always available. The Automated Attendant performs the following functions:

• Plays a list of choices to callers
• Routes calls to:
  • Extension
  • Mailbox
  • Company Directory
  • Designated Operator
  • Call Center Skillset (Control DN/Virtual DN)
• Allows users to:
  • Record greetings for time of day and non-business hours
  • Change company greeting and business status locally or remotely
• Routes calls to ACD Call Skillset (Control DN/Virtual DN)
• Enables Custom Call Routing (CCR) configuration by time of day
• Enables Graphical User Interface (GUI) for CCR programming
• Answers faxes.

When enabled, the Automated Attendant answers the company’s incoming telephone lines with greetings that can be played according to the time of day. When this component is enabled, the Automated Attendant menu prompt provides a list of options allowing a caller to:

• Reach a DN or a mailbox in the company
• Leave a message in a mailbox
• Select an alternate language
• Look for an extension or mailbox in the company directory
• Reach the company receptionist or designated operator
Open a Personal Mailbox as a mailbox owner.

The Automated Attendant provides callers with commands for using each of these options. A caller must press the button associated with the option he or she wants to activate; for example, “To use the Company Directory, press the pound key.”

The Automated Attendant has a Fax Answering feature that recognizes fax tones and routes incoming fax calls to a designated fax extension on the BCM. This feature means that a business can have just one main telephone number for both voice and fax calls and can avoid the expense of a separate trunk for fax calls.

Voice Messaging Call Answering Overview

Figure 36 shows an example of call flow with Automated Attendant:

*Figure 36.*
Custom Call Routing (CCR)

Custom Call Routing (CCR) is an application that works with Voice Messaging and Automated Attendant to provide a call routing path that directs incoming calls. CCR enhances Voice Messaging call routing abilities by allowing incoming callers to direct their own calls along paths created by the system administrator.

Designing and Building a CCR Tree

Designing a CCR Tree involves:

- Determining frequently requested departments
- Determining frequently called extensions
- Making a list of goods and services for promotion in information messages
- Selecting mailboxes assigned to leave message points
- Determining call destination types
- Recording the prompts.

CCR Trees can be configured and programmed using the graphical user interface (GUI) as shown below. The ability to program CCR with this tool from a PC allows the administrator to actually view the call flow as well as save the tree configuration to a file on his or her PC.

Home Menu

The Home Menu is the introductory voice prompt that the system administrator records. It provides a list of single-digit options to a caller. After listening to the Home Menu, a caller selects an option by pressing a number on any touchtone telephone.

Options in the Home Menu can route a caller to:

- An information message
- A mailbox to leave a message
- An extension
- Another menu.
**Company Directory**

The Company Directory is a list of mailbox owners registered with Messaging. Before any mailboxes can be used, the owners must record their names in the Company Directory. The administrator can change the names included in the Company Directory at any time.

**Mailboxes**

A mailbox is a storage place for messages. Subscriber Messaging mailboxes are orderable in seat license increments of 1, 4, 8, 16, and 32. (Up to 200 network delivery mailboxes can be configured on the system without requiring a Keycode.)

Types of Messaging mailboxes are:

- Special or Guest
- Personal
- Information
- Network Delivery
- Fax Overflow and Fax On Demand.

**Special or Guest Mailboxes**

Special Mailboxes are administered by the company’s system coordinator.

The system coordinator uses the System Coordinator Mailbox as a personal mailbox in which employees can leave messages.

**Operator Manager**

In the same way a user can manage his or her mailbox with Personal Mailbox Manager over the LAN, the system coordinator can use Operator Manager (main menu screen shot shown in Figure 37) to administer and program the Messaging system (BCM200/400 only).
Figure 37.

Operator Manager can also be used to manage other mailbox types, including:

- General Delivery
- Information
- Guest and Network Delivery.
Operator Manager gives the administrator a fast and easy way to manage the Automated Attendant function without ever having to leave his or her desk.

**Personal Mailboxes**

The system administrator assigns Personal mailboxes, which are maintained by the mailbox owners. A Personal mailbox can be a User or Guest mailbox. Personal Mailbox Manager (BCM200/400 only) can help users manage most of their mailbox functions and settings and even allows users to select, change and record greetings over the network without having to be at their office. Once users log in to their mailbox on the network, they are presented with folder choices for any changes they may wish to make.

The easy-to-use, Windows-based interface shown below makes setting up and making changes to features (like Off Premise Message Notification) simple: users just point and click on the options they want and type in the telephone numbers on their keyboard.

*Figure 38.*

- User mailboxes can be assigned to each user who has a Business Series Terminal, M7XXX series set or analog set, or a Digital Mobility handset. These mailboxes store messages for users who are unable to answer their telephones.
- Guest mailboxes do not have operating extensions but provide temporary employees and guests with access to internal messaging and call routing features.
Information Mailboxes

Information mailboxes are designed to provide an informative message to callers. This type of mailbox differs from the other mailboxes because it does not take messages. It plays a personalized greeting, but does not prompt for or allow the caller to leave a message. These mailboxes can be used to supply callers with information, like directions to the company, without involving expensive human intervention.

Network Delivery Mailboxes

Network Delivery mailboxes are used with the optional AMIS or VPIM networking applications to simplify addressing to remote locations.

AMIS (Audio Messaging Interchange Specification) Option

AMIS is the industry-standard specification for an analog networking scheme that allows different messaging systems to network voice messages over the public telephone network. For example, a customer with BCM and Octel or Centigram voice messaging systems in their sites would be able to send and receive networked messages using the AMIS software.

There are three ways to address a message for networking that are the same for both AMIS and VPIM:

- **Direct Addressing** – users must input the entire 10-digit number, in the case of long-distance messaging, when they address the message.
- **Site-based Addressing** – a network site-addressing table on the system contains the addresses of the remote sites to be networked; users only need to know the site address code and the mailbox number of the person to whom they want to send a message.
- **Network Delivery Mailbox** – users are only required to enter the called party’s extension or mailbox number. The system recognizes the dialed extension or mailbox and automatically outdials the complete 10-digit number.

AMIS is a proven analog networking application; however, there are a couple of shortcomings with AMIS:

- AMIS is analog and uses the public network. The quality of a message diminishes with every hop the message takes to get to its destination, so messages can sometimes be hard to understand.
- Because messages are transported over the public network, they incur toll charges each time they are sent outside of the system’s LATA.
**Voice Profile for Internet Mail (VPIM)**

VPIM meets two networking requirements. First, it provides digital message networking in a BCM-only network and also allows BCM systems to message network with Norstar Messaging systems and Communication Server 1000 systems that are equipped for VPIM networking.

Secondly, VPIM is a message networking specification that allows voice and fax messaging across different vendors’ messaging systems over the Internet. The Internet uses Transmission Control Protocol/Internet Protocol (TCP/IP), which means that the messages are sent over the Internet in digital format.

One of the benefits of VPIM networking is that since messages are sent in digital format, they retain their original voice quality regardless of how many times the message may be forwarded. And perhaps the most important VPIM benefit is that messaging over an IP Network Internet connection means the messages are sent at no cost to the sender.

**Fax Overflow and Fax On Demand Mailboxes**

If the fax machine cannot answer an incoming fax call, Messaging answers the call and temporarily stores the fax message in the Fax Overflow mailbox. Later, when the fax machine is ready to print, Messaging sends the stored fax messages to the fax machine.

The Fax On Demand mailbox allows a user to retrieve stored documents. It contains a greeting and all the documents the user has stored.

**Unified Messaging**

Unified Messaging provides single-point, local or remote access to voice, fax and email messages from a multimedia PC. The BCM system contains a LAN card that connects the system to the customer’s LAN. The LAN provides the access from users’ PCs to their Messaging mailbox so they can play their messages on their multimedia PCs using speakers or a headset for privacy.

The Messaging headers are displayed on the PC screen (as shown in Figure 39) along with any email and fax message headers. Users can see at a glance how many messages they have, as well as certain message details, especially where Calling Line ID is in use. This means users can choose which messages are more important to them and manage their time based on their priorities. Also, with Unified Messaging, users can easily save messages to a folder or file.

Unified Messaging supports 200 clients on BCM200/400. However, an additional 256 MB of RAM (512 MB total) to support more than 100 clients.
The following dialog box appears when a voice message is played. This user-friendly and interactive tool helps users be more productive because it lets them be more efficient and organized in their message management.
Unified Messaging can really benefit people who travel frequently. They can save their voice messages to a folder and, working off-line, listen to their messages using a headset while they are in-flight. Unified Messaging also gives users the option to archive voice messages and store them to specific folders on their PCs.

**Unified Messaging User Interface**

BCM uses the Unified Messaging Client that has been enhanced to support several other popular email packages, including:

- Microsoft Outlook Express 4.0 and 5.0
- Microsoft Outlook 98 and 2000 in Internet Mail mode
- Netscape Messenger 4.5, 4.6 and 4.7
- Lotus Notes 4.51, 4.6, 4.61, 5.0 and 4.x or 5.0 server running on Windows NT
- Qualcomm Eudora Pro 4.02 and 4.2
- Novell GroupWise 5.5 clients and 5.5 Server running on Novell NetWare 5.x.
In addition to expanded email support, the Unified Manager installation process is automated and considerably easier for the customer to install and set up.

**Messaging Feature Codes**

When using Messaging from a telephone, the user must enter a feature code to access the different functions and options. Users do not have to remember the codes, as they can activate these codes by pressing the feature buttons. The feature codes are the numbers that appear in the parentheses below.

**Leave Message Feature Code (Feature 980)**
- Enables mailbox owners to leave a message in another mailbox on the Messaging system

**Open Mailbox Feature Code (Feature 981)**
- Allows mailbox owners to open their Personal Mailboxes
  - All Personal Mailboxes are protected by a password that is established by the mailbox owner

**Operator Status Feature Code (Feature 982)**
- Enables the System Coordinator, receptionist or Designated Operator to set the Operator Status
  - For example, when an Operator is not available, the System Coordinator alerts the Automated Attendant of this information by setting the Operator Status to “NO”
  - Is protected by a password
  - Is used to establish whether a business is open or closed.
    - The GUI equivalent of this code is Operator Manager, the desktop equivalent of functions performed using Feature 982.

**Directory Number (DN) Feature Code (Feature 985)**
- Allows the user to determine the BCM voice messaging Directory Number
- Allows mailbox owners to forward a telephone to Voice Messaging

**Transfer Feature Code (Feature 986)**
- Allows mailbox owners to transfer calls to a mailbox
  - While the call is active (the call is not put on hold), the user presses the memory button where Feature 986 is programmed, then enters the mailbox number where they want to direct the call
  - The caller is then transferred

**Interrupt Feature Code (Feature 987)**
• Enables mailbox owners to interrupt Messaging while a caller is recording a message or listening to the Personal Mailbox Greeting

• Allows mailbox owners to retrieve calls from Messaging and speak with callers who have reached their mailbox

Record a Call Feature Code (Feature 989)

• Allows mailbox owners to record a phone conversation in the mailbox corresponding to the DN of the telephone from which the feature is activated

  • The party who did not initiate the call may hear standard hold tones while the system conferences in Messaging. Both parties then hear the prompt, “this call is being recorded,” followed by a recording beep tone. If more than one extension is targeted to one mailbox, pressing Feature 989 from any of those telephones will record the conversation in the assigned mailbox.

Single Button Call Forward to Messaging (Feature 984)

• Allows mailbox owners to program a single button to forward all calls automatically to voice messaging.

Feature Descriptions

Administration

Table 6.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Backup and Restore</td>
<td>Applies to all system configuration attributes and messages to be backed up</td>
</tr>
<tr>
<td></td>
<td>Allows users to save system data in the event of operational problems</td>
</tr>
<tr>
<td>Create Mailbox Utility</td>
<td>Seeks out all the extensions that do not have mailboxes and creates mailboxes for extensions with the following default characteristics:</td>
</tr>
<tr>
<td></td>
<td>• Identical mailbox and extension numbers</td>
</tr>
<tr>
<td></td>
<td>• The mailbox name (set name)</td>
</tr>
<tr>
<td></td>
<td>• The directory listing, if available</td>
</tr>
<tr>
<td></td>
<td>• “Yes,” for message waiting.</td>
</tr>
<tr>
<td></td>
<td>• Will not create mailboxes if</td>
</tr>
<tr>
<td></td>
<td>• A mailbox with the same number already exists</td>
</tr>
<tr>
<td></td>
<td>• The extension is used by some other mailbox</td>
</tr>
<tr>
<td></td>
<td>• The extension is identified as a voice messaging channel or other “system” extension.</td>
</tr>
</tbody>
</table>
### AMIS Networking

**Table 7.**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Direct Addressing</td>
<td>Allows a user to send a message to any mailbox in the network by entering the parameters required by Voice Messaging</td>
</tr>
<tr>
<td>Network Delivery Mailbox Addressing</td>
<td>Stores the parameters required by Voice Messaging to reach the destination mailbox</td>
</tr>
<tr>
<td>Site-based Addressing</td>
<td>Allows subscribers to send voice messages to remote Messaging systems</td>
</tr>
</tbody>
</table>

### Digital Networking

**Table 8.**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Delivery Options</td>
<td>Network messages can be highlighted as Certified, Urgent or Private</td>
</tr>
<tr>
<td>Direct Addressing</td>
<td>Allows user to send a message to any mailbox in the network by entering the appropriate parameters</td>
</tr>
<tr>
<td>Forward Network Messages</td>
<td>The network delivery mailbox stores the parameters required by Voice Messaging to reach the destination mailbox</td>
</tr>
<tr>
<td>Non-Delivery Notification</td>
<td>When an error preventing delivery occurs, a network Non Delivery Notification message is generated by the intended recipient system.</td>
</tr>
<tr>
<td>Reply to Network Messages</td>
<td>The person receiving a network message can reply and create an outgoing message that is already addressed to the originator.</td>
</tr>
<tr>
<td>Site-based Addressing</td>
<td>The local subscriber can send voice messages to other company locations using the site address which is usually the same as, or similar to, the telephone number of the addressee</td>
</tr>
</tbody>
</table>
**VPIM (Voice Profile for Internet Mail) compatible Voice Messaging**

VPIM is a voice messaging networking standard that allows systems from different vendors, to exchange voice and fax messages, over the internet. The BCM Voice Messaging is compliant with VPIM version 2.0. As of the end of 2001 the following vendors had VPIM compliant products:

- Active Voice
- Alcatel
- AVT Corporation
- MITEL /Baypoint Innovations
- Centigram/ADC
- Comverse Network Systems
- Data Connection
- ELETEL
- Glenayre
- Lucent
- Media Gate
- Microsoft – Exchange
- MJL Korea Ltd.
- Nortel – Nortel Messaging
- Nortel – Business Communications Manager
- Nortel – Norstar Voice Messaging
- Siemens
- UniExchange
- Unisys Corporation
- Voice Data Systems

**Automated Attendant Features**

*Table 9.*

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>AA Menu Prompt</td>
<td>Can be turned on or off for each greeting table</td>
</tr>
<tr>
<td>Call Transfer – Blind</td>
<td>Transfers calls directly to an extension with ringing starting immediately</td>
</tr>
<tr>
<td>Feature</td>
<td>Description</td>
</tr>
<tr>
<td>---------</td>
<td>-------------</td>
</tr>
<tr>
<td>Call Transfer – Screened</td>
<td>Prompts callers to record their name, which is then played to the destination extension. Allows the called party to accept or reject the call without the caller’s knowledge.</td>
</tr>
<tr>
<td>Caller Display (Call Screening Support on Call Forward)</td>
<td>When call forwarding is enabled, all incoming calls are immediately forwarded to voice messaging. When the mailbox owner designates that they want to see caller information displayed at their telephone set, the display will show the name (or number) of the caller as provided by the central office. The information is displayed and accompanied by an alert tone when the call is being forwarded.</td>
</tr>
<tr>
<td>Calling Name Display</td>
<td>Stores Calling Name with the message if the BCM is equipped with CLASS/CMS; the telco delivers Name Display.</td>
</tr>
<tr>
<td>CLID Dialing Table Report</td>
<td>Lists all entries in the Call ID table. Contains, in each entry, a telephone number, destination type and destination number.</td>
</tr>
<tr>
<td>Dial Extension Number from CCR</td>
<td>Allows callers to dial any extension number from any menu point on a Custom Call Routing tree.</td>
</tr>
<tr>
<td>Customized AA Menu Prompt Per Greeting Table</td>
<td>Each greeting table has four default time slots to reflect time of day (morning, afternoon, evening and non-business). Each time slot can have a unique greeting. The business can assign the same greeting to each table or can have unique greetings for each table. The system will support up to 40 company greetings.</td>
</tr>
<tr>
<td>Dual Language System Support</td>
<td>Allows callers and users to switch between two languages at either the Automated Attendant or Personal Greeting level of system prompts.</td>
</tr>
<tr>
<td>External Transfer on Centrex</td>
<td>Allows a multisite company to transfer callers between locations.</td>
</tr>
<tr>
<td>External Link Transfer, Single Trunk</td>
<td>(See Miscellaneous – Single Trunk External Link Transfer)</td>
</tr>
<tr>
<td>Flexible Business Hours for Company Greetings</td>
<td>Allows users to assign greetings to specific greeting tables for time of day and for each day of the week, rather than on a systemwide basis. Allows businesses to pre-record greetings to match business hours.</td>
</tr>
<tr>
<td>Flexible Line Rings for Auto Answer</td>
<td>Transfers calls after a preset number of rings. Allows users to customize the system to meet their individual needs.</td>
</tr>
<tr>
<td>Greeting Tables</td>
<td>Each greeting table can be customized for the line they answer.</td>
</tr>
<tr>
<td>Feature</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>(e.g. sales vs. service, hours of operations). See also Customized AA Menu Prompt per Greeting Table</td>
<td></td>
</tr>
<tr>
<td>Holiday Schedule</td>
<td>Auto attendant schedule can be set up on a weekly basis, from Monday to Sunday, and includes an exception calendar that allows programming for holidays or special occasions. Scheduling can be administered via voice messaging manager or NCM, and it applies to Messaging, Auto Attendant and CCR.</td>
</tr>
<tr>
<td>Military ‘A’ tone detection</td>
<td>The Automated Attendant can detect the military digit ‘A’. When this happens, the call is transferred to a specific Hunt Group Directory Number.</td>
</tr>
<tr>
<td>Multiple Operators</td>
<td>Each greeting table can have its own attendant</td>
</tr>
<tr>
<td>Park and Page</td>
<td>When a caller makes an auto attendant selection, the system can put a caller on hold and page the announcement associated the selection. Alternatively, the call can be transferred to a specific extension. If the call is not answered it can be parked – then the appropriate page is made.</td>
</tr>
<tr>
<td>Remote Administration Menu</td>
<td>Remotely record company greeting</td>
</tr>
<tr>
<td></td>
<td>Remotely set business open or closed</td>
</tr>
<tr>
<td></td>
<td>Allows administrator (using Feature 983) to change any company greeting or remotely set the business status to open or closed</td>
</tr>
<tr>
<td>Routing Calls Based on CLID</td>
<td>Allows the system coordinator to assign up to 100 unique telephone numbers to the Calling Line ID table</td>
</tr>
<tr>
<td></td>
<td>Gives each telephone number a destination type, which determines where the call will be routed</td>
</tr>
<tr>
<td></td>
<td>Is programmed by area code, exchanges, or individual telephone number</td>
</tr>
<tr>
<td><strong>Note:</strong></td>
<td>For the Calling Line ID table to operate, customers must subscribe to these telco Call Display services: Call Line Identification Automatic Number Identification</td>
</tr>
<tr>
<td>Single Digit Menus (CCR)</td>
<td>Available with Custom Call Routing, allows a caller to select a menu option by pressing a single digit</td>
</tr>
<tr>
<td>Time of Day Auto Attendant Blocking</td>
<td>Allows the administrator to prevent dialing of designated extensions during specific hours of the day. When AA is blocked, calls can be routed directly and immediately to associated Messaging mailboxes with other valid CCR tree menu selections unaffected.</td>
</tr>
<tr>
<td>Touch Tone Gate for Auto Attendant/CCR</td>
<td>Allows the system to quickly determine if the caller has DTMF capability and expedite the call if no DTMF is detected</td>
</tr>
<tr>
<td></td>
<td>Eliminates hold time in areas where rotary phones are common, or where the public network does not provide reliable answer supervision</td>
</tr>
</tbody>
</table>
### Feature Description

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transfer Point to an External Number from CCR</td>
<td>Transfers callers to a number outside the BCM system</td>
</tr>
<tr>
<td>Transfer (via Feature 986) of an External Caller to a Specific CCR Tree</td>
<td>Directs callers to a specific CCR Tree</td>
</tr>
</tbody>
</table>

### Fax Messaging Features

*Table 10.*

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Fax</td>
<td>Lets users send and receive faxes</td>
</tr>
<tr>
<td>Fax Messaging</td>
<td>Allows users to store incoming faxes as “fax messages” in their Messaging mailbox</td>
</tr>
<tr>
<td>Fax Answering</td>
<td>Transfers fax calls to a designated fax extension on the BCM via the Automated Attendant</td>
</tr>
<tr>
<td>Fax Overflow</td>
<td>Temporarily stores fax messages if the fax machine cannot answer an incoming call</td>
</tr>
<tr>
<td>Fax On Demand</td>
<td>Allows a user to retrieve documents stored in Voice Messaging in special mailboxes</td>
</tr>
<tr>
<td>Fax Broadcasting (Group Message)</td>
<td>Contains a greeting and all the documents the user has stored</td>
</tr>
</tbody>
</table>

### Group Lists

*Table 11.*

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Group Distribution Lists</td>
<td>Allow users to deliver the same message to a group of users by entering only one address destination or distribution list number</td>
</tr>
<tr>
<td>Max. Group Lists: 99</td>
<td></td>
</tr>
<tr>
<td>Max. Members to a Group List: 999</td>
<td></td>
</tr>
</tbody>
</table>
## Mailbox Features

*Table 12.*

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Assigning Target Attendants</td>
<td>Allows mailbox owners to assign an extension as their dial-0 set</td>
</tr>
<tr>
<td>Auto Answer with Personal Greeting</td>
<td>Answers calls with mailbox owner’s Personal Greeting after a preset number of rings</td>
</tr>
<tr>
<td>Automatic Reply to Internal or External Messages</td>
<td>Enables mailbox owners to automatically reply to a message with one keystroke where CLID and ANI are used</td>
</tr>
<tr>
<td>Broadcast Messages</td>
<td>Enable System Coordinators to record a message and send it to every mailbox. Play automatically and are then erased as soon as the subscriber ends the session.</td>
</tr>
<tr>
<td>Called Party Cancellation of Off-Site Notification</td>
<td>Allows the party receiving a remote notification call to turn off notification to this destination. Is useful when a subscriber enters an incorrect destination telephone number. When the called party cancels notification the system removes the number from the subscriber message notification destination list.</td>
</tr>
<tr>
<td>Cascading Off-Premise Message Notification</td>
<td>Allows mailbox owners to program five internal or external numbers: they can be notified when a message is received in their mailbox. Each number is called in sequence if the preceding number does not answer. Numbers can be designated as a phone, pager or intercom. Note: Depending on the Class of Service programming, each number can be called up to nine times at intervals of 5, 10, 15 or 30 minutes per attempt. If a pager is notified, the user must phone in to receive the message. If a phone is notified, users can access their mailbox once they enter their password.</td>
</tr>
<tr>
<td>Enable or Disable General Delivery Mailbox</td>
<td>Can be disabled or enabled in System Administrator’s Mailbox</td>
</tr>
<tr>
<td>Envelope Information</td>
<td>By pressing “7” during or after a message, mailbox owners are notified of its receipt time and date. Provides the sender’s name for internal calls</td>
</tr>
<tr>
<td>Feature</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Express Internal Messaging</td>
<td>Allows mailbox owners to send internal messages without opening their mailbox</td>
</tr>
<tr>
<td></td>
<td>Automatically includes the sender’s name and extension</td>
</tr>
<tr>
<td>Forward Copy with or without Comment</td>
<td>Allows users to forward mailbox messages to other mailboxes with or without comments</td>
</tr>
<tr>
<td>General Delivery Mailbox</td>
<td>Collects messages after hours, from rotary dial telephones, or for people who don’t have a mailbox. It can be enabled or disabled as required</td>
</tr>
<tr>
<td></td>
<td>When enabled, it allows callers to leave a message. When disabled, allows callers to press zero (“0”) at any time to reach the operator.</td>
</tr>
<tr>
<td></td>
<td>Note: If the operator is not available, the Automated Attendant voice prompts plays. No Keycode required. This is a “last stop” mailbox for unsuccessful call transfers returned to the operator who is, at that time, also unavailable.</td>
</tr>
<tr>
<td>Guest Mailbox</td>
<td>Is useful for people who do not have an extension number, yet need Messaging access</td>
</tr>
<tr>
<td></td>
<td>Can be provided for a mailbox owner’s favorite customer or supplier</td>
</tr>
<tr>
<td>Information Mailbox</td>
<td>Allows businesses to play frequently requested information only</td>
</tr>
<tr>
<td></td>
<td>Does not have message-taking capabilities</td>
</tr>
<tr>
<td>Message Delivery Options:</td>
<td>The following four options increase the user’s control over message delivery:</td>
</tr>
<tr>
<td>Normal, Certified, Private,</td>
<td>Normal: the message is delivered automatically (default)</td>
</tr>
<tr>
<td>Urgent</td>
<td>Certified: the sender receives confirmation when the message is read</td>
</tr>
<tr>
<td></td>
<td>Private: messages cannot be forwarded to another mailbox</td>
</tr>
<tr>
<td></td>
<td>Urgent: a message can be queued to play after broadcast messages, but before “Normal” messages</td>
</tr>
<tr>
<td>Message Waiting Notification</td>
<td>Displays “Message for You” on the user’s set</td>
</tr>
<tr>
<td></td>
<td>Allows the user to hear the number of new and saved messages upon opening his or her mailbox</td>
</tr>
<tr>
<td>Message Waiting Indicator/Visual Ringing Indicator</td>
<td>Red light on the T Series sets indicates that a message is waiting</td>
</tr>
<tr>
<td>Name Confirmation when Sending</td>
<td>LCD display shows the name and number of the called party or mailbox being contacted</td>
</tr>
<tr>
<td>Feature</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Name Directory or Extension Accessibility</td>
<td>Allows users to find any system mailbox extension by spelling the called party’s last name on the dial pad</td>
</tr>
<tr>
<td>Never Full Mailboxes</td>
<td>Allows external callers to always leave voice messages in a personal mailbox, even if the mailbox is full.</td>
</tr>
<tr>
<td></td>
<td>To control misuse of the disk storage space, users with full mailboxes will not be able to retrieve new messages, or create, send, copy or reply to messages until they delete at least one saved message.</td>
</tr>
<tr>
<td></td>
<td>Note: The only time an external caller cannot leave a message in a mailbox is when the system is full.</td>
</tr>
<tr>
<td>Outbound Transfer from Mailbox</td>
<td>Allows callers to press “7,” while listening to a personal greeting, to be transferred to an external number specified by the mailbox owner.</td>
</tr>
<tr>
<td></td>
<td><strong>Note:</strong> The mailbox owner may choose to include this instruction as part of his or her greeting or keep it as a private arrangement for certain callers. When this feature is included in the mailbox class of service, the mailbox owner can turn this feature on and off.</td>
</tr>
<tr>
<td>Personal Greetings Based on CLID</td>
<td>Play to Calling Line ID callers only</td>
</tr>
<tr>
<td></td>
<td>Allow a mailbox subscriber to program up to three specific telephone numbers, each with its own greeting.</td>
</tr>
<tr>
<td>Playback Controls</td>
<td>Allow subscribers to move within or between messages without listening to each message entirely.</td>
</tr>
<tr>
<td>Personal Mailbox</td>
<td>A mailbox can be assigned to a particular person and extension number for his or her exclusive use.</td>
</tr>
<tr>
<td>Prerecorded Greetings Storage</td>
<td>Stores up to 100 prerecorded greetings</td>
</tr>
<tr>
<td>Primary and Alternate Greetings</td>
<td>Allows mailbox subscribers to switch between prerecorded primary and alternate greetings.</td>
</tr>
<tr>
<td>Recovering Deleted Messages</td>
<td>Enables a user to revisit a previously deleted message during a mailbox session and save the message.</td>
</tr>
<tr>
<td>Remote Call Forwarding to Voice Messaging</td>
<td>Allows mailbox owners to turn Call Forwarding to Voice Messaging on or off from a remote location.</td>
</tr>
<tr>
<td>Feature</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Reply Based on CLID</td>
<td>Creates automatic replies to numbers collected from CLASS/CMS or ISDN with ANI Voice Messaging will dial CLID with message by simply pressing the “call” soft key</td>
</tr>
<tr>
<td>Saved Message Queue and Retention Periods</td>
<td>Allows users to save messages for a preset time period as determined by Class of Service; saved messages are stored in a queue and played after any new messages</td>
</tr>
<tr>
<td>System Coordinator Mailbox</td>
<td>Enables the system coordinator to send broadcast messages and use the mailbox for administration</td>
</tr>
<tr>
<td>Urgent Message Notification</td>
<td>Displays an urgent message with the prompt “This message is urgent” Moves urgent messages ahead of non-urgent messages to the front of the new message queue</td>
</tr>
</tbody>
</table>

### Miscellaneous Features

*Table 13.*

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Screening per Set</td>
<td>Requires the caller to enter his or her name Transfers the call to the extension entered via the Automated Attendant, announces the caller’s name and offers the called party the option of accepting the call or transferring it to Messaging Is particularly useful where Calling Line Identification (CLID) information is not available, or when the called party has a set without display capabilities Note: This feature is enabled on an individual mailbox basis from Mailbox Administration</td>
</tr>
<tr>
<td>Enable or Disable the Company Directory</td>
<td>Allows the System Coordinator to enable or disable access to the Company Directory for internal and external users</td>
</tr>
<tr>
<td>Enable or Disable Voice Messaging Feature</td>
<td>Allows a System Manager to globally enable or disable the Voice Messaging feature Note: If Voice Messaging is disabled, the subscriber’s mailbox will not answer the calling party. Instead, the caller will be directed back to the Automated Attendant or CCR application for more options. When disabled, only Feature 980 and Feature 986 will allow callers to leave messages</td>
</tr>
<tr>
<td>External Volume Control</td>
<td>Allows users listening to messages from outside the company to increase the playback volume by pressing the star key (***))</td>
</tr>
<tr>
<td>Feature</td>
<td>Description</td>
</tr>
<tr>
<td>---------</td>
<td>-------------</td>
</tr>
<tr>
<td><strong>Interrupt Feature</strong></td>
<td>Allows mailbox users to retrieve calls that have been forwarded to voice messaging. While a caller is leaving a message in the user’s mailbox, the user can interrupt the message and talk to the caller.</td>
</tr>
<tr>
<td><strong>Multiple Recipients Per Message</strong></td>
<td>Allows users to send messages to multiple recipients with one set of delivery options applied to all recipients. Lets users choose to add recipients or delivery options in any order prior to sending the message.</td>
</tr>
<tr>
<td><strong>Record a Call</strong></td>
<td>While on a call, the user can activate record a call by dialing Feature 989. Voice Messaging will then record the call until the call is disconnected or the user stops recording by dialing Feature 989 a second time. When this feature is activated, both parties will hear “this call is being recorded.” When the recording has been completed it appears as a normal message to the user. The user can listen to it in the same manner as a normal message, forward it, delete it or, in conjunction with Unified Messaging, can “archive” the message as a wave file to a PC.</td>
</tr>
<tr>
<td><strong>Semi-interruptible Greeting (Extended Absence Greeting)</strong></td>
<td>Allows mailbox owners to inform callers of an extended absence. Prompts callers who attempt to bypass it that this is a special greeting and the system will give them the option to play the greeting again. Follows a special tone that alerts callers that it is in effect.</td>
</tr>
<tr>
<td><strong>Single Button Call Forward to Voice Messaging</strong></td>
<td>Lets users deploy Feature 984 to forward their calls to voice messaging. Feature 984 can be programmed onto a button on BCM telephone set, thereby allowing the user to forward call to messaging by simply pressing a key.</td>
</tr>
<tr>
<td><strong>Single Trunk External Link Transfer</strong></td>
<td>Transfers callers externally out of voice messaging without tying up two trunks for the duration of the call.</td>
</tr>
<tr>
<td><strong>Timed Delivery of Messages</strong></td>
<td>Allows a subscriber to create a message and delay delivery of that message until after a specified date and time. Delays message delivery up to the number of days specified in the message retention class of service parameter for a given mailbox. Note: If the Voice Messaging system is using the AMIS protocol for networking messages, timed Delivery messages will follow the standard AMIS rules with respect to call blocking (only urgent messages will be sent during call blocking periods).</td>
</tr>
<tr>
<td><strong>Trunk Answer On/Off</strong></td>
<td>Permits System Coordinators to turn on or off incoming trunk lines programmed for answering by the Auto Attendant.</td>
</tr>
</tbody>
</table>
### Messaging BCM Handbook

#### Feature

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice Messaging Option</td>
<td>May be enabled or disabled at any time</td>
</tr>
<tr>
<td></td>
<td>The default status is enabled</td>
</tr>
<tr>
<td></td>
<td>When enabled, allows callers to access all mailboxes</td>
</tr>
<tr>
<td></td>
<td>Transfers callers who reach a busy extension to the extension’s mailbox</td>
</tr>
<tr>
<td></td>
<td>When disabled, does not allow callers to leave messages in any mailbox unless manually transferred to a mailbox; callers can access Information Mailboxes</td>
</tr>
</tbody>
</table>

### Reports

**Table 14.**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Handling and Channel Usage Report</td>
<td>Provides traffic statistics on the types of calls handled and the traffic against each port used by BCM</td>
</tr>
<tr>
<td>CCR Usage Report</td>
<td>Provides the greeting table from which the CCR tree is currently referenced and a seven-day rolling count of the number of calls received by the tree and the number of times each path is visited</td>
</tr>
<tr>
<td>Numeric Mailbox Information Report</td>
<td>Previously known as the Numeric Subscriber Report</td>
</tr>
<tr>
<td></td>
<td>Includes more information about the mailbox</td>
</tr>
<tr>
<td>Fax On Demand Usage Report</td>
<td>Lists all the Fax On Demand requests and shows the date and time, item requested, delivery fax number and caller CLID</td>
</tr>
<tr>
<td>System Configuration Report</td>
<td>Shows how the system is configured to include the number of ports, outdial channels, group lists and any options that may be installed on the system, such as AMIS</td>
</tr>
<tr>
<td>Report Generation</td>
<td>Various reports can be printed at the request of the system coordinator</td>
</tr>
</tbody>
</table>
## Security Features

*Table 15.*

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Change of Operator Password</td>
<td>A password can be changed from default “OPERATOR” (67372867) to any four to eight-digit sequence</td>
</tr>
<tr>
<td>Centrex Transfer Feature Restrictions</td>
<td>Provides toll fraud protection for Centrex installations using the Centrex Transfer feature to transfer calls to other Centrex extensions. If a BCM has Centrex extensions, it can transfer a call to a Centrex extension. For example, if a caller requests an extension that is not a valid BCM extension, the BCM assumes it’s a Centrex extension and performs a “hook switch flash” transfer to the Centrex extension. However, this presents the opportunity for toll fraud. For example, a malicious caller or user could attempt a transfer to 9011, which results in sending the call to the international operator. The Centrex Transfer restrictions prevent this by specifying dialing sequences that will be denied.</td>
</tr>
<tr>
<td>Forced Password Change</td>
<td>As a security measure users can be forced to change their password. The interval can be set to 0 (interpreted as never having to change the password), 30, 60 or 90 days. If the password has expired, it does not prevent access to the mailbox, but the user is prompted that the password must be changed.</td>
</tr>
<tr>
<td>Double Entry of New Passwords</td>
<td>Prompts mailbox owners to enter a new password twice when changing their password</td>
</tr>
<tr>
<td>Incorrect Password Detection and Lockout</td>
<td>Tracks, through the Voice Messaging system, the number of incorrect login attempts since the last successful login Note: When the number of unsuccessful attempts exceeds a threshold, the mailbox will be “locked out” and cannot be opened, even with the correct password, without administrator intervention. The Class of Service controls the maximum number of login attempts.</td>
</tr>
<tr>
<td>Internal Set Initialization</td>
<td>If this option is selected on initial Voice Messaging installation, mailboxes can only be initialized from a telephone on the same BCM system as the Voice Messaging</td>
</tr>
<tr>
<td>Outbound Calls Restricted to Preset Line/Poo</td>
<td>Allows the administrator to specify which line or pool is to be used for Voice Messaging outgoing calls</td>
</tr>
<tr>
<td>Set Based Restrictions for Outbound Calls</td>
<td>Outgoing calls initiated by Voice Messaging are subject to set based restrictions, regardless of the line pool selected as the outgoing facility</td>
</tr>
</tbody>
</table>

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Voice Over IP (VoIP)

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Call Center
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Voice Over IP (VoIP)

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 Keycode-enabled options that support both IP trunks to communicate with other systems, as well as IP stations, such as the IP Phones 2001, 2002, 2004, and 2007, as well as IP Softphone 2050 and emobility wireless VoIP solutions.

Subscribers can activate a maximum of 60 simultaneous VoIP trunk gateway sessions on the BCM200/400 or a up to 12 on the BCM50 with the purchase of software Keycodes and two additional PEC III DSP processors (BCM200/400 only). The VoIP trunk gateway Keycodes are available in increments of 2, 4, 8, 16 and 32. VoIP stations or clients are available in increments of 1, 4, 8, 16, and 32. The BCM200/400 can support as many as 90 IP stations, depending on the other capabilities planned to be configured. The BCM50 can support up to 32 IP stations regardlous of other capacities. BCM50 does not support IP Phone 2007.

VoIP reduces businesses’ communication costs by routing voice traffic over private Internet Protocol (IP) networks. With VoIP, customers can make telephone calls over any intranet connected to the BCM system. Quality of Service (QoS) in the BCM VoIP is measured through continuous network monitoring and is maintained via a combination of fallback to circuit-switched voice facilities and the use of the differentiated services model and coding.

BCM VoIP uses the bandwidth-reducing technologies of speech compression (using codecs) and silence compression (or suppression). These technologies allow customers to save money by sharing bandwidth with other data traffic.

VoIP for BCM50

Consistent with the existing BCM platforms, the BCM50 is a key component of the Nortel Networks Enterprise VoIP strategy, a portfolio of solutions that deliver choice in voice service deployment with rich multimedia applications to enhance enterprise communications, drive more powerful customer interaction resulting in higher potential revenue, and lower costs by removing barriers of distance and location. Now small businesses can utilize VoIP to consolidate disparate voice and data networks into a single network for all communications needs and manage the network through a single interface.
BCM50 supports the complete range of IP telephony capability offered by existing BCM products. These features are enabled through the use of keycodes and require no additional hardware:

- VoIP Gateway (H.323): Up to 12 VoIP trunks
- VoIP Telephony Clients: Up to 32 VoIP Telephony clients, supporting the range of Nortel Networks Internet Telephones.

When utilizing the two BCM50 main modules with integrated routers (BCM50e and BCM50a), businesses can enable multisite VoIP trunking using secure VPN tunnels, as well as wide area VoIP applications with remote user support.

For BCM50 installations requiring data networked VoIP trunking, BCM50 may function as a DHCP client to a network DHCP server for automated IP address assignment.

### Voice over IP Trunk Capability

BCM VoIP provides least-cost routing of voice traffic through a corporate intranet. BCM VoIP offers the following features:

- Basic calls with answer and disconnect supervision
- Direct Inward Dial (DID) and Direct Outward Dial (DOD)
- Calling name and number
- VoIP to Communication Server 1000 capability
- ITU-H.323v4 compatible gateway
- ITU-H.323v4 gatekeeper interoperability
- Economical bandwidth use through voice compression
- Economical bandwidth use through silence compression
- Quality of Service (QoS) monitoring of gateways
- Circuit-switched voice facilities fallback capability
- MCDN signaling for enhanced communication with other BCMs and/or Communication Server 1000.

### VoIP Features

The BCM VoIP Gateway supports ITU-H.323v4 gatekeeper operation, which allows the configuration of IP address information to be centralized. Three call-signaling protocols are available:
• **Direct routed** – BCM uses a locally maintained table for IP resolution. The locally maintained table is the remote gateway table.

• **Gatekeeper routed** – BCM uses a centralized gatekeeper for address resolution and call setup. In this mode, the gatekeeper handles all call control signaling. The remote gateway table is not used.

• **Gatekeeper resolved** – BCM uses a centralized gatekeeper for address resolution only. In this mode, the BC handles all call control signaling. The remote gateway table is not used.

BCM also offers H.323 interoperability with Communication Server 1000 and RadVision ECS 3.0/3.2 Gatekeeper.

**VoIP Gateway**

The VoIP gateway:

• Allows communication with other supported H.323v4 gateways via system to system (trunk) calls

• Uses Digital Signal Processors (DSPs) for voice coding

• Supports compression algorithms (codecs) G.711, G.723 and G.729A (BCM200/400)

• Supports fallback to circuit-switched voice facilities

• Monitors the data network and establishes new calls via the conventional circuit-switched voice facilities if Quality of Service (QoS) over the data network does not meet a customer set threshold

• Allows the system installer and administrator to determine the acceptable QoS level over the data network for each endpoint

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.
Network Quality of Service

Quality of Service is a measure of the caliber of the service a business provides to its customers. QoS is largely dependent on end-to-end network performance and available bandwidth. It takes on great importance with networks that are carrying both voice and data because voice traffic is extremely bandwidth and delay sensitive.

The following parameters determine the VoIP QoS over the data network:

- **Packet loss** – is the percentage of packets that do not arrive at their destination. Transmission equipment problems, high delay and congestion cause packet loss. In a voice conversation, packet loss is heard as distortion in the conversation.

- **Packet delay** – is the time between when a packet is sent and when it is received. The total packet delay time consists of fixed and variable delay. Variable delay is the more manageable delay, since fixed delay is dependent on the network.

- **Delay variation (Jitter)** – is the amount of variation in packet delay. Jitter affects the receiving gateway’s ability to assemble voice packets received at irregular intervals into a continuous voice stream.

QoS in the BCM VoIP is maintained through network monitoring, fallback to circuit-switched voice facilities and an adaptive jitter buffer implementation.

Network Monitoring

The VoIP in BCM includes a Quality of Service Monitor (QMON) that periodically monitors the delay and jitter of IP networks between two peer gateways by using a proprietary protocol in common with the Meridian Internet Telephony Gateway (ITG RLS. 24). QMON measures the QoS between the local gateway and each of the remote gateways on a continuous basis. The following guidelines apply:

- QoS monitoring is supported for BCM and ITG product locations
- Acceptable QoS levels are set for Tx and Rx directions at each gateway
- Fallback is triggered for a new call if the QoS measurement to the far end gateway is below the preset threshold.

Fallback to Circuit-switched Voice Facilities

This feature reroutes calls to alternate trunks such as the Public Switched Telephone Network (PSTN) until the network QoS improves. When the QoS meets or exceeds the threshold, new calls route over the IP network.

Calls will also fall back if there is no response from the destination, the remote gateway table is configured incorrectly, or if there are insufficient DSP resources available to handle the new call.
If the fallback feature is disabled, calls are sent over the Voice over IP trunks regardless of the QoS. The fallback feature is only in effect at call setup. A call in progress will not fall back if the QoS degrades. A call that has been rerouted through the fallback feature will not revert to the Voice over IP trunk.

**Network Performance Utilities**

Two common network utilities, Ping (Packet InterNet Groper) and Traceroute, provide a method to measure Quality of Service parameters to help with network engineering.

- **Ping** – sends an Internet Control Message Protocol (ICMP) echo request message to a host, expecting an ICMP echo reply to be returned. This allows the round-trip time to a particular host to be measured. By sending repeated ICMP echo request messages, percentage of packet loss for a route can also be measured.

- **Traceroute** – uses the IP TTL (time-to-live) field to determine router hops to a specific IP address. Traceroute can be used to measure round-trip times to all hops along a route, thereby identifying bottlenecks in the network.

Both of these tools are available directly through the BCM Unified Manager.

**Codecs**

The term codec refers to the voice coding and compression algorithm used to convert signals. It is important that all gateways in the intranet support the same codec types. The codec type used on a per VoIP call basis is determined at call setup. The originating gateway will indicate to the remote gateway which codec types it supports, beginning with the preferred order of usage. Depending on its capabilities, the remote gateway chooses one of the codec types and continues with the call.

With BCM’s Media Path Codec Renegotiation and Administration feature, when an incompatibility is detected, the codec is renegotiated and a compatible codec is found, if available. This improves network reliability by eliminating a number of scenarios in which calls are dropped because of codec negotiation failures. In addition, this capability simplifies system administration by eliminating the need to configure the media parameters for all VoIP Gateways in a network identically.

If both ends cannot agree on a codec type, the call fails.

On BCM, the following order is recommended for codec selection:

- **G.711** – provides the best audio quality but uses the greatest amount of bandwidth. This codec delivers “toll quality” audio at 64 Kbps. This codec is optimal for speech since it has the smallest delay and is very resilient
to channel errors. However, it consumes the largest bandwidth. North America uses G.711-LAW and international markets use G.711 A-LAW.

- G.729 – uses less bandwidth, but reduces audio quality. This is the default and preferred codec for Voice over IP. It provides near toll quality with a low delay. This codec uses compression to 8 Kbps.

- G.723.1 (6.3 Kbps or 5.3 Kbps) – uses the smallest amount of bandwidth, but reduces audio quality. This codec uses the greatest compression, 5.3 Kbps or 6.3 Kbps. Available on BCM200/400 only

- G.729 – provides the best balance of quality audio plus bandwidth savings.

Each gateway needs to be configured with the possible codecs that are available for negotiation, as well as the preferred order of usage. Given that the trade-off is quality versus bandwidth, the codecs’ configuration should reflect available bandwidth on the network. Silence compression is supported on G.723.1 and G.729, Annex B.

Using the Unified Manager, you can configure payload size per codec for IP trunks and stations. The payload options are 10, 20, 30, 40, 50, 60, 70, 80 and 90 milliseconds. This minimizes mismatched payload sizes.

**Silence Compression**

Key to the success of VoIP in business applications is minimizing WAN bandwidth consumption. Beyond speech compression, the best bandwidth-reducing technology is silence compression, also known as silence suppression. Silence compression technology recognizes the periods of silence in a conversation and stops sending IP speech packets during those periods. Telco studies show that in a typical phone conversation, only about 36% – 40% of a full-duplex conversation is active. When one person talks, the other listens (this is called half-duplex). There are also significant periods of silence due to the speaker pausing between words and phrases.

By applying silence compression, full-duplex bandwidth consumption is reduced by the same amount, freeing up bandwidth for other voice/fax or data communications. The following figures illustrate how silence compression allows two conversations to fit in the bandwidth otherwise used by one. This 50% bandwidth reduction develops over a 20 to 30 second period as the conversation switches from one direction to another.

To provide a more natural sound, comfort noise is added at the destination gateway during the silent periods to calls where silence compression is active. In some cases, silence compression may cause a perceived degradation in audio quality. Silence compression can be disabled, increasing bandwidth consumption.
Figure 41.

TWO CONVERSATIONS WITHOUT SILENCE COMPRESSION

Hello, Fred, this is Susan

Silence

Silence

Hi, Peter, this is Carol

Between the talk-listen silence (around 50%), and the inter-word and pause silence (around 10%), a speech conversation overall is about 60% silence.

Figure 42.

TWO CONVERSATIONS WITH SILENCE COMPRESSION

Hello, Fred, this is Susan

Hi, Peter, this is Carol

With silence compression, two conversations can share the bandwidth normally used by one conversation, thus doubling transmission efficiency. Data and LAN traffic can also share bandwidth and ride along on the silence of a speech conversation.
Echo Cancellation

When a two-wire telephone cable connects to a four-wire PBX interface or a telco central office (CO) interface, a special electrical circuit called a hybrid is used to convert between two and four wires. Although hybrid circuits are very efficient in their conversion ability, a small percentage of telephony energy is not converted; instead, it is reflected back to the caller. This is called echo.

If the caller is near the PBX or CO switch, the echo comes back so quickly that callers cannot discern it. If the delay is more than about 10 ms, however, the caller can hear an echo. To prevent callers from hearing an echo, gateway vendors include a special code in the DSPs that listens for the echo signal and subtracts it from the listener’s audio signal. Echo cancellation is especially important for gateway vendors because the IP network delay can easily be 40–50 ms, so the echo from the far-end hybrid would be quite pronounced at the near end. Far-end echo cancellation eliminates this.

Echo cancellation sometimes causes choppiness in a low audio conversation. Although echo cancellation can be disabled, it is not recommended.

Non-linear Processing

Non-linear processing (NLP) is part of echo cancellation. It improves echo cancellation by further reducing residual echo. NLP mutes background noise during periods of far-end silence and prevents generation of comfort noise. Some listeners find muted background noise annoying. NLP can be disabled to prevent this muting, but it may cause increased perceived echo.

Jitter Buffer

IP network packet delay and network jitter contribute significantly to reduced voice quality. Network delay describes the average length of time for a packet to traverse a network. Network jitter describes the variability in a packet’s arrival time.

To allow for variable packet arrival time and still produce a steady outgoing stream of speech, the far-end gateway does not send out the speech as soon as the first packet arrives. Instead, it holds it for a certain time in a part of its memory called the jitter buffer and then plays it out. The amount of this hold time is the measure of the jitter buffer (e.g. a 50 ms hold time implies a 50 ms jitter buffer).

As the network delay (the total time, including codec-processing time) exceeds about 200 ms, the two speakers will increasingly adopt a half-duplex communications mode, while one speaks, the other listens and pauses to make sure the speaker is done. If the pauses are ill timed, they end up “stepping” on each other’s speech. This
problem occurs when two people converse over a satellite telephony connection. The result is a reduction in perceived voice quality.

When a voice packet is inordinately delayed and does not arrive at the far end in time to fit into the voice stream going out of the far-end gateway, it is discarded and the previous packet is replayed. If this happens too often or twice in a row, the listener will perceive reduced voice quality.

The jitter buffer hold time adds to the overall delay. If the network has high jitter, callers will hear a long delay in the voice stream. For example, a network might have a moderately average delay of 50 ms with a variability of 5 ms. The network has approximately 5 ms of jitter, a low figure. As the jitter buffer hold time is only 5 ms, the effective network total delay will only be 55 ms, which is still moderate.

On the other hand, assume the network has a low average delay of 15 ms, but 10% of the time the delay goes out to a long 100 ms. Meanwhile, 90% of the time the delay is a brief 4 ms. The jitter buffer would have to be 100 ms and the total network delay would be 115 ms (a long delay). In many VoIP applications, network jitter can be more important than average delay.

BCM VoIP voice calls can be set to use an adaptive jitter buffer that changes the hold time over the duration of the call. The installer or administrator is also able to configure the jitter buffer to fixed values (0, 0.06, 0.12 and 0.18 seconds).

Voice over IP Stations

The IP station portfolio consists of the IP Phones 2001, 2002, 2004, and 2007 and the IP Softphone 2050 software-based phone, in addition to other H.323 client devices such as WLAN.

The IP Phones 2001, 2002, 2004, and 2007 support paging through the set and handsfree intercom. Since the 2050 is a “software phone,” it has no set or speaker and cannot receive pages or hands-free intercom calls. The IP Softphone 2050, however, can originate paging and hands-free intercom calls.

IP stations provide the same functionality as Nortel 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.

Although BCM supports both digital and IP stations, it connects IP to IP calls entirely over IP Networks and is therefore considered a “pure IP” solution.

Note: the IP phone 2007 is not available on BCM50.
Media Path Management on LAN

BCM has the ability to handle management between two IP devices, without being converted to TDM and back to IP. During call setup, an IP station such as an IP Phone 2004 communicates with the BCM Call Server via a signaling protocol called Unistim to determine the destination of the dialed number. If the call is to another IP station, the BCM informs the originating IP station of the destination station’s IP address and the compression to be used (G.711, G.729, etc.), and the call is established directly via IP.

Since the two IP sets communicate directly and not through the BCM, their IP traffic has less impact on the network. Less traffic means less congestion and less latency, which means higher quality. This becomes especially important when we look at examples of IP stations at remote sites.

If the two IP sets are connected to the same LAN switch, the VoIP traffic between them does not impact the rest of the network.

**Media Path Management/LAN**

*Figure 43.*

If the call is to a digital or analog station or to an analog or digital trunk, the call is routed to the TDM switch on the BCM and is connected to the digital station.
Media path management is even more important when there are IP sets at multiple BCM sites. The originating IP set communicates with the BCM call server to determine the location of the called destination. If the destination is another site, the originating BCM call server communicates with the distant BCM call server via H.323 to negotiate the appropriate codecs between sites. The call is established from the originating IP set to the destination IP set directly via IP.
Once again, this is important, as many other solutions convert calls from IP to TDM and back to IP, sometimes several times. Each time voice quality degrades. BCM establishes the call entirely via IP with no degradation or delays. But there are some situations where BCM does convert the call from IP to TDM and back.

Before routing a call across the WAN, BCM makes a decision about the voice quality that can be supported. If the IP WAN is too congested, the BCM will automatically route to the IP gateway and establish the call over the PSTN. If the destination is a digital set at the distant BCM, the originating IP set communicates with the local BCM call server to determine the location of the called destination. The originating BCM call server communicates with the distant BCM call server via H.323 to negotiate the appropriate protocol between sites. The call is established from the originating IP set to the gateway of the destination BCM via IP, where the call is converted to TDM and a connection to the digital set is established. There is still only one conversion to IP and back to TDM.
Media Path Management/WAN

*Figure 46.*

Once again, this is important, as many other solutions convert IP to TDM and back to IP, sometimes several times. Each time there is degradation of voice quality.

Media Path Remote Site

One of the real benefits of IP Telephony is its ability to support satellite offices, where a number of IP sets are supported at another location. The BCM has this capability. The media path management knows that if one remote IP station is calling another IP station at that site in the same subnet, the call will still be placed directly via IP between the two remote IP sets. Only the signaling channel, which requires minimal bandwidth, travels from the remote site to the BCM and back.
Integrated IP Music on Hold

This feature leverages the Internet and the native capabilities of BCM to provide background music for callers while on hold. By connecting to a data source, audio information is passed directly to BCM. Common formats such as .wav or .ra formats are supported. With this feature, customers now have two ways to provide music while on hold: by playing audio from an external source, whether streaming or finite file, or playing audio from files stored directly on BCM. IP Music on Hold is offered on BCM200/400 only.
Voice Networking

In addition to public network connections, BCM can be integrated into an existing private network of BCMs, Norstars and Communication Server 1000 to form a corporate telecommunications network.

BCM uses enhanced signaling on certain trunk types to join Nortel or other manufacturers’ equipment in a private network. Authorized users can also access tie- lines, central office lines and BCM features from outside the system.

Meridian Customer Defined Networking (MCDN) or IP Trunks

Meridian Customer Defined Networking (MCDN) is a Nortel proprietary ISDN-PRI signaling protocol used to interface a BCM to another BCM, a Communication Server 1000 system or a Norstar system.

MCDN is used to network voice-switching capabilities only and, on the BCM provides networking features such as Calling Party Name Display, Network Messaging Services and Message Waiting Indication. (Additional networking features are supported between Communication Server systems.)

Lines/Trunks Used for Networking

External lines provide the physical connection between BCM and other systems in a private or public network. BCM provides the following types of lines for use in networking applications:

- Analog lines
- T-1 trunks (loop, E&M, DID and Ground start)
- PRI trunks
- BRI lines
- IP trunks.

Analog Lines

Conventional analog lines allow basic inbound and outbound voice connectivity to the telco central office. The BCM supports loop start lines and support disconnect supervision. All analog trunk interfaces support Calling Line ID/CLASS features when enabled by the telco central office.
T-1 Trunks (Loop Start, E&M, DID, Ground Start)

T-1 is the standard for digital transmission in North America and is 1.544 Mbps digital circuits partitioned into 24 talk paths or channels of 64 Kbps each. T-1 trunks are used for connecting networks across remote distances. Ground start trunks work with T-1 only. DID and E&M signaling is also supported over T-1.

Digital Drop and Insert MUX (DDIM)

Universal T-1 Trunks

These trunks support both TDM voice and data over the same T-1. Voice is assigned to a fixed number of 64Kbps channels and data is a fixed number of 64 Kbps channels, not to exceed 24 channels total. By combining both voice and data on the same T-1, businesses can potentially eliminate a second T-1 and realize substantial monthly savings. The Digital Drop and Insert Mux (DDIM) module supports this application. The DDIM module passes the data to the BCM internal router or external routers through a variety of different V.35 cables.

PRI Trunks

Primary Rate Interface (PRI) Trunks are T-1 Trunks with Integrated Services Digital Network (ISDN) PRI signaling. They are auto-answer trunks and provide users with incoming and outgoing access to an ISDN network. PRI lines are set to auto answer by default and cannot be changed.

PRI is a fast, accurate and reliable means of sending and receiving data, images, text and voice information. PRI lines allow for faster transmission speeds and the addition of a variety of powerful business applications, including remote LAN access, videoconferencing, file transfer and Internet access.

BRI Lines

Basic Rate Interface (BRI) lines provide users with incoming and outgoing access to an ISDN network. BRI provides two bearer B-channels, operating at 64 Kbps and a data D-channel that operates at 16 Kbps. The D-channel primarily carries call information.

Like loop start trunks, BRI lines can be configured as manual answer or auto answer. For ISDN BRI service, the service provider supplies Service Profile Identifiers (SPIDs), Network directory numbers (Network DNs), Terminal Endpoint Identifiers (TEIs) and other information as required for programming the BCM, TE and other ISDN equipment.
BCM supports the BRI S/T interface only. Typically, North American telcos deliver BRI via BRI U interface. Third party NTIs are available to convert the telco-provided BRI U interface to a BRI T interface that the BCM can accept.

**IP Trunks**

IP Trunks support Voice over IP infrastructure and use standard H.323 signaling.

**DPNSS**

Digital Private Network Signaling System (DPNSS) is a networking protocol that gives users limited number of markets.

DPNSS Trunks are E-1 circuits with DPNSS signaling. When installed in a network, these trunks offer significant enhancements to BCM networking capabilities, similar to those provided via MCDN. DPNSS is not available in North America; however, it is available outside of North America and requires the DPNSS Keycode to activate.

DPNSS makes it easier to support centralized network functionality within private networks for operators and attendants dealing with large numbers of calls. Its routing capabilities provide businesses with more of the larger-network capabilities without the expense of installing a new system, reconfiguring all the nodes, in addition to having a lot of downtime. Most functionality over DPNSS lines is transparent once the DPNSS is programmed into the system.

Corporate offices that are separated geographically can be linked over DPNSS to other BCM systems, bypassing the restrictions of the PSTNs to which they may be connected. This allows connected BCM systems to function like a private network.

**Networking Applications**

**Dialing Plans**

Two types of dialing plans exist: coordinated and universal.

**Coordinated Dialing Plan**

The user dials an extension to reach any set in the private network. The main advantage of this plan is that any node in the network appears, to the caller, to be on the same node as the caller from a dialing perspective.
Coordinated dialing plans are typically used with a network of systems with a three-to seven-digit dialing access between them.

BCM has a routing feature that allows the user to set up a coordinated dialing plan with other systems in the public network. The goal is to have a networkwide dialing plan where all telephone numbers are unique and uniform.

**Universal Dialing Plan**

The user dials an extension to reach sets on his or her own node. To reach any set in the rest of the private network, the user dials an access code and a location code, plus the extension. The advantage to this plan is that calls on the same node require shorter extension numbers, meaning that the user has fewer digits to dial.

**Access Using BCM**

Callers using BCM can:

- Call directly to a specific telephone
- Select an outgoing line to access a private network
- Select an outgoing line to access features that are available on the private network with authorization
- Select an outgoing central office line to access the public network
- Use all of the BCM features.

**Public Network**

Callers in the public network can:

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

**Private Network**

Callers in the private network can:
- Call directly to one or more BCM telephones
- Call into BCM and select an outgoing tie-line to access other nodes in a private network
- Call into BCM and select an outgoing central office line to access the public network
- Call into BCM and use remote features

Remote System Access to BCM

The remote access feature allows callers elsewhere on the private or the public network to access BCM by dialing directly without going through an attendant. Once in the system, the remote user can access some of the system’s resources. The lines, features and dialing capabilities available to a remote user are determined with their password authorization, when using DISA DN, or by line restrictions.

BCM supports remote system access on the following trunk types:

- Public Network: BRI Network Interface, Analog Loop, PRI
- Private Network: T-1 E&M, T-1 Direct Inward Dial (DID), PRI (DPNSS, Q.SIG)

PRI Networking Using Call-by-Call Services

The following example in figure 6-1 highlights the use of PRI call-by-call services. It shows two offices of a company, one in Toronto and one in New York. Each office is equipped with a BCM and a PRI line and has to handle incoming and outgoing calls to the public network. In addition, employees at each office often call colleagues in the other office. To reduce long-distance costs and to allow for a coordinated dialing plan between the offices, the business uses private lines to handle inter-office traffic.
If call-by-call services were not used, each BCM might have to be equipped with the following trunks (for a total of 28 lines):

- 12 DID trunks needed to handle peak incoming call traffic
- 8 E&M trunks needed to handle inter-office calls
- 8 trunks needed to handle outgoing public calls.

If BCM were using T-1 trunks, then two T-1 spans would be required at each office. Note that the total of 28 lines represents the worst-case value for line usage. In reality, the total number of lines in use at any one time will generally be less than 28. For example, during periods of peak incoming call traffic, the demand for outgoing lines will be low.

With PRI call-by-call services, it is not necessary to configure a fixed allocation of trunks. Each of the 23 lines on the PRI can be used for DID, private tie or outgoing public calls. This consolidation means that it may be possible for each office to use a single PRI span, rather than two T-1 spans. With PRI call-by-call services, the only limitation is that there are no more than 23 calls in progress at any one time and individual limits can be set for each type of call.
How Calls Are Made

Dialing plans are created for each BCM site. The dialing plan at each BCM site is configured to determine the call type based on the digits a user dials. If a user in Toronto wishes to dial a colleague in New York, he or she dials the four-digit private DN (such as 6221). The dialing plan recognizes this as a private network DN and routes the call using tie service with a private numbering plan.

Incoming tie calls are routed to sets based on the digits the network receives; in this case, the network would receive the four-digit private DN. If a user in either location wishes to dial an external number, they dial “9,” followed by the number (such as 9-555-1212). The dialing plan recognizes this as a public DN and routes the call using public service.

Incoming DID calls will be routed to sets based on the trailing portion of the digits the network receives. For example, if a public network user dials an employee in the Toronto office, the network will deliver digits 4167632221. BCM will route the call using the last four digits (2221).

IP Telephony and Communication Server 1000 Networking

Figure 49 shows a private network composed of one central Communication Server 1000 and two smaller sites, with BCM connected over IP trunks through a corporate IP network.

The network could represent a large head office (with the Communication Server 1000) connected to several smaller branch offices. In this example, only the head office has trunks connected to the public network. The branch offices access the public network using IP trunks to the head office. This configuration allows for cost savings by consolidating the public access trunks.

Users at all three locations access the public network by dialing “9,” followed by the public number. For example, a user in the west-end branch might dial 9-555-1212 (for a local call) or 9-1-613-555-1212 (for a long-distance call). The BCM routing table directs these public calls to the Communication Server 1000. Routing tables at the Communication Server 1000 will then select an appropriate public facility for the call.

Users make private network calls by dialing a four-digit private network DN. For example, a user in the west-end branch wanting to call a user in the east-end branch within the private network would just dial 6221.
For simplicity, this example does not show fallback to central office trunking. If the quality of the IP connection were considered too low during the call set-up phase, the call would fail unless QoS Monitoring was turned off for these links.

Public Versus Private Received Digits

Outgoing calls and incoming calls terminated on target lines will support both Public/Private calls for PRI-MCDN, IP-MCDN, PRI-QSIG and PRI-DMS100/PRI-DMS250. Public/Private calls will also be supported on incoming calls that tandem out on another trunk for PRI-MCDN & IP-MCDN.

BCM and a Gatekeeper

BCM supports the use of an ITU-H.323 gatekeeper, third-party software installed on a server to centralize IP address configuration information. Each port on the network is assigned an alias name. Instead of having remote gateway tables on each BCM, the gatekeeper contains the remote gateway table. The alias name is resolved to an
IP address by the gatekeeper. If an IP address changes, only the gatekeeper needs to be updated, as the alias name stays the same.

For example, a caller in Tokyo at DN300 dials DN400. The least cost routing is over the Internet. The Tokyo Communication Server 1000 connects to the destination IP address in its remote gateway table for DN4xx. This address is the gatekeeper. The gatekeeper recognizes DN4xx as a Santa Clara DN. The gatekeeper performs address resolution from the Santa Clara alias name to the IP address 10.10.10.11. The call is connected to the Santa Clara BCM and routed to DN400.

BCM interoperates with the following Nortel products using H.323 IP Telephony standards:

- **Communication Server 1000 Release 3.0**: Includes interoperability of H.323 components when internal DN-to-IP address resolution tables are used, or when the IP Peer Gatekeeper is used for address resolution or call setup. MCDN networking capability with IPT is required only for calls directly between the BCM and the IPT.

- **Communication Server 1000 Release 2.0**: Communication Server 1000 Release 2.0 and the IP Peer 1.0 application. This will require H.323 and MCDN interoperability for both gatekeeper routed and gatekeeper directed calls.

- **Norstar IP Gateway and Radvision Gatekeeper ECS 3.0**: Toll Bypass with VoIP Gateway

Figure 50 shows a private network composed of two BCMs, one in Toronto and one in Ottawa, connected over IP trunks through a corporate IP network. In this network, each BCM has a PRI trunk to the central office and IP trunks to the other BCM. Calls from the Toronto system to the Ottawa system and the Ottawa public network are made over IP trunks with fallback to the PRI trunks when IP trunks are congested.

Congestion occurs if insufficient DSP resources are available for a new call, or if the data network is experiencing delay and packet loss that causes the QoS level to fall below the threshold level set by the system administrator. This configuration allows for cost savings by using the corporate IP network whenever possible, thereby bypassing toll charges that would be incurred by using the public network.

When a call gets rerouted over the PSTN due to congestion, the user will hear a tone and see a prompt indicating that the line has been redirected. The warning indicates that toll charges may be applied to this call.

Users at both locations access the public network by dialing “9,” followed by the public number. For example, a user in Toronto might dial 9-555-1212 (for a local call), or 9-1-613-555-1212 (for a long-distance call to Ottawa). Local calls would be sent directly to the central office over PRI trunks. Long-distance calls to Ottawa would be sent over IP trunks; the Ottawa system would tandem these calls to the local central office over PRI trunks.
Users make private network calls by dialing a four-digit private network DN. For example, if a user in Toronto wants to call a user in Ottawa within the private network, they dial 6221.

**Note:** BCM VoIP gateway requires a Keycode.

*Figure 50.*

The gateway at the Toronto office examines the dialed digits and determines that the call should be routed to the IP address corresponding to the Ottawa office. The Ottawa office receives the call, sees that the leading digit(s) match its private network access code and uses a destination code to route the call over its public trunks to the PSTN.

This is a simplified example where only calls to the 613 area code are routed by the Ottawa node. In a real-world configuration, it would also be desirable to handle area codes that are “close,” for example, Montreal: 514.
MCDN Features

Customers can benefit from the BCM to Communication Sever 1000 networking on either PRI or IP trunks by taking advantage of centralized messaging, centralized trunking and centralized attendant.

Table 16 shows features of MCDN and their primary purposes:

**Table 16.**

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<th>Purpose</th>
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<tbody>
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<tr>
<td>ICCL/TRO/TAT</td>
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<td>Camp-On/Break-In</td>
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**Network Call Transfer**

Network Call Transfer allows the transference and the transfer destination to have each other’s name and number identification when a call is transferred in an MCDN network. The major benefit for a customer using this feature is Caller Line ID (CLID) for calls that are transferred from headquarters.

Figure 51 shows CLID on transferred calls.

**Figure 51.**
Network Call Redirection Information (NCRI)

Network Call Redirection Information (NCRI) adds the ability to redirect a call across a private network and provide the necessary redirection information to the endpoint. The Network Call Redirection Information feature ensures that messages get to the correct mailbox and central messaging point.

Figure 52 shows Set B using NCRI designating the correct mailbox.

Message Waiting Indication (MWI)

The Message Waiting Indication (MWI) feature complements centralized messaging networking because, using PRI or IP, it provides the ability to turn on a message waiting light at the remote sites that use the headquarters messaging system. This capability is supported between BCM and Communication 1000 systems only and is not supported between BCM systems.

In previous releases of BCM, the Business Series Terminals and the Norstar digital telephones (the M7000 series) informed the user that there was a voice message waiting by displaying the “Message for You” prompt on the LCD display.

The BSTs, T7316E, T7208 and T7100, are equipped with a Visual Ringing Indicator.
BCM enables the Visual Ringing Indicator of the Business Series Terminals to also function as a MWI, or a message waiting lamp.

Figure 53 shows the MWI “lights” remote site message waiting lamp.

**Figure 53.**

**ISDN Call Connection Limitation (ICCL)**

In an MCDN network it is possible for a call to be routed through many PBXs and create “loops.”

Endless loops can tie up trunks in a network, but not with the ISDN Call Connection Limitation feature enabled. The ICCL feature limits the number of call connections to prevent endless loops that can tie up the trunks in a network. On an IP call, only the signaling would be looped through many PBXs.

Figure 54 shows ICCL preventing endless loops by limiting call connections. In this scenario, the forwarding of the call would automatically stop once the number of sites in the call forwarding chain exceeded the ICCL limit. In the figures on the following pages, the arrows depict both the media and the signaling for PRI trunks. For IP trunks only the signaling is shown.
Trunk Route Optimization (TRO)

Trunk Route Optimization is an MCDN feature that alleviates unnecessary tandeming and tromboning of trunks by searching for better call routes during the alerting phase of a call. When a call is made, TRO can create a loopback and occupy unused trunks. With the TRO feature enabled, the trunk routes can be optimized and logically connected from branch to branch.

Figure 55 and Figure 56 are examples of TRO that illustrate when Set A calls Set B which is forwarded to Set C. In the first figure (without TRO), the call utilizes two trunks, even when the call is forwarded from the Communication Server 1000 to the second BCM. The second diagram depicts the call path with TRO enabled, showing that the end result is that the call is established directly between the two BCMs, using a single trunk.
Without TRO (2 PRI/IP Links)

Figure 55.

With TRO (1PRI/IP Link)

Figure 56.
Trunk Anti-Tromboning (TAT)

Like the Trunk Route Optimization feature, Trunk Anti-Tromboning is an MCDN feature that alleviates unnecessary and tromboning of trunks by searching for better call routes during a transfer on an active call. The BCM is establishing a logical connection, always optimizing the best paths and routes for trunks.

Figure 57 and Figure 58 illustrate when Set A calls Set B and is transferred to Set C. In the case of a call being transferred from Set B to Set C when TAT is not enabled, two trunks on the same link are utilized: one for the original call from Set A to Set B and one for the second call from Set B to Set C. When TAT is enabled, the TAT signaling allows the originating switch (i.e. the first BCM) to transfer the call internally, freeing up both the outgoing and incoming trunks.

Without TAT (2PRI/IP Channels)

Figure 57.
With TAT (1 PRI/IP Channels)

*Figure 58.*

**Camp-On**

Camp-On allows an attendant on the Communication Server 1000 to camp-on a call to a user already on the phone. When the call is camped on, the user is informed that another call is waiting. The user can then accept the camped-on call by clearing one of his or her already established calls. The called party can also reject the camped-on call by using the Feature Reject code, F814, or the Do Not Disturb feature, F85.

Figure 59 shows the camp-on process. The red arrow depicts the call that Set A is actively engaged in, while the blue arrow shows the call flow for a call where the attendant at the Communication Server 1000 activates the Camp-On feature against Set A at the BCM.

*Figure 59.*
Break-In

Break-In is a feature available to a centralized attendant in a network that allows the centralized attendant to reach a person who is already on a call. Sometimes critical situations require special attention and the Break-In feature lets the attendant politely interrupt if a matter of importance cannot wait for the called party to complete their phone conversation.

Figure 60 illustrates the call flow when the attendant at the Communication Server 1000 uses the Break-In functionality on an active call at the BCM.

Figure 60.

MCDN Tandem E. 164

This feature allows tandeming of Local, National, International and Special call types using a MCDN protocol variant. This includes MCDN over PRI and MCDN over IP.

Support for Messaging between BCM and DMS100

BCM may be connected directly with a DMS100 or SL100 via PRI-DMS100 or indirectly with these switches via a MCDN link with Meridian. The Message Waiting Indication (MWI) over PRI-DMS100 will be the same as over PRI-MCDN.

This supplements other functionality that allows Centralized Messaging with Meridian using MCDN, MCDN over IP supporting Meridian IPT and Communication Server for Enterprise, and BCM-to-BCM and BCM-to-Norstar.
Introduction
Hardware
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> Call Center

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The BCM provides a call center strategy designed specifically to meet the needs and dynamic nature of small- to medium-sized businesses. The Call Center features on BCM enable businesses to increase customer support and customer satisfaction by enabling their customers to speak to the right agent every time.

Call Center for BCM50

BCM50 supports Basic Call Center. Designed for the smaller, information call center, Basic Call Center offers a complete set of agent and call routing features to more effectively address call handling requirements. It provides high value entry-level Automatic Call Distribution (ACD) capability with call routing capabilities and the following features:

- Up to:
  - 10 Active Agents
  - 20 Configured Agents
  - 2 Skillsets
  - 15 Assigned Lines
- 10 Built-in user Recorded Announcements
- Longest Idle and Top-down routing
- Overflow
- Agent Log In/Log Out
- Skillset Mailboxes
- Break/Cancel Break Wrap/Cancel Wrap
- Auto Logout/Auto Make Busy
- Queue Threshold Alerts
- Primary and Secondary Threshold
- Queue Status
- Agent Priority
- Silent monitor of agents with supervisors.
A typical BCM Basic Call Center customer will:

- Have a small-formal, or informal Call Center application
- Not have an immediate need to expand the call center beyond 10 agents and/or two groups.

Basic Call Center is Keycode-enabled and available on a try-and buy-basis.

Call Center on BCM200/400

In addition to Basic Call Center, BCM200/400 also supports Call Center Professional with 50 queues and 80 Active Agents. BCM also offers Silent Monitor for Hunt Groups and Silent Monitor for Call Center. The Call Center applications are pre-installed on the hard drive of the BCM and can be activated with a software Keycode.

Customers can enhance Basic Call Center with the optional Reporting for Call Center Package and/or upgrade to the Professional Call Center. The Professional Call Center includes reporting capabilities. With the Reporting for Call Center Package, both the Basic Call Center and Professional Call Center support an IP wallboard and softboard. BCM50 does not support Reporting for Call Center or IP Wallboard.

BCM Professional Call Center supports a number of advanced capabilities that allow businesses to compete more effectively. Professional Call Center offers the following features:

- Up to:
  - 80 Active Agents
  - 250 Configured Agents
  - 100 Assigned Lines
  - 50 Skillsets
  - 150 Built-in user Recorded Announcements
- Call Transfer
- Overflow
- Agent Log In/Log Out
- Skillset Mailboxes
- Break/Cancel Break (Wrap/Cancel Wrap)
- Make Busy/Auto Make Busy
- Log Out/Auto Log Out
- Primary and Secondary Queue Thresholds
- Queue Status
- Agent Dynamic Priority (on a skillset level)
- Silent monitor of agents with supervisors.

Multimedia Call Center Keycodes can be added to either Professional or Basic Call Center (except on BCM50) and has the following features:

- Web Refresher messages
- Follow me co-browsing
- Text chat
- Screen capture send.

A typical BCM Professional Call Center customer will:

- Have a larger, more formal Call Center application
- Require more than 10 agents and two skillsets

Users of BCM Call Centers generally require more enhanced queuing and routing capabilities than hunt groups or Uniform Call Distribution (UCD) can provide. The Call Center provides agent features, built-in announcements and call routing flexibility not available with hunt groups or UCD.

The Reporting for Call Center Package generates Real-time and Historic reports of the call traffic and performance of a Call Center’s resources. These reports allow management to quickly and easily analyze the efficiency of the Call Center and make changes as required. The Reporting Package can generate Historic reports to provide detailed or summarized information from specific defined time periods.

**Intelligent Integration of Call Center**

The Call Center applications are part of the BCM suite of applications; they are not third-party add-on applications. Nortel has developed and will continue to develop, these applications in unison with the other applications and components of the BCM. Customers and distributors do not have to worry about an application from a third party that gets “up-issued” and causes another tool to not function properly.

The Call Center applications are specifically designed to integrate with the BCM and telephone sets. To ensure easy integration, the Call Center applications:
• Are programmed with the same GUI administration tool used for Messaging. If the distributor, technician or system administrator is familiar with messaging, the Call Center applications are an easy, intuitive next step.

• Use the telephone set’s LCD display to assist users in using features. For example, if an agent presses the “Cancel wrap” key before logging in, a message will instruct the agent to log in first. The functions of the soft keys will also change by activity, bringing up functions when appropriate, thus “teaching” the agent how to use the system.

• Also use the LCD display to show system information and messages such as the type of call (e.g. sales or service).

• Allow agents to log in from any BCM telephone.

• Do not require station sets to become dedicated call center sets when an agent logs in. For example, a logged in agent can still make an intercom call.

• Allow users to program agent functions onto memory keys on station sets.

The Call Center applications can integrate incoming Call Line ID (CLID) directly from the BCM. The CLID information is displayed on the LCD window of the agent receiving the call and can be used in routing decisions.

Professional Call Center and Basic Call Center

Generally, there are two types of call centers:

• Informal

• Formal, or traditional.

The Basic Call Center is targeted at smaller, informal call centers that exist in almost every business. The Professional Call Center, on the other hand, with its greater capacity and more advanced features, is positioned for larger, more formal call centers. An upgrade is available from the BCM Basic Call Center to the Professional Call Center.

The following table highlights the differences between Basic Call Center and Professional Call Center.

<table>
<thead>
<tr>
<th>Features</th>
<th>Basic Call Center</th>
<th>Professional Call Center</th>
</tr>
</thead>
<tbody>
<tr>
<td>Skillsets</td>
<td>2</td>
<td>50</td>
</tr>
<tr>
<td>Configured Agents</td>
<td>20</td>
<td>250</td>
</tr>
<tr>
<td>Agent Ids</td>
<td>20</td>
<td>250</td>
</tr>
<tr>
<td>Active Agents</td>
<td>10</td>
<td>20, expandable to 80</td>
</tr>
</tbody>
</table>

Table 17.
## Features

<table>
<thead>
<tr>
<th>Features</th>
<th>Basic Call Center</th>
<th>Professional Call Center</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of Active Calls in all “Call Center” Skillsets</td>
<td>15</td>
<td>48</td>
</tr>
<tr>
<td>Agent Priorities</td>
<td>20</td>
<td>20</td>
</tr>
<tr>
<td>Dynamic Agent Priorities</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>Maximum Number of Active Calls per ACD Skillset</td>
<td>15</td>
<td>48</td>
</tr>
<tr>
<td>Number of Lines which can be configured (answered) for the Call Center</td>
<td>15</td>
<td>100</td>
</tr>
<tr>
<td>Number of Voice Ports (shared with Messaging dedicated)</td>
<td>12</td>
<td>12</td>
</tr>
<tr>
<td>Number of Routing Tables per Queue</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>Number of Recorded Announcements</td>
<td>10</td>
<td>150</td>
</tr>
<tr>
<td>Number of Steps per Day Routing Table</td>
<td>20</td>
<td>20</td>
</tr>
<tr>
<td>Number of Steps per Night Routing Table</td>
<td>20</td>
<td>20</td>
</tr>
<tr>
<td>Number of Overflow Rules per Queue</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Number of Queue Mailboxes</td>
<td>2</td>
<td>50</td>
</tr>
<tr>
<td>Supervisor Silent Monitor</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Intelligent Routing <strong>Basic</strong> is the system’s ability to route a call to the Operator, Auto Attendant, or Queue Mailbox based on programmable single digit caller input.</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Intelligent Routing <strong>Advanced</strong> provides tremendous flexibility. Callers can be routed to other extensions, skillsets, or external numbers.</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>Intelligent Routing – CLID/ANI</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>Intelligent Routing – DiD/DNIS</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>CLID/ANI or DID/DNIS Priority</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>Intelligent Routing – customer input</td>
<td>Partial – Single Digit</td>
<td>Yes</td>
</tr>
<tr>
<td>Intelligent Overflow allows calls to overflow or move based on preprogrammed conditions.</td>
<td>Available</td>
<td>Available</td>
</tr>
<tr>
<td>Features</td>
<td>Basic Call Center</td>
<td>Professional Call Center</td>
</tr>
<tr>
<td>--------------------------------------------</td>
<td>-------------------</td>
<td>--------------------------</td>
</tr>
<tr>
<td>Call Distribution – Linear</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Line Priority</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Line Priority – Dynamic</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Call Distribution – Longest Idle</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Forced Announcement</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Forced and Manual Call Presentation</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Delay Answer Routing Step</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>No Answer Routing Step</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Disconnect Routing Step</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Agent Not Ready/Make Busy</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Auto Agent Not Ready/Make Busy</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Auto Agent Log Out</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Multiple Language Support</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Queue Status</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Queue Status Threshold Alerts</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Day Mode of Operation</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Night Mode of Operation</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Record a Call</td>
<td>Yes (Messaging)</td>
<td>Yes (Messaging)</td>
</tr>
<tr>
<td>Automatic or Manual Mode of Operation change (Day/Night)</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Day of Week Service, allowing the user to specify the start and end times for the day and night queue for each day of the week.</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Wallboard Support</td>
<td>Yes (With Reporting for Call Center)</td>
<td>Yes (With Reporting for Call Center)</td>
</tr>
</tbody>
</table>

**Call Center Features**

BCM offers a range of Call Center features that greatly increase its flexibility and efficiency. These features can be grouped into the following categories: the routing of calls, agent features, and system features.
# Routing Features

*Table 18.*

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
</table>
| Recorded Announcements        | Play for callers as a step in the Routing Table  
Are user-recorded and can be easily changed to reflect the business’ daily requirements  
Users can record up to 150 announcements and do not require additional equipment  
**Note:** Many competitive systems require that a business purchase a separate device to record announcements. |
| Forced Announcements          | Play from start to finish regardless of whether an agent is available  
Can provide information about products or services, or advise callers to have their customer identification available |
| Simultaneous Announcements    | 16 Recorded Announcements can be played at one time                                                                                                                                                         |
| Skillset Name                 | Is the name programmed to the skillset and is displayed on the agent’s set when he or she receives a call  
Is particularly important when agents are logged in to more than one skillset or when calls overflow |
| Intelligent Routing           | Method for moving a call around the call center based on conditions including how busy the call center is, the availability of agents, the source and destination of the call and the information entered by the caller |
| Intelligent CLID Routing      | Routes calls according to where the originated                                                                                                                                                            |
| Intelligent DID/DNIS Routing  | Routes calls according to the destination or the number dialed by the caller                                                                                                                                 |
| CLID/ANI or DID/DNIS Priority | Allows the administrator to determine the order in which routing decisions are made                                                                                                                           |
Intelligent Caller Input Routing

This feature lets the administrator create rules that route calls based on caller DTMF input. These locations can be skillsets, extensions, mailboxes or external numbers. The caller can enter between 1 and 50 digits; wild cards are supported. The system then matches the input against a table that supports up to 2000 entries. Based on a match, the application then assigns a priority to the call and follows the instructions on routing the call.

For example, callers may be provided with an announcement that asks them to enter their account number. Based on the number the caller enters, the system gives the call a priority and routes it accordingly. These announcements can be provided when the caller initially enters the system or after they have been waiting for some time. In the latter case, the caller might receive the following announcement: “Thank you for your patience. All of our agents are presently busy. If you wish to continue holding, please do so. If you would like to leave a message, press 1. To transfer to service, press 2.”

Intelligent Overflow

This feature provides greater flexibility with respect to the handling of calls waiting in a skillset queue for an agent. It increases the probability that a qualified individual will answer calls in a shorter period of time.

Users can specify that a waiting call:

- Overflows to another skillset and keeps its conditions and original skillset greetings
- Moves to another skillset where the call loses its conditions, becomes part of the new skillset and hears the new skillset greetings
- Transfers to the skillset mailbox
- Transfers to an extension or mailbox
- Transfers to an external telephone number
- Changes in its priority level.

Intelligent Overflow Routing handles calls differently depending on the rules the Call Center administrator establishes. Each rule is based on a mode, one or more conditions and one or more actions.

- **Mode** – refers to a skillset’s mode of operation. A skillset can be in Day mode, Night mode or 24- Hour Service. Intelligent Overflow Routing looks at the skillset’s mode to determine how to handle a call. Each mode can have its own rules for how to handle calls.
- **Condition** – After Intelligent Overflow Routing determines the skillset’s mode, it determines what conditions apply to the call. The two possible conditions are:
• Whether the timer expires. The call center administrator establishes the length of time a call waits for an agent before the call is sent to the overflow destination. The maximum time a call can wait is one hour (59:59).
• Whether agents are logged into the skillset. If no agents are logged in, the call is sent to the overflow.

**Dynamic Agent Priorities**

BCM Professional Call Center supports Dynamic Agent Priorities. The agent priority is based on the agent’s skillset. Agents are prioritized for each skillset they can log in to. This prioritization reflects the agent’s skills in each area. For example, if an agent were particularly skilled in Spanish, he or she would be given a priority of 1 for the Spanish skillset. Similarly, if the agent were less skilled in another area, he or she would be given a lower priority for that skillset (higher numbers indicate less skill). If more than one agent is logged in to a given skillset, Professional Call Center will always route the caller to the agent most qualified to handle the call. BCM Professional Call Center has 20 different dynamic priorities. Basic Call Center does not support Dynamic Priorities.

**Forced/Manual Call Presentation**

This either forces a call on an agent or lets the agent manually answer the call.

**Line Priority**

This feature lets the administrator set a call’s priority based on its incoming line. For example, because a business pays a premium for 1-800 service, these calls can have a higher priority than those coming in on a local line. Therefore, agents will receive 1-800 calls first. However, if the Call Center is swamped with 1-800 calls it is conceivable that local calls would not be answered. To avoid this situation, after a local has been waiting a reasonable time, local calls can have their priority increased ahead of 1-800 calls.

**Line Priority – Dynamic**

Call Dynamic Priority can be used in several locations to increase the Call Center’s flexibility/efficiency as well as customer service. Using the Call Dynamic Priority feature, the administrator can have the priority of a call changed in:

- **Intelligent CLID/DNIS Routing table** – The caller can be assigned a priority for answering based on the source of the call.
- **Intelligent Caller Input Routing** – The priority of the call can be established based on a string of digits entered by a caller. Caller input routing can be used when the call first enters the Call Center or after the call has been waiting for a preprogrammed period of time.
• **Intelligent Overflow Routing** – A call’s priority can be changed when it overflows from one skillset to another.

**Routing Steps**

Routing Steps describe how a call will be routed. These steps consist of actions such as “Distribute,” which occurs when the system attempts to distribute the call to an available agent. If no agents are available, the caller will receive music while on hold until an agent becomes available.

**Table 19.**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Distribution</td>
<td>There are two methods for distributing calls to agents:</td>
</tr>
<tr>
<td></td>
<td>Longest Idle – the system selects the agent who has been available the longest since last handling a call</td>
</tr>
<tr>
<td></td>
<td>Top Down – the system will always select the agent at the top of the agent list first with the highest priority, before moving to the second and so on</td>
</tr>
<tr>
<td>Delay Answer Routing Step</td>
<td>Instructs the Call Center to delay answering incoming calls for a specified time period</td>
</tr>
<tr>
<td></td>
<td>Provides the caller with ringing tones until the delay time passes or an agent becomes available</td>
</tr>
<tr>
<td>No Answer Routing Step</td>
<td>When this command is the first step in the routing table, the call will continue to ring until the caller hangs up</td>
</tr>
<tr>
<td></td>
<td>Typically used after hours</td>
</tr>
<tr>
<td>Disconnect Routing Step</td>
<td>Disconnects callers when they encounter this step</td>
</tr>
<tr>
<td>Skillset Mailboxes</td>
<td>Belong to each skillset</td>
</tr>
<tr>
<td></td>
<td>Allow callers to leave messages for the appropriate skillset</td>
</tr>
<tr>
<td>Expected Wait Time</td>
<td>Plays an audio message for traditional voice-originated contacts, or sends a text message for chat inquires and Web-initiated voice calls.</td>
</tr>
<tr>
<td></td>
<td>The message, whether text or audio, provides the caller with information about how much time they can expect to remain in queue prior to being connected to an agent.</td>
</tr>
<tr>
<td></td>
<td>This feature may be further enhanced by playing the announcement and providing additional options to the caller such as leave a message or transfer to another skill set or location.</td>
</tr>
</tbody>
</table>
Agents Features

The Call Center applications have a variety of features that apply mainly to the agents and supervisors. These features allow the agents to more effectively address the requirements of the customer, the organization and themselves.

Table 20.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
</table>
| Agent IDs | Numbers assigned to each individual  
| | Can have different characteristics  
| | A single employee could have multiple Agent IDs and use the appropriate ID based on the call center’s requirements at the time  
| | Allow agents to log in on any set on the system |
| Agent Log In | Allows an agent to log in to any set on the BCM with an Agent ID and password  
| | At the time of log in, the agent set will display the skillsets that the agent is set up to log into  
| | The agent can then log into all of these or select which skillsets they will log into |
| Break or Wrap (post-call completion timer) | The length of time an agent is provided between calls to process paperwork  
| | Its duration is user-definable (0 – 60 seconds) |
| Not Ready or Agent Make Busy | Notifies the system that an agent is logged in but does not want to receive calls  
<p>| | An agent can activate Not Ready while a call is ringing on his or her telephone. The call is then placed back in the skillset |
| Auto Not Ready or Agent Make Busy | If a call is presented to an agent and is not answered within a preprogrammed time, the system can be programmed to automatically place that agent’s set in the Not Ready mode and return the call to the queue |
| Auto Agent Log Out | Logs the agent out rather than identify the agent as Not Ready. |
| Record a Call | If an agent or supervisor has BCM Messaging, they can use the messaging feature Record a Call to record their conversations. Both the agent and the caller will receive notification that the conversation is being recorded and the recorded call is deposited in the agent’s Messaging mailbox |
| Supervisor Silent Monitor | Allows supervisors to silently monitor the conversation of both agents and callers. Supervisors can monitor both incoming and outgoing calls |</p>
<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
</table>
| **Skillset (Queue) Status**   | Displays the following information to an agent through the LCD on their set:  
Skillset number and status: Enabled, Disabled or Uninitialized  
Number of agents currently logged on to that skillset  
The number of calls currently in the skillset  
Amount of time the oldest call has been waiting in the skillset.  
The wait time appears in minutes and seconds                                                                                                                                                                           |
| **Skillset (Queue) Status**   | **Alerts**  
Allows the agent to receive a visual indication as to how busy the queue is for the skillset the agent is logged in to  
The LCD associated with the Skillset Status feature key displays the following:  
If the LCD is off, all of the calls that are queued to the skillset are within the acceptable wait time  
If the LCD is flashing slowly, at least one waiting call has exceeded the marginal wait time  
If the LCD is flashing quickly, at least one waiting call has exceeded the acceptable wait time                                                                                                                                 |
| **Agent Help**                | Allows a Call Center agent to request help from a supervisor by pressing a programmed telset feature key. The supervisor may choose to accept or decline that request. When accepted, the system invokes a Silent Monitor session so that the accepting supervisor begins monitoring the requesting agent’s current call.  
In the event that the Supervisor is not at their telephone, a message will be left indicating that assistance was requested. Agent information and the time the agent requested assistance is included in the message.                                                                                                                                  |

**Silent Monitor for Hunt Groups**

This feature is part of the BCM core telephony (for both Basic and Professional Call Center) and not part of the Call Center functionality.

This feature allows a non-call center supervisor to monitor Hunt Group members either silently, without their knowledge, or by providing notification with a conference tone, depending on the system programming. Silent Monitoring can be programmed for a pre-defined number of sets. It is password protected and supervisors can activate it by pressing F*550 and entering the silent monitoring password. Once logged in, the supervisor can select the Hunt Group member to be monitored using the DN. If the DN entered is a Hunt Group member and is on an active Hunt Group call, monitoring starts immediately in listen only (muted) mode.
Silent Monitor for Hunt Groups is a way for a business to allow supervisor monitoring of calls between customers and staff when the business has not chosen one of the BCM Call Center options. Only the T7208 and T7316E may be used for initiating silent monitoring.

**Silent Monitor for Call Center**

This feature is suitable for informal and formal call centers. It allows a call center supervisor, whose focus is to aid, monitor and help agents handle calls for a specific skillset, to monitor an agent’s call in a manner that does not alert the agent, or the caller, to the supervisor’s presence on the call.

Silent Monitor for Call Center monitors the telephone set rather than the call. Once a monitoring session of an agent’s telephone set is established, all calls on that set will automatically be silently monitored by the supervisor. That is, when the current call is completed, the monitoring session of that agent will continue. When the agent receives the next call, the supervisor will again be monitoring the audio. Neither caller nor agent will hear the act of setting up a monitoring session. The caller will not hear any hold treatment. The agent will not hear ringing, and no intercom key will be active or lit.

While an agent is logged in, the supervisor can monitor all audio. This monitoring capability includes non-Call Center calls, intercom calls and while the agent is listening to messages. To avoid monitoring of their audio, agents should log out. That telephone set can then be used for all non-Call Center activity and cannot be monitored.

The following sets cannot be used for initiating silent monitoring: IP sets, basic set, IDSN set, portables or analog sets (because they require a two-line display to properly monitor calls.)
System Features

The Call Center has the following system features:

Table 21.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Delegated Call Center Administration</td>
<td>Lets the BCM administrator delegate and give passwords for call center administration to a call center delegated supervisor</td>
</tr>
</tbody>
</table>
| Multiple Language Support            | *Set Display*  
The Call Center telephone set display is available in:  
  - North American English  
  - Latin American Spanish  
  - Danish  
  - Dutch  
  - French  
  - German  
  - Norwegian  
  - Spanish  
  - Swedish  
  - UK English  
  - Italian  |
| Automatic or Manual Mode of Operation change (Day/Night) | Lets the administrator set up skillsets to automatically switch from Day to Night Mode of Operation and Night to Day Mode of Operation at preprogrammed times, or this can be done manually |
Wallboard Support

The Reporting for Call Center software provides wallboard support. The software allows connection of an ITEL IP wallboard to the BCM Call Center. No additional server or software is required. The wallboard can display the following information:

- A summary of activities in the call center for the last half hour or for the current day
- A variety of real-time statistics, such as number of calls in queue, longest waiting call, number of agents available, number of abandoned calls
- Customized messages and alarms, such as number of agents logged in has dropped below ”x” or “service group staff meeting at 10:00 a.m.”
- ipView wallboards available with the BCM with Reporting for Call Center Package. (Not available for BCM50.)

The Reporting for Call Center portion provides more detail on wallboards. Additional information is available at www.itel-business-solutions.com.

Hardware Requirements

BCM Call Center applications are compatible with all BCM hardware platforms. Agents can use the following telephones:

- T7208
- T7316E
- IP Phone 2004
- IP Softphone 2050

Multimedia Call Center

The primary function of traditional call centers is to more effectively handle incoming calls from the Public Switched Telephone Network (PSTN). Multimedia Call Center differs from these traditional call centers, as it permits businesses to tap into the power of the Internet to expand and reach new customers. Multimedia Call center is available on BCM200/400, but not BCM50.

Every day the Web is playing a more significant role in conducting business. Customers regularly search the Internet for information and businesses are increasingly using the Web for sales, service and more. If a business has a Web site or anticipates that they may have a Web site, they should be sure that their communications
solution supports Web contacts. They should have a system that will allow customers who visit their Web site to easily communicate with their business in real time.

**Multimedia Call Center Features**

**Chat or Voice Call**

The Multimedia Call Center application on the BCM allows small or medium-sized businesses to leverage their Web investment by allowing them to voice- and chat-enable their Web site.

The appropriate Web pages can have voice buttons (Multimedia enabled) installed to encourage customers to contact the business’ Call Center. As customers are surfing Web sites, they can click a Multimedia voice button to talk or chat with the appropriate customer service representative. The user is then asked how he or she wants to communicate – by text or by a voice call.

Callers with a separate Internet connection and PSTN voice lines can have a PSTN voice call and/or chat while they view, receive, or send Web pages to agents. Callers who have only one connection for both the Internet and voice are able to use chat but will not have a voice connection.

**Blended Call Treatment**

When a caller clicks the Multimedia Call Center voice button, this action places a request for an agent into the business’ Call Center. The treatment and flow of this request is governed by the rules established in the Multimedia Call Center’s Intelligent Routing. The application blends all contacts regardless of source and media type.

The Call Center administrator establishes the rules for blending all incoming contacts and routes them accordingly. The rules established consider the source, destination, media and resources available to respond to contacts. Requests that come in from the Web can be given priorities, can overflow and have their priority changed in the same manner as PSTN requests.

**Web Messages**

If an agent is not immediately available, the caller can receive periodic HTML messages (Web Refresh) programmed by the Call Center administrator. The messages can thank callers for their interest, inform them that no agents are currently available and tell them that they will be connected to the first available agent. The customer will receive text messages in the same manner as if they had placed a 1-800 call into the business.

As callers wait, they can receive additional update text messages thanking them for their patience and encouraging them to hold.
Web Sync

When the request reaches an agent, he or she is presented with the Web page that the customer was visiting. This feature allows the agent to provide the customer with context-specific support.

Web Page Push/Pull

The agent can push Web pages of information to the caller and the caller can push pages to the agents. This Page Push/Pull capability significantly increases the amount of information that can be shared between customers and agents. Increased customer satisfaction and employee efficiency quickly translates to greater revenue at a lower cost.

The Multimedia Call Center in the BCM supports:

- Click for agent
- Chat
- Page Push/Pull and Follow-me browsing.
- Page Push/Pull:
  - Allows the agent to send pages to the caller as they are discussing alternatives
  - Allows the customer to send pages to the agent. For example, if the customer was considering a competitor’s product, he or she could send the appropriate page to the agent for comments.

The Multimedia Call Center Application has common programming and user interfaces. The programming is the same as the existing Call Center programming that is a natural extension of BCM messaging application. By adding the Multimedia Call Center application, a business can leverage its relationship with the Business Communications Manager and migrate or upgrade customers to services offered by a Web-enabled call center.

Call Center Reporting

Reporting for Call Center is a complete, turnkey solution that delivers real-time displays and historical reports of call center activity. These real-time displays include visual alarms and thresholds levels that enable team leaders to quickly resolve potential problems.

A flexible, Web-based reporting package, Reporting for Call Center does not require the installation of client software. Call center managers can easily point their browser to the Reporting for Call Center Web server and access specific information anywhere on the IP network.
Reporting for Call Center helps manage the peaks and troughs in call traffic. It provides information such as call waiting times, queue length and agent status. Reporting for Call Center provides a comprehensive range of management reports that includes information critical for accurate business planning. In addition, it supports multiple Wallboards, which can be individually configured to display the information that the agents require.

All of this information helps a Call Center to run more efficiently. Having the right information allows a manager to respond to problems in a timely manner.

The Reporting for Call Center functionality is standard with Professional Call Center and optional with Basic Call Center.

The Reporting for Call Center software:

- Provides real-time status displays, current reports and historical reports that can be accessed at either a system level or a skillset level
- Supports 16 multiple wallboards, or unlimited ipView wallboards
- Provides password protection
- Reports information for Supervising, Multiple Agent Logins
- Provides open, standards-based data stream available for third-party call center applications such as Work Force Management
- Allows export of all reports so that they can be worked with in other applications, such as Crystal Reports
- Is consolidated with Hunt Group and IVR data so that customers can leverage the reporting package for a consolidated view of performance metrics.

**Real-Time Status Display**

Real-time agent, group and system information can be accessed through the Reporting for Call Center application. The Reporting for Call Center application stores information about all activity on the system so that it can be analyzed in real-time or as part of a historical review.

**Real-Time Agent Detail – Report Screen**

The Agent Detail screen provides real-time visibility into agent activity. Call center managers can see:

- Agent status of all agents
- Status duration
- Answered calls by hour and day
- Outgoing calls by hour and day
- Answered, non-call center calls

Real Time Call Summary Screen

The Real Time Call Summary screen provides a quick and easy-to-understand overview of agent activity in each skillset. The window displays:

- Number of answered and abandoned calls for the current hour and day
- Call wait times
- Grade of service
- Skillset message waiting indicator
BCM Reporting for Call Center offers two reporting periods: Current and Historical. These reports are accessed through the Management Info menu, which also offers a choice of viewing at system level or skillset level. The skillset level will report activity for each individual line and group for the selected skillset and the system level will report activity for the PSTN portion of the call center. Current Reports provide data for either the current hour or the current day. Reporting for Call Center provides the following reports:

- Answered Call
- Abandoned Call
- Incoming Call
- Agent Activity
- Average time
- Abandoned CLID Report
- System Capacity Report
- Call Profile
- Agent Profile Report
- Summary Report
- Configuration Report.

Historical Reports are available for periods other than the current day. By selecting from a calendar, users can view Historical Reports hourly, daily, weekly or monthly. All reports go as far back as needed and users can store them on diskettes for later retrieval.

**Hardware Requirements of Reporting for Call Center**

Reporting for Call Center has the following hardware requirements:

- IBM-compatible PC
- Microprocessor speed: 120 MHz (dedicated PC), 166 MHz (shared PC)
- RAM: 16 MB (dedicated PC), 32 Mb (shared PC)
- Hard disk drive space for application: 10 MB
- Hard disk drive space for data: 30 MB (per year)
- One free serial port if wallboards are used
- Windows 95/98/2000/XP operating system
- Network Interface Card (NIC)
- TCP/IP protocol
- SVGA display.
Wallboards and Softboards

With the addition of the optional Reporting for Call Center package, BCM Call Center can support up to 16 customized wallboards or unlimited ipView softboards. Each wallboard and/or softboard has a great deal of flexibility and is able to display a variety of parameters, text messages and alarm conditions.

Wallboards (BCM200/400 only)

One of the main benefits of wallboards on BCM is that they do not require additional hardware or software. The wallboards are driven from the Reporting for Call Center application and can be located anywhere on the customer’s LAN. The wallboard configuration is a simple, straightforward extension of the Reporting for Call Center administration.

BCM Call Center only supports wallboards manufactured by Itel. Details on these wallboards are available on the Itel Web site at:

www.itel-business-solutions.com

Information Displayed

Depending on the type of wallboard used, it can display two, three, four or six of the following parameters:

- The number of incoming calls received:
  - in the current hour
  - in the current day

- The number of abandoned calls received:
  - in the current hour
  - in the current day

- The number of outgoing calls made:
  - in the current hour
  - in the current day

- The grade of service provided:
  - in the current hour (%)
  - in the current day (%)

- The number of agents:
  - on outgoing calls
  - on incoming calls
  - available to receive calls
  - in the busy state
  - logged in

Current queue length for number of calls in the queue
Current queue time for the longest waiting
(secs.)

The wallboard will automatically provide a summary for the previous hour and for the current day so far:

- Total number of incoming calls:
  - for the hour
  - for the day
- Total number of outbound calls:
  - for the hour
  - for the day
- Total number of abandoned calls:
  - for the hour
  - for the day
- Grade of service:
  - for the hour of the day

**Wallboard Messages**

Wallboards can display text messages. These messages can be a combination of text characters and call center statistics. For example, when a new record for number of calls handled in an hour has been set, a message could automatically display that information. A message might also display individual birthday wishes.

**Wallboard Schedule**

Messages can be scheduled for display at specific times; for example, they can remind agents to log in at the beginning of the day. Or, they might provide reminders for staff meetings.

**Wallboard Alarms**

These alarms can be set up to send a message to the wallboard when any of the parameters reaches a critical threshold. The system can be programmed so that if any of the parameters is equal to, equal to or greater than, or equal to or less than a customized message can be sent to the wallboard.

For example, if there is a greater than acceptable number of calls in queue, or if the number of agents logged in has dropped below a particular number, an appropriate alarm message can be sent to the wallboard.
**ipView Softboard**

*ipView* Softboard is a Windows software application that provides a soft wallboard, or softboard, on the desktop of a Call Center agent’s PC. This application is included with the BCM Reporting for Call Center software. A site license is provided; therefore, there is no limit to the number of agent desktops that can have *ipView* Softboards installed.

*Figure 63.*

![Table showing real-time statistics](image)

Call Center statistics and messages from the BCM’s Reporting for Call Center application are displayed in real-time by the *ipView* Softboard on the agent’s PC.

Agents will benefit from the real-time Call Center statistics, alarms, messages and summary reports provided on their desktop by *ipView* Softboard. Audible alerts can be associated with certain events and histograms can be displayed to graphically show the changes in a particular Call Center parameter over a rolling period of 15 minutes.

Configuring *ipView* is an intuitive logical extension of configuring the IP hard wallboards. OA & M is accessed through the appropriate tab from BCM’s Reporting for Call Center.

Like the wallboard, the *ipView* softboard can be installed anywhere over a customer’s LAN/WAN and no additional server is required. All the information is provided directly from the Reporting for Call Center.

*ipView* softboard supports English, French and Spanish.

The agent can select the Multiple Viewing formats shown in the following figures.
Parameter View

*Figure 64.*

<table>
<thead>
<tr>
<th>ipView</th>
</tr>
</thead>
<tbody>
<tr>
<td>QL</td>
</tr>
<tr>
<td>AD</td>
</tr>
<tr>
<td>ID</td>
</tr>
<tr>
<td>QT</td>
</tr>
<tr>
<td>SH</td>
</tr>
<tr>
<td>AA</td>
</tr>
</tbody>
</table>

Incoming calls in the current Day

History View

*Figure 65.*

<table>
<thead>
<tr>
<th>ipView</th>
</tr>
</thead>
<tbody>
<tr>
<td>QL</td>
</tr>
<tr>
<td>AD</td>
</tr>
<tr>
<td>ID</td>
</tr>
<tr>
<td>QT</td>
</tr>
<tr>
<td>SH</td>
</tr>
<tr>
<td>AA</td>
</tr>
</tbody>
</table>

Grade of Service for the current Hour

Recent History (Minutes)
Summary View

Figure 66.

Message View

Figure 67.
PC Requirements

Table 22.

<table>
<thead>
<tr>
<th>Component</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Platform</td>
<td>IBM®-compatible PC</td>
</tr>
<tr>
<td>Microprocessor</td>
<td>Pentium® 1 (or equivalent)</td>
</tr>
<tr>
<td>Microprocessor speed</td>
<td>200MHz minimum</td>
</tr>
<tr>
<td>RAM</td>
<td>16Mb minimum</td>
</tr>
<tr>
<td>Free hard disk space</td>
<td>2Mb minimum</td>
</tr>
<tr>
<td>Network Interface</td>
<td>Network Interface Card</td>
</tr>
<tr>
<td>Network Protocol</td>
<td>TCP/IP protocol</td>
</tr>
<tr>
<td>Display Type</td>
<td>SVGA display</td>
</tr>
<tr>
<td>Display (Graphics) Card</td>
<td>SVGA graphics card</td>
</tr>
</tbody>
</table>

Call Center Keycodes

Basic Call Center Keycode

This Keycode option provides basic call center capabilities, including two skillsets, 10 active agents, 20 configured agents, 10 recorded announcements and two Skillset Mailboxes. The Basic Call Center cannot expand beyond 10 active agents by adding Keycodes, but it can be upgraded from Basic to Professional Call Center. Basic Call Center is available on BCM200/400 and BCM50.

Reporting for Call Center Keycode

This Keycode option enables the Reporting for Call Center capabilities for Basic Call Center. Professional Call includes Reporting for Call Center; therefore, this Keycode is not required. The package supports up to 16 supervisor workstations and runs alongside other applications on a customer provided Windows 95/98/2000 capable PC. Reporting for Call Center is available on BCM200/400.

The Reporting for Call Center also contains the ipView Softboard application software. For more details, please see the section on ipView Softboard.

Please note: Reporting for Call Center does not report on the activity of the Multimedia Call Center, as Multimedia Call Center has reports that are included with it.
Basic Call Center to Professional Call Center Upgrade Keycode

This Keycode option upgrades the Basic Call Center to the Professional Call Center, increasing size and core Call Center capabilities. If the optional Reporting for Call Center was previously installed, it will now run with the Professional Call Center and report on the increased capacity.

Basic Call Center to Professional with Reporting Upgrade Keycode

If the optional Reporting for Call Center was not installed prior to the upgrade, customers need to purchase the upgrade with Reporting Software Keycode.

Professional Call Center Keycode

This Keycode option enables the Professional Call Center. It provides greater Call Center capacity, such as 50 skillsets, 80 active agents (shipped with 20), 150 recorded announcements, 50 queue mailboxes and increased capabilities. The Professional Call Center is initially shipped with 20 active agents. The maximum configuration of 80 active agents is available by adding Keycodes. Professional Call Center includes the Reporting for Call Center Package. Available on BCM200/400 only.

Multimedia Call Center Keycode

This Keycode option enables the Multimedia Call Center, which can be added to both Basic and Professional Call Center. When Multimedia Call Center is enabled, all existing enabled call center agents can have multimedia capabilities. Any agent Keycodes added after Multimedia Call Center has been installed can be used as multimedia agents. Available on BCM200/400 only.

Agent Keycodes

Agent Keycodes can be added in increments of 1, 4, 8, 16 and 32, up to the Maximum Agent Keycode, which allows 80 agents to be activated. One Keycode adds one agent, four Keycodes add four agents and so on. Agent Keycodes can only be added to the Professional Call Center.
> **Interactive Voice Response (IVR)**

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Interactive Voice Response (IVR)

Nortel Portal Solutions portfolio is a complete suite of advanced voice processing and speech technology products and services. BCM provides advanced IVR self-service functionality, leveraging the proven technology and market leadership of these capabilities. The Media Processing Server 100 (MPS 100), with its rich feature set, can be preinstalled as part of BCM200/400.

IVR Features

The IVR on BCM is the Media Processing Server 100 (MPS 100) from Nortel Portal solutions. It is a compact, aggressively priced IVR system designed specifically for the small- to medium-sized contact center environment. This advanced solution, embedded in the BCM platform, provides support for several powerful technologies designed to enhance the efficiency of any business, including CTI, browser-based access to traditional IVR applications and remote system management.

Flexibility Built in for Multiapplication Environments

Offering both power and flexibility, BCM is ideal for a diverse range of communications environments. Whether the organization is interested in simple information delivery services or complex call processing applications, this cost-effective solution is a logical choice. The application processor can run multiple applications simultaneously while connecting to multiple databases, using a variety of protocols.

Designed for Smooth Migration and Growth

The BCM provides seamless integration of the new IVR features and technologies, while protecting a business’ current investment in application software, systems platform operations and support training. All BCM features and functions are included in the base unit. Functionality, including IVR, can be added through the addition of Keycodes. As the IVR grows, a business can add additional ports with Keycodes. No additional hardware is required.

The BCM is a converged solution for enterprise level telephony, data and applications in a single platform. IVR is one more high value application that is included on the BCM. Integration is part of the design and has been proven in our labs.
Professional Services

For assistance in implementing IVR and creating a custom solution, call the highly skilled Nortel Voice Portal Professional Services Organization (PSO). Around the globe, Nortel Professional Services Organization can complement a customer’s in-house experts with professionals who have the in-depth technical knowledge and practical experience to turn their broad strategies into specific implementations, or they can be engaged to supply a complete turnkey solution.

PSO is ready to assist business customers with every facet of customization, planning and project management. Our business consulting services include: needs analysis, assessment of current processes and technologies, identification of management goals, step-by-step implementation plans and calculation of their investment payback. We can also help with application development and system integration, as well as managing implementation through various milestones, including quality control, final testing and administrative training. Whatever the scope of a business’ requirements – telephone or on-site assistance, a traditional maintenance program, a customized test plan or comprehensive system design and integration services – our Professional Services team has the resources to provide a complete, highly customized solution.

Nortel Customer Contact and Voice Portal Solutions

IVR on BCM is part of the broad range of Customer Contact and Voice Portal Solutions that Nortel has designed to help businesses increase customer loyalty and improve profitability.

These innovative solutions reflect a broader-based, fully integrated approach aimed at helping customers do business – anywhere, any way and anytime.

Having established more than 30,000 contact centers worldwide, Nortel has the business and technological expertise within our Customer Contact and Voice Portal Solutions team to create a scalable, flexible and resilient solution that will grow and change with any organization. By offering a complete portfolio, we can help businesses achieve their objectives quickly and effectively. We can also help them better integrate contact center strategies into the company’s overall operations by giving them the tools to manage and understand customer relationships more effectively, and to maximize the return on those relationships. And, backed by our Customer Contact and Voice Portal Solutions team’s global reputation for quality and reliability, we offer a single, responsive point of contact for all of their service needs.
Standard Features

IVR on the BCM consists of two components: the IVR Run Time Engine (RTE) and the Application.

Run Time Engine

The Run Time Engine (RTE) is the application enabler that provides the basic functionality required by all IVR applications. It must be able to:

- Understand and execute the IVR application
- Interface with the telephony services by providing instructions and receiving and processing input (e.g. DTMF)
- Provide databases access
- Access and process information for other medias (e.g. fax).

The BCM IVR RTE will:

- Answer calls from the Messaging Auto Attendant
- Speak prerecorded prompts
- Collect input via DTMF
- Retrieve customer information from commercial databases
- Speak results back to the caller using prerecorded prompts.

BCM has the IVR RTE preinstalled on its hard drive. It is a Keycoded option that is sold in channel or port increments with a maximum capacity of 24 IVR channels/ports.

The Application

The Application is the second building block of an IVR solution. It turns the runtime engine functionality into the features and functionality that the customer uses. The application is developed for specific customer needs and, in many cases, is integrated with databases to enable real-time queries and updates. Some examples are:

- A financial IVR application allowing customers to access their accounts, receiving real-time account balances and moving funds between accounts
- A scheduling IVR application could allow a customer to book a tee time on a golf course, review existing bookings and or change bookings
• A retail IVR application could allow customers to check the availability of merchandise, make purchases and check the delivery of existing orders.

The Nortel Portal Solutions Professional Services Organization has significant experience having developed applications in virtually every vertical segment. To supplement this experience, there is a suite of application development tools and training available. The application development and system management tools provide a totally graphical environment for the entire life cycle of the system, including design, implementation, test, operation and modification.

Figure 68.

System Development Tools

• PeriProducer (runtime) – Graphical application development environment

• PeriStudio – GUI-based prompt/speech recording and development facility

System Management Tools

• PeriView – GUI-based tools for administration, monitoring and control of application, ports and nodes. This feature includes a host of separate but integrated applications for viewing and controlling individual or multiple systems. They include:
  • Application Manager – distribute and activate applications
  • Activity Monitor – for applications and circuits
  • Alarm Viewer -- system and application alarms
• File Transfer – moving files in a network
• Task Scheduler – automating activities
• SPIN – diagnostic tool.

**IVR and BCM: Other Applications**

The IVR application on the BCM can leverage all telephony features and other BCM applications. Calls come in from any telephony interface support on the BCM and callers can be transferred from one application to another. Calls can come in to the BCM’s IVR and routed to the Call Center and then Messaging. Or, they can start with Messaging Auto Attendant and go to the Call Center and then IVR and some other combination.

**Media Processing Features**

BCM IVR offers the following media processing features:

• Virtually unlimited prompt and/or message length
• Optimized/minimum concatenation for speech output
• Prompts/messages may be recorded in a studio, or locally, or over the phone
• Caller message recording with random message retrieval
• Call simulator for volume testing and capacity planning

**Database Support**

The following databases are generally supported by IVR on BCM; however, due to the dynamic nature of software, database compatibility should be confirmed through Nortel PSO.

• Oracle
• Sybase
• MS SQL Server
• MS Access
• DB2

Both ODBC access and native access are supported for the above databases. In addition, the IVR software will support host access via TN3270 or TNVT100 terminal emulation.
Nortel's Professional Service Organization can do and has done other integrations based on Tuxedo, Corba and even PDX.

**IVR Park & Page**

This feature allows customers to reach a live person if needed. It puts a caller on hold and automatically pages the department the caller selected from the IVR. Park & Page is also available on Auto Attendant, as is the ability to record holiday greetings in advance.

**IVR Messaging Integration**

This feature allows IVR callers to be sent to a Messaging mailbox and then sent back into the IVR script for further call options.
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Data Capabilities
Messaging
Voice Over IP (VoIP)
Voice Networking
Call Center
Interactive Voice Response (IVR)

> Mobility Solutions

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Mobility Solutions

BCM offers solutions for workspace and workplace mobility that provide wireless functionality without losing the benefits of the wireline system. Users can publish one telephone number and receive all calls on both their desk set and their portable.

For customers who require workplace mobility with roaming capabilities, BCM supports the Wireless LAN (WLAN) 2200 Series portfolio. Support for WLAN 2200 is included in the base BCM software. Each wireless handset requires an IP Telephony Client software license, available in increments of 1, 4, 8, 16, and 32.

For customers who have not migrated to an IP-enabled or a pure IP environment, BCM also supports a digital wireless solution in the form of the Digital Mobility Solution or the T7406 Cordless Telephone. The Digital Mobility Solution offers customers full roaming wireless mobility to provide an immediate or future multi-cell digital mobility capability, whereas the T7406 Cordless Telephone is a full-featured, multi-line digital telephone for businesses that would benefit from a desk-centric mobility solution. It covers an area of up to 282,000 square feet and supports 1-6 employees, enabling employees to be more productive while moving about the office.

Digital Mobility Solution

The Digital Mobility Solution is a modular wireless voice communication system that is based on the Digitally Enhanced Cordless Telephony (DECT) technology. The Digital Mobility Solution is supported on the BCM200/400 and scales from 1 to 64 users and covers an area up to 1.5 million square feet for true campus-wide mobility. The solution consists of the following components:

- Digital Mobility Controller 080 (DMC 080)
- Digital Mobility Controller 320 (DMC 320)
- Repeater (2 channel)
- Handsets (3X16 displays)
- Handset Accessories
- OA&M Package
- Service Tool (H/W & S/W) for handsets and repeaters
- Deployment Tool
**Digital Mobility Controllers**

The following controllers are available within the Digital Mobility offering:

- Digital Mobility Controller 080 (DMC 080)
- Digital Mobility Controller 320 (DMC 320)
The two levels of controllers allow customers to start small and grow their mobility solution to meet their needs today and tomorrow without being charged for additional equipment until the need arises.

The DMC 080 will support up to eight handsets and two base stations. Two DMC 080 modules can be linked together, if incremental growth is required (2 DMC 080 linked together will support 4 base stations and 16 handsets). The DMC 320 will support 32 handsets and 8 base stations. The DMC 320 can also be linked to expand up to 64 handsets and 16 base stations.
The DMC module provides power to each base station via twisted pair cable. The base station can be located up to 4921 feet (1500 m) from the DMC module. The two-channel repeater is used to extend coverage from a base station, up to 50% additional coverage. Up to six repeaters can be registered to a specific base station to extend the range of a base station. The repeaters are linked to the base station in a wireless manner, so that no physical connection is required. Local power is required for each repeater.

The solution does offer an external antenna which can be used in conjunction with a repeater to extend a base station signal up to 0.62 miles (1 km) away. A typical application would be a campus with multiple buildings requiring coverage.

**Table 23. Coverage Area**

<table>
<thead>
<tr>
<th>Digital Mobility Solution Configuration</th>
<th>Maximum Number of Base stations</th>
<th>Maximum Number of Users</th>
<th>Maximum Number of Simultaneous Calls</th>
<th>Maximum Coverage: Square Feet</th>
<th>Maximum Coverage: Square Meters</th>
</tr>
</thead>
<tbody>
<tr>
<td>DMC080</td>
<td>2</td>
<td>8</td>
<td>8</td>
<td>180,000</td>
<td>20,000</td>
</tr>
<tr>
<td>DMC080 + DMC080</td>
<td>4</td>
<td>16</td>
<td>16</td>
<td>360,000</td>
<td>40,000</td>
</tr>
<tr>
<td>DMC320</td>
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<td>32</td>
<td>720,000</td>
<td>80,000</td>
</tr>
<tr>
<td>DMC320 + DMC080</td>
<td>10</td>
<td>40</td>
<td>40</td>
<td>900,000</td>
<td>100,000</td>
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<tr>
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<td>16</td>
<td>64</td>
<td>64</td>
<td>1,440,000</td>
<td>160,000</td>
</tr>
</tbody>
</table>

**Digital Mobility Base Stations**

Each Digital Mobility base station supports up to four simultaneous calls and provides up to 5,000 square meters of coverage. One base station can support up to six repeaters to further extend the Digital Mobility coverage.

**Figure 72. Digital Mobility Base Station**
Digital Mobility Repeaters

Each Digital Mobility repeater provides two channels and supports four simultaneous conversations while extending coverage from a base station by providing up to 50 percent additional mobility coverage. Up to six repeaters can be registered to a specific base station in order to further extend base station range. Repeaters are linked to the base station wirelessly, so no physical connection is required. Local power is required for each repeater.

The Digital Mobility solution also includes an external antenna which can be used in conjunction with a repeater to extend a base station signal up to one kilometer away. A typical application would be a campus with multiple buildings requiring coverage.

Figure 73. Digital Mobility Repeater

Digital Mobility Handsets

A portfolio of handsets will be supported with the mobility portfolio. The 7420, 7430 and 7440 handsets will be launched with the original release of the program in early 2005. The 7440Ex will be released mid- to late-2005.

Digital Mobility Handset 7420

The Digital Mobility Handset 7420 is a traditional, highly featured handset that can be optimized for hands-free use and PC integration with a docking station. The 7420 is for use in North America only.
Digital Mobility Handset 7430

A high value and low cost handset fulfilling all basic requirements for a mobility handset. Targeted at industry, warehouse, and retail segments.

Digital Mobility Handset 7440

A classic high-quality rugged handset with a high level of functionality, targeted at office, hospital, and customer support centers. This handset has an IP 54 classification.

Common Handset Attributes

- Large alphanumeric display with backlight (3X16) 7420 only, graphical display for 7430, 7440 and 7440Ex
- Auto Login - Roaming between 4 different systems
- Headset jack (Not included with 7430 handset)
- Loud speaker (Hands free mode) – 7430, 7440 and 7440Ex only
- Telephone book with storage for 80 names and numbers
- Can be set to vibrate mode
- Text messages
- Silent mode (mute all sounds)
- Redial function (last 10 numbers)
- Adjustable volume
• Nine different ring tones
• Key lock
• Microphone mute
• Caller-ID presentation
• Automatic hook-off
• Standby/speech-time: 90/10 hours
• Temperature compensated charging
• Weight including battery is 4.3 ounces (121 grams)
• Size: 5.6” x 1.9” x 1.0” (143 x 48 x 26 mm)

Digital Mobility Software

The Digital Mobility Solution is supported by an easy to use Operations, Administration, and Management (OA&M) software package, a Mobility Software Service Tool, and a comprehensive Deployment Tool.

Operations, Administration, and Management Software Package

The OA&M software package allows the following aspects of the Digital Mobility solution to be managed:

• Digital Mobility Controller administration
• Handset registration and subscription
• System backup and restore programming information
• Statistical package for debug
• Broadcasting test messaging
• Updating base station firmware
• Remote system administration via series-IP converter

Mobility Software Service Tool

The Mobility Software Service Tool allows the following aspects of the Digital Mobility solution to be managed:

• Repeater programming
• Handset firmware upgrading
- Handset audio-gain adjustment for noisy environments
- Microphone and loudspeaker gain adjustment

Note: Use of this tool requires a programming cable and service tool handset cradle.

**Deployment Tool**

The Deployment Tool can be used to assist with challenging deployments. For normal deployments a handset can be used to deploy the Digital Mobility Solution. Each handset can be used to perform the following administrative activities:

- General deployment requirements
  - Register master handset
  - Register and subscribe handsets
  - Subscribe multiple systems
- Base station/Repeater registration

The Deployment Tool and the ability of the wireless handsets to provide coverage information provide effective tools for quick, sure installations and coverage with minimal gaps or costly overlapping. The Deployment Tool provides superior deployment to competitive offerings and will ensure a more cost effective, more accurate installation.

**System Wide Call Appearance support**

With the introduction of the new Digital Mobility handsets to BCM, S/W enhancements will be implemented to support System Wide CallAppearance (SWCA) functionality across the new handsets. This enhancement will allow the new handsets to have SWCA call presence assigned to them in a limited manner. The handsets will not have a button/lamp to indicate SWCA call presence but the user will be able to use 3 SWCA features on the new handsets (Feature*520, Feature*537 and Feature*538). For example, the user can invoke Feature*520 to park a call on the next free SWCA. The handset will display the SWCA number the call has been parked on. The user will also be able to retrieve the oldest SWCA call parked (Feature*537) or the newest SWCA call parked (Feature*538).

**Answer DN / DSS Key Enhancement**

This enhancement allows the same key to be used for Answer group or DSS functionality. To re-use the DSS key for Answer DN also if configured.
Wireless LAN 2200

The Wireless LAN (WLAN) 2200 portfolio is supported on the BCM200/400 and provides a secure mobile networking environment, featuring true mobility across the campus, strong encryption to protect the network and intuitive management for control of the wireless environment. The WLAN 2200 Series was designed to seamlessly extend existing wired LAN infrastructure into the realm of wireless. It protects investments made in the existing LAN, delivering all the benefits associated with wireless mobility to the small-to-medium businesses and branch offices.

Wireless Handsets

Nortel WLAN Handsets are simple to use, require minimal training, and are durable enough to withstand the rigors of workplace use. The rugged design has no moving parts or external antenna so there is nothing to break or come loose. A complete set of accessories is available including headsets, chargers, and carrying cases.

Based on global standards for wireless LANs, Nortel WLAN Handsets 2210 and 2211 support traffic over a common wireless network, providing wireless IP telephony communications as well as two-way messaging. To ensure excellent voice quality, the solution allows for voice prioritization using SVP as well as MAC address filtering.

Nortel WLAN handsets are also designed to support emerging standards for wireless Quality of Service (QoS), such as 802.11e or its subset WME (Wi-Fi Multimedia Extension), allowing enterprises to take advantage of improvements in wireless network performance and protection.

WLAN 2200 Series Handsets Key Features

The WLAN 2200 Series handsets have similar features to the Nortel Networks IP Phone 2004 including:

- Login/out & Not Ready
- Secondary DN
- Calling Line ID (CLID/CNPD)
- Call Forward
- Call Park / Call Park Retrieve
- Call Pickup
- DN / Directed / Group
- Call Transfer
• Conference
• Group Call
• Make Set Busy/Not Ready
• Message Waiting Indication / Messaging Access
• Multiple Appearance DN / Single Call / Multiple Call
• Multiple DNs on a Single Set
• Page
• Call Hold
• Last Number Redial
• Speed Call - System

**WLAN 2210 Handset**

The WLAN Handset 2210 is a lightweight, durable handset, specifically designed for mobile workplace use within a facility using the supported Nortel IP Telephony Call Servers and 802.11b Access Points (APs) in a Wireless LAN. The WLAN Handset can receive calls directly, receive transferred calls, transfer calls to other extensions, and make outside and long-distance calls (subject to the restrictions applied in your facility). The WLAN Handset is solely for use on-premises; they are not cellular or satellite phones. It has a standby time of 40 hours and talk-time of up to four hours.

**WLAN 2211 Handset**

The Nortel WLAN Handset 2211 is engineered for demanding environments. The industrial-grade durable design has all the features of the WLAN Handset 2210, in addition to push-to-talk (PTT) functionality which allows broadcast communication between employees, eliminating the need for two-way radios or walkie talkies. The PTT functionality uses IP multicast addresses, requiring that multicasting be enabled on the subnet. To reduce the effects of broadcast and multicast traffic from devices in other network segments, the Wireless LAN can be placed on a separate VLAN or subnet.
Wireless Access Points

Nortel WLAN Access Point 2220 is dual-mode, simultaneously supporting 802.11b (up to 11 Mbps in the 2.4-GHz radio band) and 802.11a (up to 54 Mbps in the 5-GHz band) radio standards. It is Wi-Fi® certified (supporting the 802.11f standard called IAPP-Inter Access Point Protocol) and therefore, will offer roaming from one access point to another, and from one radio mode to another, even in a mixed vendor environment.

Wireless Mobile Adaptor

The Nortel WLAN Mobile Adaptor 2201 is an IEEE 802.11a/802.11b standard compatible Card Bus PC Card, which supports high data rate up to 54Mbps (for 802.11a) or 11Mbps (for 802.11b) over the Ethernet speed. It is easy to install and improves workplace mobility and efficiency.

The WLAN Mobile Adaptor 2201 has the following features:

- High speed for wireless LAN connection, 54 Mbps data rate for 802.11a with enhanced “turbo mode” for extended range or speed up to 108 Mbps. 11Mbps data rate for 802.11b
- Support seamless roaming within the IEEE 802.11a or 802.11b WLAN infrastructure
- Configuration utility
- Auto fallback data rate under noisy environment
- Wireless data encryption with 64-, 128-, 152-bit encryption for security
- Built-in dual diversity antenna
- Firmware upgrade-able by only changing driver

The WLAN Mobile Adaptor 2201 has the following applications:

- Home networking for device sharing
- Wireless multimedia
- Wireless office for extension Ethernet range
- Mobile networking for notebook PC, PDA, Web Pad or Wireless Gateway Build-in Application

Business Series Terminal T7406 Cordless

The T7406 Cordless Telephone is a desk-centric mobility solution, designed for use in a small- to medium-sized enterprise site, branch office or franchise site. These businesses typically require a coverage area of 282,000 feet or less, with one to six users. The T7406 Cordless Telephone enables businesses to deliver better customer service
while making your employees more productive with workspace mobility. The T7406 is the mobility solution for
the BCM50, and is also supported on the BCM200/400 as well.

**Features**

An integrated multiline phone, the T7406 leverages the features of the popular Business Series Terminals
portfolio and has the following capabilities:

- 900 MHz digital spread spectrum frequency hopping
- Operating range of 300 feet (97 meters) with 282,000 square feet coverage
- Backward compatibility with Norstar ICS and BCM applications
- Integration with Norstar and BCM applications and features
- Maximum site density enables six handsets and two wall bases, each supporting three T7406 handsets
- Nickel metal hydride battery delivers up to five hours minimum continuous talk time or 72 hours standby
time
- Accessories include a belt clip, wrist strap and leather pouch for easy portability.

**Accessories**

- Spare battery pack
- Standard and custom leather carrying case.
Figure 74.
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Computer Telephony Integration (CTI)

Computer Telephony Integration (CTI) connects the intelligence of the PC with the power and flexibility of the BCM, providing companies with the ability to turn a desktop computer into a powerful communications tool.

With Microsoft Telephony Applications Programming Interface (TAPI), businesses can increase employee productivity and customer satisfaction by using a variety of CTI applications that combine telephone and computer functionality in exciting new ways. BCM TSP is the interface between the BCM system and Microsoft TAPI that allows companies to use TAPI applications on the BCM system.

Several software components work together with the BCM system: Microsoft TAPI, BCM LAN Computer Telephony Engine (CTE) and the Personal Call Manager application. LAN CTE enables TAPI applications such as Personal Call Manager to communicate between the BCM system and a desktop PC. In other words, BCM LAN CTE works with Microsoft TAPI to allow a telephony application to communicate with and control a telephone.

BCM LAN CTE seats enable the use of TAPI or CTE applications on the desktop. Customers can purchase software Keycodes for LAN CTE seat licenses in increments of 1, 4, 8, 16, and 32 seats. A try-and-buy Keycode is also available for customers who wish to try LAN CTE before purchasing.

TAPI

The Telephony Application Program Interface (TAPI) is a standard program interface that enables users and computers to communicate, via telephone or videophone, with people or phone-connected resources elsewhere in the world.

TAPI converts industry standard commands received from Windows via the BCM TSP into a format that BCM LAN CTE can understand. It also converts messages from BCM LAN CTE into an industry standard format and gives them to Windows.

BCM LAN CTE

BCM LAN Computer Telephony Engine (CTE) is the interface between the BCM system, BCM TSP and Microsoft TAPI. This interface allows users to run TAPI or CTE applications on the BCM system. LAN CTE is pre-installed on the BCM and replaces the TSP on earlier releases. LAN CTE is easier than TSP to implement and does not require the presence of an NT Domain to function.
LAN CTE Client allows CTE applications to function in the same capacity as CTE applications written for Norstar ACCESS 3.1. TSP software is installed as part of the LAN CTE Client installation. Third party applications extend the customer’s range of tools, enabling users to improve productivity. LAN CTE provides an Application Program Interface (API) for third party applications to function on PCs on the same network as the BCM and allows customers to implement customized solutions for their business. Examples of this are hotel/motel packages or text messaging programs.

Today’s market offers a wide variety of CTI applications that businesses can use in conjunction with the BCM. Customers can choose CTI applications from Nortel, such as Personal Call Manager and from third party vendors of TAPI applications.

The strategy behind the BCM desktop CTI portfolio is to keep our customers’ options open and allow them to choose the solution that best meets their needs. Personal Call Manager, developed by Nortel, is a downloadable application on the BCM system; users can install it and use it after installation of the LAN CTE Client. Its use requires an LAN CTE license.

To access Personal Call Manager or use other CTI (TAPI or CTE based) applications, users must first purchase LAN CTE seat licenses. Although Personal Call Manager resides on the BCM platform, a system administrator must download it and the LAN CTE client to each user’s PC via the browser-based administration tool. This CTI application is supported in a multitasking environment on any Windows 95/98/Me/2000 NT 4 and XP.

The BCM Personal Call Manager provides tools that make call processing easier. For this reason, a user’s telephone must be equipped with the handsfree option or a headset for privacy. When using a handsfree telephone or headset, users can access all of the BCM call manager functions without picking up their telephone handset.

CTI improves productivity by allowing users easy, on-screen access to point-and-click telephone directories. It also allows users to route calls correctly and more efficiently, which translates into faster service and increased customer satisfaction.

**Personal Call Manager**

Personal Call Manager (PCM) is an award-winning TAPI-based telephony application from Nortel. Designed for use on the Windows 95/98/Me/2000/NT 4/XP operating systems, this application operates with the BCM platform alone, or alongside existing applications.
With PCM, a user can access an internal database directory, double click on a name to dial and conference or transfer calls, all with the ease of a mouse. This ease of use eliminates any guesswork and perceived frustration a user may experience over using advanced telephone features. With PCM, users can answer calls, dial, build conference calls and even see call activity – all on the PC screen. When integrated with Calling Line ID (CLID), PCM can “pop” a dialog box to a PC, showing the user the name of the calling party in addition to other associated information that may be stored in the PC database. PCM has two main modes of operation: operation from the Windows task tray and operation from the main application window. Users can perform the most common telephony tasks, including dial, hang-up, transfer and conference, from the task tray icon. This icon also provides access to the last callers/called list and quick dials. For more complex telephony operations as well as access to the personal address book, PCM provides a main application window. The look and feel of this window follows the Windows 95 style guidelines.
**Address Book**

The Address Book allows users to store contact information. Each entry in the Address Book is called a contact. A contact is normally the telephone number of a person or business, but can be any number the user wants to keep track of or store in the PCM. The address book includes:

**Hierarchical tree view of contacts** – users can sort and organize the folders within the Address Book to meet their individual needs. The left side of the window shows the tree structure (i.e. how folders are organized) and the right side of the window shows the contents of the highlighted folder. Users can view the contents as:

- Large icons
- Small icons
- List
- Detail.

The PCM organizes files into “Quick Dials” and “My Contacts.” Extended information on each contact includes:

- Business phone
- Home phone
- Fax
- Mobile phone.

**Personal Call Manager Features**

Users can activate all PCM features with either a keyboard or a mouse. The primary benefit of the PC interface is that users can manage calls visually and handle them easily with “drag and drop” menus. Common features include:

- **Dial** – The user can make a call using task tray or Calls menu, the toolbar, the Address Book or the Quick Dial list.
- **Answer/Hold/Unhold** – PCM notifies the user of an incoming call and the CLID (if the user subscribes to the service). The icon changes as the status of the call changes from active to held.
- **Multiple Calls** – PCM can control several calls at once, equal to the number of lines available on the BCM telephone. All calls display in the main window, but only one can be active at any given time (with exception of a conference call, when two calls are active at once). When a user answers a second call, the current call is automatically placed on hold.
• **Conference** – This type of call connects one user and two others on a single call. A minimum of two lines are required to appear on the BCM telephone. As with the BCM set, when a user places a conference call on hold from the PCM, a conference call placed on hold from PCM puts both of the callers on hold, enabling them only to speak with each other. Splitting a conference call (via the right mouse button) ends the conference, putting one caller on hold and keeping one caller active.

• **Transfer (blind and announced)** – Transferring a call to another BCM telephone is as easy as the click of a mouse. An attendant can transfer a call “blind” (without talking to the recipient of the call prior to transfer), or “announced” (letting the caller know the name of the party whose call the attendant is transferring).

• **Call Forward** – The number to which the user’s telephone is forwarded appears on the Status Bar. Call forwarding the user’s telephone to an external telephone number is not supported when using PCM.

• **Do Not Disturb (DND)** – The telephone will not ring but the line indicator will flash. The dialog box, “You have a new call,” will still appear on the PC screen when the phone is on DND.

• **Calling Line Identification (CLID)** – If the user subscribes to CLID service, telephone numbers of incoming calls will display on his or her telephone set.

• **Call Duration** – Next to the active call icon, the telephone set displays a running count of how many minutes the call has been connected.

• **Last Callers/Call Log** – PCM keeps a call log of both inbound and outbound calls, showing who called/was called and the time the calls were received/made. This is a useful feature to identify callers who might not have left a voice message.

• **Duplicate Contact Records** – PCM notifies the user when he or she enters a duplicate contact record.

• **One-step call release** – If only one call is displayed on PCM and a user requests disconnect, the call will be disconnected without requesting the user to select the appropriate call.

• **Printing capabilities** – The call log, a single contact and the contents of a folder are all print-enabled.

• **Open contact for incoming calls** – A new option will be added to the Tools/Preferences menu to enable a contact record to automatically pop up when an incoming call is answered.

• **Sort capability in Call Log** – Users can sort both the incoming and outgoing call log by any of the fields (name, phone number, date and time or call duration).

### System Requirements

The following minimum system requirements must be met to run the PCM software:

• Windows 95 or greater operating system

• Microsoft TAPI 2.1 or greater
• 8 MB RAM (16 MB recommended)
• 8 MB free disk space
• 486 DX or greater processor
• LAN CTE that supports Windows 95/98/Me/2000/NT 4.0 SP 4 or greater and XP.
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> **Virtual Private Networks (VPN)**

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Virtual Private Networks (VPN)

A virtual private network (VPN) uses a public or shared network as an extension of the corporate intranet. The shared network is usually the Internet, but a shared frame relay network could also be considered a VPN. With the advent of VPN technology, businesses can use public networks to cost-effectively broaden the reach of intranet applications.

Figure 76.

There are three basic applications for VPNs, including

- Remote access outsourcing: Mobile workers and telecommuters access the headquarters by way of a remote access server.
- Extended intranets: VPNs are used as an alternative to leased lines for linking corporate sites.
- Extranets: Connect a corporate intranet to customers, suppliers, contractors and business partners by way of a public IP/FR/ATM network, providing private communications links established between companies to enhance their ability to work together effectively through dynamic network connections.

Virtual Private Networking on BCM50

For sites with under 20 employees, BCM50 is ideal. Businesses can utilize BCM50’s VPN capability to support a small mesh network of BCM50s. Typically, they would be interconnected on secure tunnels for data and VoIP virtual network applications. If a business chooses to use BCM50s within larger sites, a branch BCM50 can be connected to a central office equipped with a centralized VPN server.

Businesses can choose Ethernet or ADSL, depending on what type of data service they currently use—BCM50e is equipped with an integrated Ethernet router, while BCM50a is equipped with an integrated ADSL router. These
BCM50s provide generic Internet access with all the standard router features, including firewall, security, Network Address Translation, and more, for secure access to the Internet. Businesses can therefore adopt typical business applications, such as employee and partner access to information, as well as Web sites.

To enable more advanced routing (to allow work-at-home users full corporate networking capabilities, for example), businesses can select a BCM50 without an integrated router and combine that with a more advanced Nortel routing solution, such as a VPN Router.

BCM50 includes the following features:

- 5 IPSec server-to-server tunnels
- 5 secure PPTP tunnels
- Support for 128-bit Triple DES, 56-bit DES, 40-bit DES encryptions, SHA1, MD5 Authentication
- Connectivity to another BCM or to a Nortel Contivity Extranet Switch

Note: BCM50 does not support IPSec Client terminations

Virtual Private Networking on BCM200/400

For small sites with more than 20 employees, BCM200/400 provide turnkey access to the Internet and private corporate networks, giving employees access to more types of communications, thereby boosting their productivity and ability to communicate. Like the BCM50, BCM200/400 features integrated routing capabilities, as well as features to enable secure data transmission over public networks, such as authentication and encryption services, which allow sensitive corporate information across public WANs instead of leased lines

BCM200/400 includes the following features:

- 20 IPSec server-to-server tunnels
- 16 secure IPSec Client tunnels
- 10 secure PPTP tunnels
- Supports 128-bit Triple DES, 56-bit DES, 40-bit DES encryptions, SHA1, MD5 Authentication
- Enables connectivity to another BCM or to a Nortel Contivity Extranet Switch
VPN Infrastructure

If businesses replace the expensive, long-haul leased lines with local connections to ISPs, they can realize savings on the mileage charges. Since local connections are less expensive, it becomes possible to connect remote offices either with dedicated lines or ISDN dialups.

Remote and mobile employees only have to dial in to a local service provider, saving on the cost of 800 numbers and modem banks. Linking to suppliers, contractors and business partners becomes both quick and inexpensive. As shown in Figure 77, the VPN infrastructure consists of a network composed of users of various locations and access capabilities, using the Internet to connect. BCM can use dedicated or dial-on-demand connectivity for smaller offices to access the Internet.

*Figure 77.*

VPN Service Models

The following three service models describe where the endpoints of the VPN reside:
- Service provider to service provider model
- Service provider to enterprise service provider model
- Enterprise to enterprise service model.

*Figure 78.*

**MOBILE WORKERS**

![Diagram showing mobile workers connected to remote access server](image1)

**TELE-COMMUNITIES**

![Diagram showing tele-communities connected to security server](image2)

**PUBLIC IP/FR/ATM NETWORK**

**SECURITY SERVER**

**HEADQUARTERS**

*Figure 79.*

**SERVICE PROVIDER TO SERVICE PROVIDER**

![Diagram showing service provider to service provider model](image3)

**REMOTE CLIENT**

**RAS**

**IP BACKBONE**

**SECURE IP TUNNEL**

**LAN**

**BCM**

**SERVER**

**SERVICE PROVIDER**

**CUSTOMER PREMISES**

Service Provider to Service Provider Model

Figure 79 shows the service provider to service provider model. In this model, the VPN begins and ends with the service provider. The service provider supplies all the equipment and expertise required to implement the VPN.

*Figure 79.*

This model is often used to support the remote user functionality of VPNs. The remote user dials in to a service provider’s Remote Access Server (RAS) and requests a tunnel to the corporate network. The RAS sets up a tunnel to a router owned by the service provider that connects to the corporate network. Once the tunnel is established, the remote user can communicate with servers on the corporate network.
Service Provider to Enterprise Service Provider Model

In the service provider to enterprise service provider model shown in Figure 80, the VPN begins with a remote access server owned and maintained by the service provider and ends with the BCM owned and maintained by the business. The only difference between this model and the service provider to service provider model is the extension of the secure IP tunnel to the BCM.

![Figure 80. Service Provider to Enterprise Service Model](image_url)

Enterprise to Enterprise Service Model

In the enterprise to enterprise model shown in Figure 81, the service provider is not involved in the formation of the VPN at all. In this example, the VPN begins with client software on the remote PC and ends at a VPN server on the corporate network. This model has the advantage of working with all service providers whether or not they support VPNs.

This model can also be used to support the extended intranet and extranet functions of VPNs. In these cases, the remote client can be replaced with a VPN server on the business partner’s intranet. BCM can support variations of the model with BCM or Contivity between the remote client and the originating RAS.
VPN Security

Because information flowing across a VPN crosses public or shared networks, security is a major concern. VPNs secure data through privacy, authenticity and integrity.

Privacy

Privacy refers to preventing unauthorized people from gaining access to the information by “sniffing” or intercepting it. Encryption provides the best protection against interception of information and can guarantee the privacy of the communications.
Authenticity and Integrity

Authenticity is a security feature that ensures that the entity on the other end of the VPN is known. Digital signatures can ensure that no one is impersonating an authorized user and can also detect data modification and ensure the integrity of the information.

Figure 82.

VPN Enabling Technologies

VPNs rely on the following enabling technologies:

- Encryption
- IPSec
- Tunneling protocols
- Authorization.

Encryption

Encryption provides privacy and security by scrambling and unscrambling data. Data security is implemented using various encryption and key network management schemes.
IPSec

IPSec was designed in part as a standards-based enabling technology for VPNs. In IPSec tunneling, encryption is applied to a protocol packet and the encrypted results are encapsulated into another IP packet. This encryption protects the identity or address of the source and destination, and permits the transport of packets across a non-IPSec-based network.

Tunneling Protocols

Tunneling is the process of encapsulating one type of packet in to another type of packet so that the data can be transported across paths that it could not have crossed otherwise. To create a tunnel, the source end encapsulates information in IP packets for transit across the Internet. Various tunneling protocols such as PPTP deal with establishing the end-to-end communication channels.

Authorization

While authentication refers to whether or not a user is permitted access, authorization refers to the characteristics and types of access that a user is permitted (e.g. what a user is authorized to do and when). Examples of authorization are:

- Which servers the user is allowed to access and when
- The method by which addresses are allocated
- Filters that apply to this group’s traffic.

Once the authentication and authorization steps are complete, the VPN tunnel is established and the user can send data across the VPN tunnel.
VPN Typical Applications

Small, medium-sized and large businesses use VPNs for typical applications. Each of these businesses, depending on its size and connection requirements, has a unique need.

Medium/Large Business

The typical medium-sized to large business needs to connect branch offices, remote users and headquarters.

The following example shows two of the three main VPN applications, remote access and site-to-site VPN, for a Fortune 1000 large customer. Using remote access with BCM at the branch offices as an extension from the headquarters location, the branch offices use VPN over the Internet to connect to headquarters resources. Using VPN as its network, this business gains cost savings because it does not have the expense and maintenance of a private network.

Figure 83.
Small- to Medium-Sized Business

The typical small- to medium-sized business needs an inexpensive way of connecting many branches.

Figure 84 shows a pure site-to-site VPN for a small- to medium-sized business. This VPN application could be used by a pizza store chain with hundreds of stores taking orders and delivering pizzas. The headquarters is staffed with a corporate call center that takes the calls for pizza orders and distributes them to the regional pizza store for that area over the Internet using VPN. Note that in this size of network, remote users would typically be supported directly from the headquarters Contivity server rather than using BCM’s client support capabilities.

**Figure 84.**

Extranets

Figure 85 shows a VPN used to support connections with an organization’s branch locations through the Internet. BCM utilizes PPTP to grant access privileges to its branches in order to increase the efficiency and information sharing of data among branches.
Nortel Complete VPN Solution

Figure 86 shows the entire range of possibilities with a complete VPN solution from Nortel. This VPN solution offers companies flexibility in their efforts to connect their offices, remote users and partners in a cost-effective way.

The key is that Nortel has a comprehensive range of VPN products that allows customers to select the appropriate product for each office or user depending on their respective size and needs.

The VPN solution leverages the Internet for flexibility and cost savings.
Figure 86.

BCM Virtual Private Networking (VPN) Support

IPSec

Internet Protocol Security (IPSec) is a developing standard for security at the network or packet-processing layer of network communication. IPSec is especially useful for businesses that want to implement a VPN. One of the main advantages of IPSec is that companies can handle security arrangements without requiring changes to individual user computers. BCM’s IPSec offering provides privacy, integrity and authenticity for networked commerce-crucial requirements for transmission of sensitive information over the Internet.

The level of encryption created is based on the choice of protocol, encryption method and the authentication method. The implementation of IPSec on the BCM supports the Encapsulating Security Payload (ESP) and Authentication Header (AH) protocols. ESP provides confidentiality for IP datagrams by encrypting the payload...
data to be protected. ESP uses the Data Encryption Standard (DES) algorithm. AH protocol provides data integrity and source authentication but does not encrypt data.

The encryption method on the BCM can be set for 128-bit Triple DES, 56-bit DES or 40-bit DES, with Triple DES being the strongest and 40-bit DES being the weakest level of encryption.

The authentication method can be either Secure Hash Algorithm (SHA1) or Message Digest 5 (MD5) Algorithm. SHA1 produces a 160-bit hash, but does not encrypt data. MD5 produces a 128-bit hash. It is used to confirm the authenticity of a packet but also does not encrypt data. MD5 also provides integrity that detects packet modifications. Both SHA1 and MD5 use Hashed Message Authentication Code (HMAC) to improve authentication. HMAC is a technique that uses a secret key and a message digest function to create a secret message authentication code. Cryptographers regard SHA1 as being more resistant to attacks than MD5.

Table 24 identifies the supported encryption and authentication levels.

Table 24

<table>
<thead>
<tr>
<th></th>
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</thead>
<tbody>
<tr>
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</tr>
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<td>Yes</td>
</tr>
<tr>
<td>AH HMAC MD5</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
</tr>
</tbody>
</table>

The BCM Network Address Translations (NATs), firewall and firewall filters will be supported in an IPSec environment. BCM only supports preshared shared keys.

The BCM IPSec capability is based on the Contivity client capabilities. IPSec on BCM200/400 allows up to 16 secure tunnels to be established between BCM and Contivity and/or BCM to BCM. BCM50 supports up to 5 Branch-to-Branch or Branch-to-Client tunnels.
IPSec Client Support

BCM200/400 supports IPSec Client Termination, in addition to branch office mode (server-to-server) connections supported in previous releases of BCM. IPSec Client Termination allows a Contivity Extranet Client to connect to the BCM from a remote PC, giving totally secure access to a private network from the remote PC. BCM50 does not support IP Sec Client terminations.

The Contivity Extranet features supported by this include:

- Idle timeout
- Perfect Forward Secrecy
- Rekey timeout
- Rekey Datacount
- Automatically-added Static Routes.

Other features provided in BCM IPSec Client Termination are:

- Split Tunneling
- DNS and WINS Server Fields
- Domain Name
- Static and Dynamic IP Address Assignment
- NAT Support
- IP Firewall Filters
- LZS Compression (optional).

16 client tunnels can be configured.

PPTP

Point-to-Point Tunneling Protocol (PPTP) is a proposed standard sponsored by Microsoft as an extension of the Internet’s Point-to-Point Protocol. Any user of a PC with PPP client support is able to use an Internet service provider to connect securely to a server elsewhere in the user’s company. The PPTP implementation on the BCM200/400 is designed for router-to-router configurations only: it does not support personal clients. In order for this facility to work, the username and password for each remote router must be setup on the BCM.

The BCM implementation of PPTP offers the following features:
- Support for multiple authentication schemes:
  - MS-CHAP
  - CHAP
  - PAP
- Support for IP address translation via encapsulation
- Support for RC4 encryption (either 56-bit or 128-bit, within the limits of U.S. export law)
- Support for compression of data packets.

A total of 10 PPTP tunnels can be configured on the BCM200/400.

Note: While BCM supports both IPSec and PPTP, they may not be used at the same time.

Secure Sockets Layer (SSL)

In addition to VPNs, BCM supports Secure Sockets Layer (SSL). Browser-based Secure Sockets Layer (SSL) requires little or no software on remote PCs, and in most cases any PC with a browser can be used to make a secure connection, by authenticating to the BCM. SSL firewall ports that the traffic uses are generally left open, so firewall reconfiguring is usually unnecessary. SSL’s simplicity translates into easy installation and long-term cost savings because of ease of ongoing support.

Secure Shell (SSH)

SSH is an industry standard (IETF) protocol that provides users with a secure means of logging into another computer over a network, allowing the user to move files and execute various commands on a remote computer. The SSH protocol is similar to the SSL protocol in that identities are authenticated and data communication is encrypted. SSH provides a secure alternative to the network Telnet interface as well as a SCP (Secure Copy) and SFTP (Secure FTP) capability. With BCM, the Secure Shell application provides a secure alternative to the network Telnet interface. This security results in confidentiality, on-the-wire tamper detection as well as protection against IP spoofing.

Access Control

This feature enhances password policies and password management. Password policy can be configured through UM for length, age, history and complexity. Upgrades from previous versions of BCM may require password changes to meet password policy. The BCM Monitor password is stored using stronger encryption techniques.
Access Control on BCM provides the following capabilities:

- **Elimination of Superfluous Accounts** – minimizes exposure to any compromised interfaces, provides single default admin account (“ee_admin”), deletes other default accounts (Supervisor, NortelTS, ModemBackup) and disables and renames NT Guest account.

- **Account Lockout Policy** – means that the Default Account is locked out for 30 minutes after five invalid login attempts. The user will be locked out for 30 minutes, or can be unlocked earlier by an administrator.

- **Remote Access Control** – provides Modem Dial-Back capability (2-Factor Security). The modem is disabled by default and can be enabled through Unified Manager. This capability also provides the ability to control Dial-In Access privileges using Unified Manager through manipulation of Dial up User Group.

- **FTP Server Improvements** – The FTP server remains disabled by default and anonymous FTP is not allowed.

- **Enhanced Authentication Policy** – allows administrator accounts to login through local interactive login interface. Network Telnet/SSH or serial console Telnet access is restricted to users that are members of the Administrative group. Apache authentication uses NT password DB (eliminates secondary internal password DB).

- **Shared Drives Disabled** – Shared drives on the network are disabled by default.

### Denial of Service (DoS) prevention

BCM helps prevent Denial of Service (DoS) due to malformed packet and stress testing of data network, thereby reducing system overload.

### Limiting BCM information revealed publicly

With BCM, the Apache revealed information in the server response header is limited. In addition, BCM does not announce itself to the Network Browse Master and is not included in Network Neighborhood list. Its Default Manager List setting (10.10.10.1) ensures no public access for SNMP operations, unless it has been enabled by a change, and should still change public and private interface (by re-naming) by the customer.
The BCM management framework comprises two fundamental components: element management with Unified Manager for BCM200/400 systems and Element Manager for BCM50, and multielement management using Network Configuration Manager.

For the BCM200/400 systems, the primary user interface for the management of the BCM as a network element is a Web browser-based set of tools collected under the umbrella of the BCM Unified Manager. Using Unified Manager, any BCM in the network can be configured and managed from any PC on the network.

With the introduction of the BCM50, a new management application Element Manager is delivered which provides a configuration and administration environment that offers improved speed and ease of use compared with the Unified Manager.

The Network Configuration Manager (NCM) is an optional multisite management tool providing a centralized inventory database and configuration capabilities for networks comprising many BCMs. Figure 87 depicts a BCM network together with the Unified Manager and NCM.
In addition, third-party management tools capable of monitoring SNMP traps can be used to provide notification of faults and critical events on BCMs via an IP connection. This enables personnel at a Network Operations Center to respond rapidly to issues that may impact service to system users, maximizing system availability.

Web-based management interfaces, such as the BCM Unified Manager, are common today as they provide platform independence on the client side. An additional benefit of Web-based management is that no software is loaded on the client computer. This means that with the appropriate identity and passwords to log into a system, a business can use any PC on the network to administer the system, or multiple systems, from a single location.

While Element Manager for BCM50 requires a client to be installed on the administrator’s PC, the benefit is the increased performance of the application in performing common BCM50 administrative tasks. The Element Manager application can be downloaded directly from the BCM50 web page for ease of installation and launch from any personal computer.

System administration of the BCM using a digital telephone set is not supported for BCM200/400 systems, however has been introduced with BCM50. This interface supports telephony and applications programming.
consistent with the Norstar Telset Admin interface, and additionally supports system IP settings and the entry of licensing information.

**BCM200/400: Unified Manager**

The Unified Manager, shown in Figure 88, is a browser-based management tool. It is the single point for managing all programming for individual BCM systems. Access to the Unified Manager is password protected, making it secure for both enterprise customers and small- to medium-sized businesses.

*Figure 88.*

Technicians can use Unified Manager and associated wizards to quickly set up users, mailboxes and directory numbers (DNs). With such speed and efficiency, a business can be up and running very quickly. Changes can also be handled remotely by logging into the element over the LAN or WAN and running Unified Manager to make configuration changes.

**Unified Manager Capabilities**

BCM Unified Manager provides the following features:
- The ability to perform BCM remote administration and configuration via the IP network
- The ability to view and change configuration settings for all applications on a BCM
- The ability to monitor events and alarms for troubleshooting purposes
- The automation of system backup and restore routines through a backup and restore scheduling capability
- Enhanced user interface to provide a greater level of intuitive navigation. Each functional area of the Unified Manager contains both data and voice administration for its respective area
- Keycode management and administration, allowing the administrator to enable a single application with a single Keycode or to enable multiple applications with the provision of a Keycode file
- Windows NT authentication login to support the use of the administrators’ existing NT logins, which means they are not required to maintain a separate ID just for system administration
- Controlled, multilevel access to users, enabling system administrators to provide user-customized access and control to Unified Manager
- A pull-down menu option to enable originating caller OLI or Last Forwarded Number OLI.
- The ability to program a Call Forward All Calls setting on per-telephone basis.
- The ability to program a Selective Line Redirect setting on a per-line basis.
- The ability to set up a static DLCI-to-IP mapping for Frame Relay. This would be done rather than using InvARP, since some Frame Relay service providers do not provide support for this protocol.
- The ability to display what Microsoft hot fix security patches have been applied on the BCM, allowing administrators to determine if a given vulnerability is applicable. This is available on the Inventory page.
- The ability to configure parameters for NetIQ’s AppManager application, which monitors system health and Quality of Service (QoS) network performance. For this functionality, the NetIQ agent must be installed on devices on the network. Settings available include Enable/Disable Agent, Set Vivinet Manager IP Address, and Specify Port.

Unified Manager can be launched from the multi-BCM Network Configuration Manager, enabling NCM to act as a single point of management interaction with a network of BCM systems. In addition, BCM can appear as an element in a network discovery diagram of the Optivity Network Management System (ONMS). The Optivity NMS Discovery and Launch enables visibility of BCM elements and supports the launch of the BCM Unified Manager from within ONMS.
BCM Management Access Options for BCM200/400

You can connect to Unified Manager via IP, either over a BCM WAN interface, the BCM ethernet interface or over a dialup connection using the internal V.90 modem and dedicated analog line to the BCM system. The ethernet interface can be local to the site through either an existing LAN connection or a direct connection from a PC to the system using a crossover ethernet cable. Businesses can also access the Unified Manager by a browser across a WAN interface or Internet connection from a PC located anywhere on their network. This capability can minimize on-site visits and simplify support.

If a secure connection is required, the BCM IPSec capability can be used to enable a secure management VPN connection over the Internet or WAN.

When an on-site technician installs the BCM system, the IP address for the system can be programmed by connecting a laptop to the serial port on site or can be set by running the BCM QuickStart wizard. This makes the BCM visible on the customer’s network and allows a remotely-located skilled administrator to continue with the configuration of the system. This flexibility means that BCM easily fits into the network of a large enterprise customer and can also easily be managed by a smaller customer with one or more sites.

*Figure 89.*
BCM Unified Manager Interface for the BCM200/400

The Unified Manager provides an intuitive interface for users to access all programmable parameters, both to install the system initially and to manage the system on an ongoing basis.

The Unified Manager has been designed to make installation and administration simple. Administrators and technicians can quickly configure data and voice services on the BCM by using the Unified Managers tabs, buttons and right-click mouse functions. Drop-down menus provide access to dialog boxes in which users can enter, modify and delete data as well as access performance-tracking charts and tables. Users can also access alarms and events and perform diagnostics through the interface.

Using a wizard, the administrator can also develop templates for use on multiple system configurations. This means the multisite administrator can create one configuration for multiple branches and simply download the configuration at other sites.

Java is used at both the Client PC and the BCM server for creating the user interface for managing the different elements of the BCM system. The use of a Web-based management system provides universal access at minimum cost to both the end user and the reseller.

The Unified Manager main page is shown in Figure 90. In addition to providing access to all programming functions via the Configure link, the main page provides access to utilities such as the Setup and Management wizards, Backup and Restore Utility and CallPilot Manager for managing the system Messaging and call center services. Documentation is also provided, as is access to product maintenance and support.
As shown in Figure 91, the product documentation suite and a product overview are provided directly on the hard disk of each BCM system.
The **Configure** link on the Unified Manager main page (Figure 92) provides access to the BCM programming interface. This is organized into five groupings entitled System, Resources, Services, Management and Diagnostics. Figure 92 also shows examples of screen shots from the Configure portion of the Unified Manager.
Figure 92.

The Services and Resources selections enable the configuration of telephony and data services and resources, as described in the following sections.
Resources Menu

Figure 93 shows the type of information displayed when a user selects the Resources command button. Users can view all the resources (such as Media Bay Modules, Media Services Cards and data networking resources) on the BCM here. These resources can then be configured for services within the Services section of the Unified Manager.

![Figure 93](image)

When the Resources command button is expanded, users can access all of the Configuration menu options to enable and configure BCM resources.

Figure 94 shows the type of information displayed when a user selects one of the buses on a pluggable Media Bay Module. When a user selects a bus, he or she can enable or disable it or view its status.
Figure 94.
Services Menu

Figure 95 shows the type of information displayed when a user selects the Services command button. Here, users can view all of the services on the BCM. When the Services command button is expanded, access is provided to all of the Configuration menu options for the configuration of the BCM services enabled on the system.

Figure 95.

Setup and Management Wizards

Also accessible from the Unified Manager main page is a collection of Setup and Management wizards, which enable installers to program the BCM quickly and simply. By applying a few required settings, a Setup and Management wizard can execute a large number of functions by making common assumptions.

Figure 96 shows the Setup and Management Wizards page with five wizards to choose from:
- Quick Start – Configure an un-initialized system
- Add User(s) – Add new users
- Edit Template – Edit a telephony template
- DN Renumber – Renumber DNs
- Network Update – Update data network settings once the Quick Start Wizard (which does the initial network setup) has been run.

If the system is equipped with DECT equipment, then two additional menu choices will appear:

- DECT – DECT Configuration (Not shown in illustration. Wireless/Cordless stand available in Europe)
- DECT – DECT Mobile Recording not shown in illustration. (Handset subscription available in Europe and selected Latin American countries only).

**Figure 96.**

### Setup and Management Wizards

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Quick Start</td>
<td>Configure an un-initialized system</td>
</tr>
<tr>
<td>Add Users</td>
<td>Add new users</td>
</tr>
<tr>
<td>Edit DN Record Template</td>
<td>Edit a DN record template</td>
</tr>
<tr>
<td>DN Renumber</td>
<td>Renumber DNs</td>
</tr>
<tr>
<td>Network Update</td>
<td>Update the network settings</td>
</tr>
</tbody>
</table>

BCM200/400 Backup and Restore

Unified Manager’s Backup and Restore Utility (BRU) facilitates the backup or retrieval of a complete copy of all or selected system programming information. Users can access this tool through the Backup and Restore Utility (BRU) link on the Unified Manager main page.
BRU provides the ability to back up programming information from the BCM hard drive to another computer on the network. In the event of a hard drive failure, the programming information can be restored from the archived copy.

The BRU tool allows the administrator to select which BCM components are to be backed up. For example, the administrator may choose to back up only telephony information and voice applications, or IVR settings. The administrator is also able to schedule how frequently the backup should be run as well as specify the time at which the backup should occur.

The voice application backup section includes all the messages that may be resident in the system at the time the backup is performed. The Restore option of the BRU tool allows the retrieval and application of a complete set of previously backed up programming information to a BCM system. The Restore feature allows the overwriting of existing programming.

When Restore is run, the system puts all of the sets in maintenance mode. Previously backed up programming data is then loaded into the system’s memory from the server’s hard disk or other data storage medium. After the data is loaded, the Backup and Restore tool takes the sets out of maintenance mode by initiating a warm start. A warm start restarts the BCM system with all of its programming intact, including any changes.

In multi-BCM networks, the BCM Network Configuration Manager tool also supports the backup and restore process. NCM enables administrators to efficiently schedule and monitor status of backup/restore processes across a network of BCM systems.

**BCM200/400 Fault Management**

Alarm and fault conditions on the BCM are available for viewing in the BCM Unified Manager Alarm Manager window. SNMP v1 traps for these events can be generated by the BCM and output over an always-on IP connection (via WAN or LAN interface on the BCM) and directed to a central monitoring facility. SNMP trap monitors or network management frameworks can be used to collect these traps for reporting or the initiating of BCM specific troubleshooting activity. The configuration of the SNMP system (basic filtering of traps by source or severity, IP address of central monitoring facility to which traps are sent, etc.) in each BCM is programmable via either Unified Manager, for individual BCMs, or via NCM, if changes are to be made across a large number of BCM systems simultaneously.

In terms of MIB support, BCM supports MIB II. This SNMP MIB enables SNMP capable management systems to poll the BCM for status and performance information in the areas of WAN and LAN interfaces. Additionally, BCM supports the Microsoft Performance MIB which provides access to information about memory, processor, network interfaces, physical disk, logical Disk, processes and other system information.
BCM50 Management Environment

The primary management application for configuring and administering the BCM50 is the BCM50 Element Manager, which is a client-based management application that runs on a Windows computer. The Element Manager allows for connection to BCM50 devices over an IP network. It is used to configure, administer, and monitor BCM50 devices. The Element Manager application can be downloaded from the BCM50 web page or from the Nortel Networks support web site.

While Element Manager is the primary management application, BCM50 also supports the programming of telephony and applications areas of BCM50 through set-based administration. This allows installers already familiar with this interface to perform programming from the keypad of any telephone connected to the BCM50 device. This alleviates the need for access to a computer at the customer site.

The Router WebGUI is used to manage the router integrated in the BCM50a and BCM50e systems. In addition to configuring the router, the WebGUI supports tasks such as backing up router software, and performing alarm and performance monitoring for the router.

BCM Management Access Options for BCM50

A 10/100 Ethernet port is reserved on the BCM50 front panel for direct management access to the BCM50 system.

For remote management access, BCM50 has an integrated analog modem that can accept an incoming modem call on any BCM50 system line. You can configure the BCM50 system to let the modem auto-answer a specific line with configuration options. Remote users can also first initiate a voice call to a person or an auto-attendant, who transfers the call to the modem. The analog modem also supports callback for management access to the BCM50, which can be used to support auto-dialout on SNMP traps, and automated sending of Call Detail Records (CDR) to a remote CDR collection point and other management tasks.

Although we do not recommend that you use the analog modem for transferring large files, the modem gives you a flexible method of remote access to perform all programming tasks remotely.

The BCM50e and BCM50a give you remote management capability with a high-speed connection. Tasks such as transferring backup files to a remote destination, and transferring software update files, CDR records and log files, can be more efficiently supported than over the analog modem.
BCM50 Element Manager

The BCM50 Element Manager is an application that leverages the standards-based access to BCM50 information with a new off-box management architecture. This client-based architecture with task-oriented information presentation delivers improved responsiveness, performance, and application functionality. It gives a simpler, faster, more information-rich and compact user interface.

Programming data is organized in the Element Manager to quickly show all of the associations between data in the form of “list browser” tables. This eliminates having to look in multiple places to construct a system view of the programming.

Figure 97.

Because the Element Manager is an installed client application on a PC, all of the information required to render the user interface is contained locally, and the only data transferred between the Element Manager and the BCM50 are queries and data in accordance with the data model. The Element Manager also offers features familiar from other common applications, such as the ability to sort information in tables.

Most of the traditional BCM administration tools have been integrated into the new Element Manager management environment, including management of backups and software updates and upgrades.

The Element Manager provides an element navigation panel for organizing a network of devices. Management sessions can be open to multiple devices at a time. Once connected to an element, a second panel provides access to both configuration and administration tasks. Within either selection, you can easily navigate through clearly-
named nodes which support the various tasks. No more than 3 keyclicks are required to reach programming screens, many of which use tabs and detailed screens to present a large amount of information.

The Configuration management environment encompasses all BCM50 configuration programming including pass through launch to the CallPilot Manager application and the router WebGUI management application for the BCM50a and BCM50e integrated router.

The method of programming BCM50 core telephony will be very familiar to BCM-knowledgable partners although the interface has been substantially improved.

A significant change has been introduced in software feature management. The BCM keycode structure has been redesigned to deliver stronger security with an SHA1 algorithm, along with the following capabilities:

- One Keycode validates ALL feature entitlements, simplifying installation for high partner value
- Flexible Keycode application:
  - Element Manager option either of GUI-based entitlement selection and license entry or importing a feature licensing file,
  - Set based interface for entering licensing information
  - Network Configuration Manager (NCM) bulk keycode distribution & entry
  - USB memory stick based keycode file entry together with Startup Profile for quick installation

The Element Manager Administration interface

Element Manager’s Administration management environment encompasses the following tasks:

- BCM50 diagnostic and maintenance tools, including launch of BCM Monitor.
- Fault management for viewing BCM50 alarms. You can set which alarms are displayed in the
- Element Manager alarm browser and which alarms trigger an SNMP trap on a per-alarm basis.
- Log management for off-box transfer of logs, including component logs and administrator logs such as alarm log, security log, configuration change log, and system log.
- Software management tools for software updates (for example, for corrective software) and software upgrades. A software update history is maintained on the element. New update software to be applied is checked against the software history and validated before it is transferred to the BCM50 for application. Software updates can be downloaded to the BCM50 either on-demand or according to a schedule. The application of the software update can also be scheduled. For example, you can schedule a download for
Tuesday night but the application can be scheduled for Friday at 2 a.m. A reboot will be invoked only if a software component being updated requires a reboot to activate the new code.

- Backup and restore. You can schedule configuration backups or application backups. An application backup includes data generated through the day-to-day use of the on-box applications as well as the configuration data. A scheduled backup provides the ability to routinely perform a backup, which can be kept on the BCM50 hard drive or transferred to an off-box destination such as network folder, FTP server or locally attached USB storage device.

For backup and restore, software management, and log management, a flexible set of source and destinations is supported along with protocols to access them. For example, you can save backups from the BCM50 to either the USB port, the Element Manager client PC (on-demand only), a shared drive available on the network, or a remote FTP or SFTP server. Likewise, software updates can be retrieved by BCM50 from any of these sources.

**CallPilot Manager OA&M Interface**

CallPilot Manager is a Web-based management tool that can be accessed from the Unified Manager or Element Manager main page for support of either BCM200/400 BCM3.x or BCM50 Release 1 systems. It is used to administer the Messaging and Call Center applications in BCM systems.

A business can use any PC that has access to the network for Messaging and Call Center administration.

Call Center supports user-programmed greetings or announcements. CallPilot Manager allows administrators to specify a text description of the greeting or announcement, making it easier to manage or change greetings or announcements. Basic Call Center supports up to 30 greetings or announcements, Professional Call Center supports up to 150.

CallPilot Manager allows Time of Day Auto Attendant Blocking, which is the ability to prevent dialing of designated extensions during specific hours of the day. When AA is blocked, calls can be routed directly and immediately to associated Messaging mailboxes with other valid CCR tree menu selections unaffected.

Management of Voice Messaging greetings, CCR trees and business hours is also provided in the BCM Network Configuration Manager tools for users who are managing large networks of BCM systems.

Note that users of the Unified Messaging application will use the Mailbox Manager tool, which allows subscribers to maintain mailbox greetings and other settings, or the Operator Manager tool, which system attendants use to control business settings. These are subscriber tools that are unrelated to the CallPilot Manager application for administration of the BCM messaging and call center applications.
System Monitoring with BCM Monitor

A valuable application for performance monitoring is the BCM Monitor. It allows the BCM administrator to see the current status of various parts of the BCM system. Statistical information is provided on system throughput and other performance-related information, including system CPU usage (graph or table format) and memory usage (graph or table format).

If a performance display is active, it is automatically updated with real-time performance information in user-selectable time increments.

The focus of the real-time monitoring capabilities is:

- Overall system status
- Utilization of resources on the Media Services Card (e.g. signaling channel usage)
- Operation of telephony applications (e.g., Messaing, Call Center, etc.).
- IP telephony activity
- D-channel monitoring for PRI, BRI and VoIP trunks

Figure 98 shows an example of the System Monitoring information that is available.

*Figure 98.*
The BCM Monitor application can be downloaded to an administrator’s PC from the BCM and pointed at a specific BCM’s IP address for monitoring. Multiple instances of the BCM Monitor application can be used on a single PC to monitor several remote BCM systems at the same time.

Backward version compatibility is supported.

**Network Configuration Manager (NCM)**

Network Configuration Manager (NCM) is an optional graphical client-server software application that enables network managers to rapidly configure and manage a network of BCM devices. NCM allows network administrators to efficiently maintain and manage a large number of BCMs on the network, sorted into logical hierarchical folders, with current configurations stored in a single database. As shown in Figure 99, NCM complements Unified Manager by providing an efficient means of effecting specific changes or collecting information from a large number of BCM systems, all via a single user interface. NCM can act as the first point of access for managing a network of BCM systems. Its database provides IP address and connect information for each BCM being managed, keeps an inventory of all applications/hardware equipped at each BCM site and provides a launch point for the BCM Unified Manager.

BCM Network Configuration Manager enables the BCM to deliver the lowest cost of ownership and deliver new customer service and revenue generation values via its ability to dynamically change configuration settings in multi-BCM networks.

NCM supports all of the BCM3.x release levels and the same NCM server will also manage BCM50 allowing for a network comprising a mix of BCM devices at different release levels.
NCM Architecture

Network Configuration Manager consists of two components:

- The Network Configuration Manager server software, incorporating a centralized database (available for Microsoft Windows and Solaris platforms)
- The NCM client software, used by the network administrators.

Additionally, a Network Configuration Manager software authorization code is required on each BCM that is to be managed.

The server software is available in three different packages. The first package supports up to 250 BCMs and allows for up to five simultaneous client administrative sessions. Packages two and three support up to 1000 and 2000 BCMs respectively, both allowing for up to twenty simultaneous client sessions. For networks over with 2000 BCMs, a custom package can be created.

The centralized database of the NCM provides:
• A storage area for all system information, including inventory, configuration file and selected configuration information
• Rapid access to up-to-date information on any system on the network
• Ability to generate audit reports about configuration activity.

NCM is available with an embedded Cloudscape database. Alternatively, the option of using it with a customersupplied Oracle database is available.

NCM Tasks

Network Configuration Manager allows the network administrator to perform the following tasks across some or all of the BCMs within a network:

• Apply the NCM Keycode to a BCM on the network that does not have the Keycode installed
• Connect to a BCM on the network and import its inventory and configuration data
• View the inventory, configuration data and all applied Keycodes
• Connect to a BCM on the network and perform a direct comparison between an archived configuration and the current configuration active on the device
• Connect to a BCM on the network, perform a backup (BRU) of its configuration and archive the backup files in the NCM database, with the option of using personal messages in the backup
• Connect to a BCM on the network and perform a restore (BRU) of its configuration from a previously archived configuration
• Export an archived configuration (backup files) to a TFTP server
• Import and archive backup files from a TFTP server
• Generate/view/print inventory and configuration reports as follows:
  • System inventory report
  • Quickstart information report
  • System identification only report
  • Telephone directory report
  • Telephone configuration report
  • Target line report
  • LAN/WAN interfaces report
  • Full configuration report
Partial configuration report allowing the user to select which fields from the NCM database they wish to have included in the report.

Port inventory report

Launch Unified Manager and Desktop Assistant Admin ProAE.

BCM Network Management Configuration Manager can produce a hardware inventory and print out of the configuration for each or all BCMs in the customer’s network, including active sets and set types and Media Bay Modules, all on one report that can be produced on demand or on regularly scheduled basis.

Programming changes can be applied to all, or a subset of, BCM systems within a network from a centralized location. The changes that can be applied include:

- CallPilot Auto-attendant greetings and hours of operation
- Holiday greetings scheduling
- Call Center greetings and prompts
- Custom Call Routing trees
- IP Music playlist and music files
- System speed dial
- Call routes and destination codes
- Call restriction filters
- Router configuration
- Routing tables
- VoIP gateway tables
- System SNMP trap/alarm settings including secure modem dialback settings
- Software Keycode distribution
- User access control management including password change
- UPS configuration settings

NCM provides the ability to schedule the listed changes and to define groups of BCM systems to which the changes may be applied.

Additionally, NCM provides the following support for efficient use of the BCM IVR application across a network of BCMS:
• The ability to deploy, upload, remove and start/stop IVR scripts from BCM(s) on a real-time or scheduled basis

• The ability to deploy audio prompt files, replace existing audio files with new files, remove audio files, upload audio files and restart the application with new audio files.

**BCM Patch Distribution With NCM**

NCM has the capability to distribute and apply patches to multiple BCM. This includes the ability to schedule deployment of patches to BCM systems, initiate the BCM patch process on designated BCMs, and confirm within the NCM audit log the successful completion of the patch process on specific BCMs. The benefits of this capability are reduced operational cost savings and improved network performance.

**NCM Voice Messaging Backup**

Voice Messaging is included in the NCM backup and restore process, so Messaging boxes can be archived off the BCM for future access. This feature benefits such organizations and environments as law firms and network operation centers.

**NCM Try-and-Buy Keycode**

A Keycode is available that will allow a 60-day trial of NCM software, enabling customers to assess the NCM capabilities to ensure it is right for their business before committing to buying it.

**BCM User Managed Functions**

In addition to the administrator management capabilities discussed above, the BCM provides a number of tools that can be used to manage the individual environments of BCM users.

The BCM allows users to program the features and autodial keys on their telephone to best perform their job. This programming ability lets users modify a set to meet the changing needs of their job without involving a system administrator.

The Desktop Assistant PRO tool allows for users to program telset buttons and to print labels for any set on the system. This application uses LAN CTE as means to connect to the BCM. Both Desktop Assistant PRO and LAN CTE Client must be installed on the Client PC in order to utilize this application. Any digital set on the system, including IP sets, can have their buttons programmed and labels printed.
Desktop Assistant Pro – Administrator Edition is similar to Desktop Assistant PRO in all aspects except that it has the ability to program any digital or IP set on any BCM in a network. This application is intended to provide an easy-to-use GUI for administrators to program individual telsets remotely. This application also uses LAN CTE as means to connect to any desired BCM. The Desktop Assistant PRO AE and LAN CTE Client must be installed on the Client PC in order to utilize this application, although the LAN CTE client license on each BCM being managed need not be dedicated to a specific management workstation.

BCM also supports the ability to “lock-in” the programming of the phones, if required by the business, in order to prevent users from changing programmed settings.

Personal Call Manager is a TAPI application that provides an easy yet powerful PC interface for complementing Business Communications Manager telephones. With Personal Call Manager, users can manage all calls from a PC. Some of the functions that Personal Call Manager can perform are:

- Make calls
- Answer calls
- Screen calls (if Calling ID facilities are available)
- Redial telephone numbers
- Conference calls
- Transfer calls
- Hold/unhold calls
- Forward and cancel forward your phone
- Set your phone to Do Not Disturb
- Manage your own personal contact list.

Software Keycodes

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 Keycodes (a password number provided to the installer).

The following tables describe the software applications that are standard and the software Keycode bundles and options for BCM.
Table 25.

<table>
<thead>
<tr>
<th>Standard Applications</th>
<th></th>
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<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>

BCM200/400 Keycodes

Table 26.

<table>
<thead>
<tr>
<th>Bundles</th>
<th>Capability</th>
</tr>
</thead>
</table>
| Bundle #1: Silver – 16 | Nortel Call Center, Basic  
VPIM & AMIS Message Networking  
BCM LAN CTE, 16 seats  
BCM Attendant Console, 1 seat  
Unified Messaging, 16 seats  
FAX Messaging |
| Bundle #2: Silver – 32 | Nortel Call Center, Basic  
VPIM & AMIS Message Networking  
BCM LAN CTE, 32 seats  
Attendant Console, 1 seat  
Unified Messaging, 32 seats  
FAX Messaging |
| Bundle #3: Silver – 64 | Nortel Call Center, Basic  
VPIM & AMIS Message Networking  
BCM LAN CTE, 64 seats  
BCM Attendant Console, 1 seat  
Unified Messaging, 64 seats  
FAX Messaging |
### Bundles

<table>
<thead>
<tr>
<th>Bundle #4: Gold</th>
<th>Capability</th>
</tr>
</thead>
</table>
| (Note that customer must have purchased a Silver bundle before upgrading to the Gold bundle.) | BCM MCDN & Q.SIG Voice Networking  
Nortel Call Center to Professional Call Center Upgrade  
Nortel Reporting for Call Center  
FAX Overflow  
FAX On Demand  
BCM IP Sec |

**Note:** Voice Messaging seats are not included in Silver or Gold bundles.

---

### Options

<table>
<thead>
<tr>
<th>Options</th>
<th>Capability</th>
</tr>
</thead>
</table>
| Try-and-Buy Options  
(Options available for 60 days from activation or until replacement.) | BCM VPIM & AMIS Message Networking  
BCM MCDN & Q.SIG Voice Networking  
BCM DPNSS Voice Networking  
BCM LAN CTE – Maximum seats  
BCM Attendant Console – one seat  
BCM VoIP Gateway – two trunks  
BCM Unified Messaging – Maximum seats  
BCM FAX Suite  
BCM IP Telephony Client – eight seats  
BCM Voice Messaging – Maximum seats |
<p>| BCM CallPilot Message Networking | Enables VPIM &amp; AMIS Message Networking |
| BCM Call Center | Enables Nortel Call Center |
| BCM Reporting for Call Center | Enables Nortel Reporting for Call Center |
| BCM Professional Call Center | Enables Nortel Professional Call Center includes Reporting for Call Center |
| BCM Call Center to Professional Call Center upgrade | Upgrades from Call Center to Professional Call Center options available with or without Reporting for Call Center |
| BCM Call Center agents | Adds 1, 4, 8, 16, 32, 80 Maximum Agent Seat License(s) (only available for Professional Call Center) |
| BCM Multimedia Call Center | Enables Nortel Multimedia Call Center |
| BCM Interactive Voice Response (IVR) Run-Time Engine and IVR Channels | Enables BCM IVR runtime engine and configures channels. Available in 2, 4, 8, 16 and 24 channels (maximum of 24 IVR channels per system). |</p>
<table>
<thead>
<tr>
<th>Options</th>
<th>Capability</th>
</tr>
</thead>
<tbody>
<tr>
<td>BCM IVR Host Communications Emulation</td>
<td>Activates host communications emulation on BCM to support access to PeriView</td>
</tr>
<tr>
<td>BCM TSP Seats 1.x and 2.x only</td>
<td>Adds 1, 4, 8, 16, 32 Maximum TSP Seat License(s)</td>
</tr>
<tr>
<td>BCM LAN CTE 2.5 only</td>
<td>Adds 1, 4, 8, 16, 32 Maximum Seat License(s)</td>
</tr>
<tr>
<td>BCM Attendant Console</td>
<td>Adds one Attendant Console License (max. five)</td>
</tr>
<tr>
<td>BCM VoIP Gateway</td>
<td>Adds 2, 4, 8, 16, 32 VoIP Gateway trunks (60 max). May require additional PEC III's</td>
</tr>
<tr>
<td>BCM Unified Messaging</td>
<td>Adds 1, 4, 8, 16, 32 Maximum Seat License(s)</td>
</tr>
<tr>
<td>BCM Voice Messaging Mailboxes</td>
<td>Adds 1, 4, 8, 16, 32 Maximum Mailboxes</td>
</tr>
<tr>
<td>BCM FAX Messaging</td>
<td>Enables FAX Messaging</td>
</tr>
<tr>
<td>BCM FAX Overflow</td>
<td>Enables FAX Overflow</td>
</tr>
<tr>
<td>BCM FAX On Demand</td>
<td>Enables FAX On Demand</td>
</tr>
<tr>
<td>BCM FAX Suite</td>
<td>Enables all FAX features</td>
</tr>
<tr>
<td>BCM Digital Mobility Portables</td>
<td>Enables 1, 4, 8 BCM Digital Mobility handsets</td>
</tr>
<tr>
<td>BCM Digital Mobility Base Station (two radios) (USA only)</td>
<td>Enables BCM Digital Mobility Base Station</td>
</tr>
<tr>
<td>BCM IP Sec</td>
<td>Enables BCM IP Sec VPN capability</td>
</tr>
<tr>
<td>BCM IP Telephony Client(s)</td>
<td>Adds 1, 4, 8, 16, 32 IP Clients</td>
</tr>
<tr>
<td>BCM Point-to-Point over Ethernet (PPPoE)</td>
<td>Enables PPPoE on a per-site basis</td>
</tr>
<tr>
<td>BCM 3.7 Software Upgrade Authorization</td>
<td>Allows installation of BCM 3.7 software upgrade on previous releases.</td>
</tr>
<tr>
<td>BCM Network Configuration Manager</td>
<td>Network Configuration Manager Software Authorization Code, enables multisite</td>
</tr>
<tr>
<td>SoftPhone Client CD</td>
<td>IP Softphone 2050 Client, enabling the IP Softphone 2050 on a per-client basis (required if your total clients exceed 16 per site)</td>
</tr>
<tr>
<td>NetIQ Keycode</td>
<td>Keycode enabling remote performance monitoring by the NetIQ AppManager application</td>
</tr>
</tbody>
</table>

### BCM50 Keycodes

<table>
<thead>
<tr>
<th>Call Center Products</th>
<th>VOICE MESSAGING</th>
</tr>
</thead>
<tbody>
<tr>
<td>BCM50 Basic Call Center SW Auth Code</td>
<td>BCM50 Basic Call Center SW Authorization Code</td>
</tr>
<tr>
<td>----------------------------------------</td>
<td>------------------------------------------------</td>
</tr>
</tbody>
</table>

**Unified Messaging - Desktop Messaging Client**

|------------------------------------------|------------------------------------------------|

**FAX Messaging**

<table>
<thead>
<tr>
<th>BCM50 Fax Messaging Auth Code</th>
<th>BCM50 Fax Messaging Authorization Code</th>
</tr>
</thead>
<tbody>
<tr>
<td>BCM50 Fax Overflow Auth Code</td>
<td>BCM50 Fax Overflow Authorization Code</td>
</tr>
<tr>
<td>BCM50 Fax on Demand Auth Code</td>
<td>BCM50 Fax on Demand Authorization Code</td>
</tr>
<tr>
<td>BCM50 Fax Suite Auth Code</td>
<td>BCM50 Fax Suite Authorization Code</td>
</tr>
</tbody>
</table>

**Voice Networking**

|-----------------------------------------|------------------------------------------------|

**LAN CTE**

<table>
<thead>
<tr>
<th>BCM50 LAN CTE - 1 Seat Auth Code</th>
<th>BCM50 LAN CTE - 1 Seat Authorization Code</th>
</tr>
</thead>
<tbody>
<tr>
<td>BCM50 LAN CTE - 4 Seat Auth Code</td>
<td>BCM50 LAN CTE - 4 Seat Authorization Code</td>
</tr>
<tr>
<td>BCM50 LAN CTE - 8 Seat Auth Code</td>
<td>BCM50 LAN CTE - 8 Seat Authorization Code</td>
</tr>
<tr>
<td>BCM50 LAN CTE - 16 Seat Auth Code</td>
<td>BCM50 LAN CTE - 16 Seat Authorization Code</td>
</tr>
</tbody>
</table>

**VOIP Gateway**

<table>
<thead>
<tr>
<th>BCM50 VoIP Gateway - 1 Trunk Auth Code</th>
<th>BCM50 VoIP Gateway - 1 Trunk Authorization Code</th>
</tr>
</thead>
<tbody>
<tr>
<td>BCM50 VoIP Gateway - 2 Trunks Auth Code</td>
<td>BCM50 VoIP Gateway - 2 Trunks Authorization Code</td>
</tr>
<tr>
<td>BCM50 VoIP Gateway - 4 Trunks Auth Code</td>
<td>BCM50 VoIP Gateway - 4 Trunks Authorization Code</td>
</tr>
<tr>
<td>BCM50 VoIP Gateway - 8 Trunks Auth Code</td>
<td>BCM50 VoIP Gateway - 8 Trunks Authorization Code</td>
</tr>
</tbody>
</table>

**IP Clients (IP sets and soft clients)**

286
<table>
<thead>
<tr>
<th>Feature</th>
<th>Code Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>BCM50 IP Telephony Client</td>
<td></td>
</tr>
<tr>
<td>- 1 seat Auth Code</td>
<td>BCM50 IP Telephony Client - 1 seat Authorization Code</td>
</tr>
<tr>
<td>- 4 seat Auth Code</td>
<td>BCM50 IP Telephony Client - 4 seat Authorization Code</td>
</tr>
<tr>
<td>- 8 seat Auth Code</td>
<td>BCM50 IP Telephony Client - 8 seat Authorization Code</td>
</tr>
<tr>
<td>- 16 seat Auth Code</td>
<td>BCM50 IP Telephony Client - 16 seat Authorization Code</td>
</tr>
<tr>
<td>Network Configuration Manager</td>
<td></td>
</tr>
<tr>
<td>- BCM50 w/o Router NCM Auth Code</td>
<td>BCM50 without Router Network Configuration Manager (NCM) Authorization Code</td>
</tr>
<tr>
<td>Analog Trunks</td>
<td></td>
</tr>
<tr>
<td>- BCM50 ANALOG TRUNKS - 4 PORT AUTH CODE</td>
<td>BCM50 Analog Trunks - 4 Port Authorization Code: Enables analog trunk ports of the BCM50 main chassis.</td>
</tr>
<tr>
<td>Analog Station Ports</td>
<td></td>
</tr>
<tr>
<td>- BCM50 ANALOG TRUNKS - 2 PORT AUTH CODE</td>
<td>BCM50 Analog Station - 2 Port Authorization Code: Enables analog station ports of the BCM50 main chassis.</td>
</tr>
<tr>
<td>- BCM50 ANALOG TRUNKS - 4 PORT AUTH CODE</td>
<td>BCM50 Analog Station - 4 Port Authorization Code: Enables analog station ports of the BCM50 main chassis.</td>
</tr>
<tr>
<td>Digital Station Ports</td>
<td></td>
</tr>
<tr>
<td>- BCM50 DIGITAL STATION - 4 PORT AUTH CODE</td>
<td>BCM50 Digital Station - 4 Port Authorization Code: Enables digital station ports of the BCM50 main chassis.</td>
</tr>
<tr>
<td>- BCM50 DIGITAL STATION - 8 PORT AUTH CODE</td>
<td>BCM50 Digital Station - 8 Port Authorization Code: Enables digital station ports of the BCM50 main chassis.</td>
</tr>
<tr>
<td>- BCM50 DIGITAL STATION - 12 PORT AUTH CODE</td>
<td>BCM50 Digital Station - 12 Port Authorization Code: Enables digital station ports of the BCM50 main chassis.</td>
</tr>
<tr>
<td>Expansion Ports</td>
<td></td>
</tr>
<tr>
<td>- BCM50 EXPANSION 1 PORT AUTH CODE</td>
<td>BCM50 Expansion - 1 Port Authorization Code: Enables the two expansion ports on the BCM50 main chassis.</td>
</tr>
<tr>
<td>Miscellaneous</td>
<td></td>
</tr>
<tr>
<td>- BCM50 BT RACE S/W Auth Code</td>
<td>BCM50 BT RACE Software Activation Authorization Code</td>
</tr>
<tr>
<td>Try and Buy - 60 days</td>
<td></td>
</tr>
<tr>
<td>- BCM50 VPIM/AMIS Message Networking 60d</td>
<td>BCM50 VPIM/AMIS Message Networking 60d Authorization Code</td>
</tr>
<tr>
<td>----------------------------------------------------------</td>
<td>----------------------------------------------------------</td>
</tr>
<tr>
<td>BCM50 VoIP gateway - 2 Trunks 60d Authorization Code</td>
<td>BCM50 VoIP gateway - 2 Trunks 60d Authorization Code</td>
</tr>
<tr>
<td>BCM50 Fax Suite 60d Authorization Code</td>
<td>BCM50 Fax Suite 60d Authorization Code</td>
</tr>
<tr>
<td>BCM50 IP Telephony client - 4 Seats 60d Authorization Code</td>
<td>BCM50 IP Telephony client - 4 Seats 60d Authorization Code</td>
</tr>
<tr>
<td>BCM50 with Router Network Configuration Manager 60d Authorization Code</td>
<td>BCM50 without Router Network Configuration Manager (NCM) 60 day Authorization Code</td>
</tr>
<tr>
<td>BCM50 without Router Network Configuration Manager 60d Authorization Code</td>
<td>BCM50 with Router Network Configuration Manager (NCM) 60 day Authorization Code</td>
</tr>
</tbody>
</table>
Server/Client Software

Some server/client software is available for performing specific functions related to the BCM products. These include the server and client software applications used to manage multiple BCM systems within a network and IVR application development tools.

Table 27.

<table>
<thead>
<tr>
<th>Product</th>
<th>Server/Client Software</th>
</tr>
</thead>
<tbody>
<tr>
<td>BCM Network Configuration Manager</td>
<td>Network Configuration Manager 2.0 (250 systems), server/client software and documentation to support centralized configuration management for up to 250 systems</td>
</tr>
<tr>
<td>BCM Network Configuration Manager</td>
<td>Network Configuration Manager 2.0 (1000 systems), server/client software and documentation to support centralized configuration management for up to 1000 systems</td>
</tr>
<tr>
<td>BCM Network Configuration Manager</td>
<td>Network Configuration Manager 2.0 (2000 systems), server/client software and documentation to support centralized configuration management for up to 2000 systems</td>
</tr>
<tr>
<td>BCM Network Configuration Manager</td>
<td>Network Configuration Manager 2.0 Upgrade (from 1000 to 2000 systems)</td>
</tr>
<tr>
<td>Interactive Voice Response (IVR)</td>
<td>IVR software and documentation for development of IVR applications for BCM</td>
</tr>
<tr>
<td>PeriProducer IVR Application Development Tool</td>
<td></td>
</tr>
<tr>
<td>Interactive Voice Response (IVR)</td>
<td>IVR software and documentation for development of IVR audio prompts for BCM</td>
</tr>
<tr>
<td>PeriStudio IVR Audio Development Tool</td>
<td></td>
</tr>
</tbody>
</table>
Introduction
Hardware
Telephony
Data Capabilities
Messaging
Voice Over IP (VoIP)
Voice Networking
Call Center
Interactive Voice Response (IVR)
Mobility Solutions
Computer Telephony Integration (CTI)
Virtual Private Networks (VPN)
System Management and Software Options

> Glossary

Index
Glossary

**AbsorbLength**
A setting that determines how many of the digits in a destination code the system will not dial. AbsorbLength is assigned under Destination codes in Services.

**access code**
Different sequences of characters used to gain access to the following features: Line pools, Call park, external lines, Direct-Dial telephone and Auto DN.

**address**
A unique identifier assigned to networks and stations that allows each device to receive and reply to messages.

**Address Resolution Protocol (ARP)**
A protocol for mapping an IP address to a physical machine address that is recognized in the local network. For example, in IP Version 4, an address is 32 bits long. In an Ethernet local area network, however, addresses for attached devices are 48 bits long. The physical machine address is also known as a Media Access Control, or MAC, address. A table, usually called the ARP cache, maintains a correlation between each MAC address and its corresponding IP address. ARP provides the protocol rules for making this correlation and providing address conversion in both directions.

**alarm code**
A displayed number that informs users that a fault has been detected in the system.

**ANSI**
American National Standards Institute.

**Answer button**
A telephone button with an indicator that monitors another telephone. This button indicates incoming calls destined for the other telephone. Someone working at a telephone with answer buttons (an attendant, for example) can receive all ringing and visual indication of incoming calls for other telephones and answer those calls when necessary.
An Answer button is automatically assigned to a telephone when that telephone is assigned an Answer DN. One telephone can have up to four Answer buttons.
Answer DN
The internal or directory number (DN) of a telephone that is monitored by an Answer button. Users can assign up to four Answer DNs to a telephone under Line Access in Terminals and Sets programming.

Application Program Interface (API)
The specific method prescribed by an operating system or by another application program when a programmer writes an application program. It is used to make requests of the operating system or another application.
Unlike the graphical user interface or command interface, which are direct user interfaces, the API is an interface to an operating system or a program.

Application
A computer program that performs a wide range of tasks as specified by the user. Examples of application programs include word processing packages, spreadsheet packages and accounting packages.

ARP
See Address Resolution Protocol.

asynchronous
A method of transmission where the time intervals between characters are not required to be equal and signals are sourced from independent clocks with different frequencies and phase relationships. Start and stop bits may be added to coordinate character transfer.

Autobumping
A feature that determines what the system does with new Call Log items when a user’s Call Log is full. When Autobumping is on, a new log entry causes deletion of the oldest entry. If Autobumping is off, a user’s system does not log calls when the log is full.

autodial button
A memory button that a user can program to provide one-touch dialing of external or internal numbers.

autolog options
A feature that lets users select the type of calls their Call Log will store. Users can choose to log calls not answered by anyone within the system, calls that were unanswered at their telephone but answered elsewhere in the system, all calls answered and not answered at their telephone, or they can choose to not have calls automatically logged.
Automatic Dial
A feature that allows a user to dial without having to pick up the receiver or select a line. Users must have a prime line to use Automatic Dial, which is assigned under Dialing options in Terminals and Sets programming.

Automatic Handsfree
A feature that automatically activates Handsfree operation when a user makes or answers a call. It is assigned under Handsfree in Terminals and Sets programming.

Automatic Hold
A feature that automatically places an active call on hold when a user selects another line. Automatic Hold (Full AutoHold) is assigned in Lines programming.

Automatic Privacy
See Privacy.

Automatic Daylight Savings time
A feature that switches the system to standard or daylight savings time at preprogrammed times. It is turned on or off under Daylight time in System programming.

Automatic Telephone Relocation
A feature that lets a telephone retain its personal and system programming when it is plugged into a different modular jack. Automatic Telephone Relocation is enabled under Set relocation in System programming.

auxiliary ringer
A separate external telephone ringer or bell that a user can program to ring when a line or a telephone rings. An auxiliary ringer may be programmed to ring only when the system is in a particular schedule. Programming of an auxiliary ringer is done in Services programming after the feature has been enabled under Capabilities in Terminals and Sets programming.

B

B channel (Bearer channel)
An ISDN standard transmission channel used for voice or data transmission.
Background Music
A feature that allows users to hear music from a telephone’s speaker. It is available only if a music source has been attached to the system and the feature has been enabled under Feature settings in System programming.

Back up
To make a duplicate copy of data files in order to store the originals in a safe place. Backing up original files protects them from damage if a hardware failure occurs.

Base Station
A Digital Mobility component that is mounted on walls and ceilings to provide a radio link to an office or other area where Digital Mobility portable telephones are used. Each Base Station houses two radios that allow portables to send and receive calls through the BCM server.

baud rate
A unit of measurement of data transmission speed. It is approximately equivalent to bits per second (BPS). Typical baud rates are 300, 1200, 2400, 4800 and 9600.

Bearer channel
See B channel.

BERT
See bit error rate test.

BIOS (Basic Input Output System)
A program contained in Read Only Memory (ROM) that acts as the interface between software programs and the computer hardware.

bit
An abbreviation for Binary Digit. It is the smallest unit of information a computer recognizes. A bit has one of two values (0 or 1) to indicate off or on.

bit error rate test (BERT)
A test that checks the transmission of data across the voice and data channels between the system and any telephone.

bps (Bits Per Second)
The speed of data transmission between two computers.
bus
A collection of communication lines that carry electronic signals either between elements on the system board or between the circuitry on the system board and any cards plugged into the system board.

Business Series Terminals

T7100 telephone
The Business Series Terminals T7100 telephone has a single-line display and one memory button without an indicator.

T7208 telephone
The Business Series Terminals T7208 telephone that has a single-line display and eight memory buttons with indicators.

T7316E telephone
The Business Series Terminals T7316E telephone that has a two-line display with three display buttons, 16 memory buttons with indicators and 8 memory buttons without indicators.

Byte
The amount of space required to store a single character. One byte equals eight bits.

C

Call Duration timer
A feature that lets users see how long they spent on their last call or how long they have been on their present call.

Call Forward
A feature that forwards all the calls from a user’s telephone to another telephone in the system. Line Redirection allows users to forward calls outside the system.

Call Forward No Answer
A feature that forwards all calls arriving at your telephone to another designated telephone in your system after a specific number of rings. Call Forward No Answer is assigned under Capabilities in Terminals and Sets programming.

Call Forward On Busy
A feature that forwards all calls at a user’s telephone to another designated telephone if his or her telephone is busy. This feature is assigned under Capabilities in Terminals and Sets programming.
Call Forward Override
An automatic system feature that lets users stop call forwarding directed to their set.

Call Information
Displays information about incoming calls. For external calls, the set can display the caller’s name, telephone number and the line name. For an internal call, it can display the name of the caller and his or her internal number. Users can obtain information about ringing, answered or held calls.

Call Log
A feature that lets users view a record of incoming calls. The Log could contain the following information for each call: the sequence number in the Call Log; the caller’s name and number; long-distance indication; indication if the call was answered; the call’s time and date; the number of repeated calls from the same source; and the name of the line on which the call came in. See Autobumping, Autolog options and Call Log for further information.

Call Park
A feature that lets a user place a call on hold so that someone can retrieve it from any other telephone in the system by selecting an internal line and entering a retrieval code. The retrieval code appears on the user’s telephone display when he or she parks the call. Users can park up to 25 calls on the system at one time.

Call Park Callback
See Callback.

Call Park prefix
The first digit of a parked call’s retrieval code. This digit cannot conflict with the first digit of any existing DNs, Line Pool access codes, the Direct-Dial digit or the external line access code. The default Call Park prefix digit is “1.” Users can set it to zero, in which case Call Park is disabled. The Call Park prefix is assigned under Access codes in System programming.

Call Pickup Directed
A feature that lets users answer a call ringing at any telephone by entering the internal number of that telephone before taking the call. Call Pickup Directed is enabled under Feature settings in System programming.

Call Pickup Group
See Pickup Group.
Call Queuing
A feature that allows users to answer calls in order of priority, when they have several calls waiting at their telephone. Priority is given to incoming calls, followed by callback and camped calls.

Callback
A feature that notifies users when calls that they have parked, camped, or transferred to another telephone are not answered. When this occurs, the call rings again at the user’s telephone. How long the system will wait before Callback occurs is set under Feature settings in System programming.

Camp-On
A feature that lets you reroute a call to a telephone even if all the lines on that telephone are busy. To answer a camped call, use Call Queuing or select a line if the camped call appears on your telephone. Queued calls are prioritized over camped calls.

Camp timeout
The length of a delay before a camped call is returned to the telephone that camped the call. The length of delay is set under Feature settings in System programming.

Channel Service Unit (CSU)
A device on the Digital Trunk Interface that is the termination point of the T-1 lines from the T-1 provider. The CSU collects statistics on the quality of the T-1 signal. The CSU ensures network compliance with FCC rules and protects the network from harmful signals or voltages.

Challenge-Handshake Authentication Protocol (CHAP)
A method of establishing security on PPP links where the peers must share a plain text identifier. The caller sends a challenge message to its receiving peer and the receiver responds with a value it calculates based on the identifier. The first peer then matches the response with its own calculation. If the values match, the link is established.
CHAP is a more secure procedure for connecting to a system than the Password Authentication Procedure (PAP).

Class of Service (COS)
The set of features and lines available to the user for a call. The Class of Service for a call is determined by the restriction filters and remote access packages assigned to the telephone in Lines programming. The Class of Service for a call can be changed by entering a six-digit Class of Service password. (Internal users cannot change their access to features with a COS password, only their restriction filters.) Class of Service and Class of Service passwords are assigned in Passwords programming. See Remote Access.
Class of Service password
A six-digit code that lets you switch from your current Class of Service to one that lets you dial numbers prohibited by your current Class of Service.

cold start
A complete startup of a system after the loss of all system programming. This can happen after a major event such as an extended power failure.

Digital Mobility Wireless
The name for the communication systems which use radio technology to transmit and receive signals between its components and the Business Communication Manager server. Digital Mobility Wireless provides mobility in the workplace. Calls that used to ring just at your telephone set can also appear and ring at your portable.

Digital Mobility portable telephone
Handheld wireless telephones that allow complete mobility within the reach of Digital Mobility Base Stations or an external antenna. Portables offer many but not all standard system features and share much of the same programming as “wired” desk telephones.

Conference
A feature that allows you to establish a three-person call at your telephone.

Conventions
The way certain information is described. For example, using underlined text to represent second-line display prompt information.

COS
See Class of Service.

D

D channel (Data channel)
An ISDN standard transmission channel which is packet-switched and is used for call setup, signaling and data transmission.
Data channel
See D channel.

Data link connection identifier (DLCI)
A unique identifier of a PVC in frame relay networks.

Defaults
The settings for all features when the system is first installed. Settings are changed from their defaults in programming.

Delayed Ring Transfer (DRT) to prime
A feature that, after a specified number of rings, transfers an unanswered call on an external line to the prime telephone associated with that line. This feature is activated under Feature settings in System programming.

Destination code
A two- to seven-digit number that the system interprets and then translates into the digits that you want dialed out. Both the code and its associated dialed digits are assigned under Routing service in Services programming.

DHCP
See Dynamic Host Configuration Protocol.

dialing restriction
See Restriction filter.

dialup connection
A temporary connection between computers that is established over an analog or digital phone line.

Differentiated Services (DiffServ)
An implementation methodology for QoS service for IP networks. DiffServ is a rule-based methodology intended to improve network performance. Instead of applying faster, more advanced technology, networks are managed by appropriate network policies. With DiffServ there is a cost associated with higher quality services and a risk with lower quality services.

Direct-dial
A feature that lets you dial a designated telephone in your system with a single digit. As many as five direct-dial sets can be established. Each telephone in the system is assigned to one direct-dial telephone. There is a single, systemwide digit for calling the assigned direct-dial telephone of any telephone. Direct-dial telephones are established in System programming. Telephones are assigned to a direct-dial telephone under Capabilities in Terminals and Sets programming.
Direct-dial #
A digit used systemwide to call the direct-dial telephone. The digit is assigned under Access codes in System programming.

Direct-dial number
The digit used to call the direct-dial telephone.

directed pickup
See Call Pickup Directed.

Directory number (DN)
A unique number that is automatically assigned to each telephone or data terminal. The DN, also referred to as an internal number, is often used to identify a telephone when settings are assigned during programming.

Disconnect Supervision
A setting that enables the system to detect if an external caller hangs up. Once an external caller hangs up, the system can disconnect its line. Disconnect Supervision is enabled under Trunk/Line data in Lines programming.

Disk drive
A mass storage device that seeks, reads and writes data on a disk.

Display
A one-line or two-line screen on a BCM telephone that shows commands and options.

Display buttons
The three buttons that appear underneath a BCM two-line LCD display.

Display options
The choices available to a user that appear on the BCM two-line display. Options appearing on the display can be selected using the display or dial pad buttons.

DLCI
See Data link connection identifier.

DN
See Directory number.

DNS
See Domain Name Server.
domain name
A string used to organize Internet names into manageable groups, such as nortelnetworks.com.

Domain Name Server (DNS)
The system in the Internet that maps names of objects, most usually host names, into IP numbers or other resource record values. The namespace of the Internet is divided into domains and the responsibility for managing names within each domain is delegated, typically to systems within each domain.

Do Not Disturb
A feature that stops calls from ringing at your telephone. Only Priority Calls will ring at your telephone. A line button will flash when you receive a call, but the call will not ring.

Digital Private Network Signaling System (DPNSS)
A networking protocol that gives operators access to BCM features over multiple combined networks in International systems only. Corporate offices, separated geographically, can be linked over DPNSS to other BCM systems, bypassing the restrictions of the PSTNs to which they may be connected. This allows connected BCM systems to function like a private network.

Driver (Device)
A program that allows a hardware peripheral, such as an NIC, to communicate with the Business Communications Manager server.

DTMF
See dual tone multifrequency.

dual tone multifrequency
Two distinct telephone signaling tones used for dialing.

Dynamic Host Configuration Protocol (DHCP)
A protocol that lets network administrators centrally manage and automate the assignment of IP addresses in an organization’s network. Using the Internet’s set of protocols (TCP/IP), each machine that can connect to the Internet needs a unique IP address. When an organization sets up its computer users with a connection to the Internet, an IP address must be assigned to each machine. Without DHCP, the IP address must be entered manually at each computer and, if computers move to another location in another part of the network, a new IP address must be entered. DHCP lets a network administrator supervise and distribute IP addresses from a central point and automatically sends a new IP address when a computer is plugged into a different place in the network.
Emergency 911 dialing
The capability to access a public emergency response system by dialing the digits 9-1-1. 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.

emergency telephone
A single-line telephone (also referred to as a 500/2500 telephone) that becomes active when there is no power to the Business Communications Manager server.

Ethernet
A widely used Local Area Network (LAN) protocol that is the original Carrier Sense Multiple Access/Collision Detect (CSMA/CD) LAN that lets PCs and/or Business Communications Manager servers listen for pauses before they communicate. Ethernet LANs use coaxial cable or twisted-pair wiring for connecting computers.

evening schedule
See Schedules and Services.

event message
A message that is stored in the system log and displayed during a Maintenance session. Event messages record a variety of events and activities in the system.

exceptions
See Overrides.

Expansion Media Bay Modules
Modules that connect expansion modules to the BCM system. There is one expansion media bay module available for the BCM system; the Fiber Expansion Media Bay Module (EE-FEM). The EE-FEM connects up to six Norstar expansion modules to the BCM system.

Extended Data-Out (EDO)
A type of dynamic random access memory (RAM) in which storing data to and reading data from the memory is performed at a faster rate.
**external code**
The number you dial to get an external line. By default it is “9,” but this can be changed under Access codes in System programming. You do not always need an external code. It is primarily to support the T7100 telephone.

**external line**
A line on your telephone used for making calls to destinations outside the system.

**external music source**
See Music source.

**external paging**
A feature you can use to make voice announcements over an externally-mounted loudspeaker connected to the Business Communication Manager server. The external speaker is not a BCM component and must be supplied by the customer.

**F**

**FAX**
A feature that works with BCM Voice Messaging, offering a caller the capability of sending a fax document to a mailbox as easily as sending a voice message.

**Feature Code**
A unique code used to access BCM features and options.

**File**
A collection of related information stored on a disk under a given name for later reference and used by an operating system or application program. Each application program that you use saves the data you create in files. Files are identified by a file name and optional extension.

**File name**
A name that identifies a file and consists of one to eight characters.

**filtering**
The process of examining a data packet on the network to determine the destination of the data and whether the packet should be passed along on the local LAN, copied to another LAN or dropped.
Forward
See Call Forward.

frame
A unit of data transmission in a local area network.

frame relay
A high-speed, packet switching WAN protocol designed to provide efficient, high-speed frame or packet transmission with minimum delay. Frame relay uses minimal error detection and relies on higher level protocols for error control.

FTP
The file transfer protocol (FTP) allows a user on one host to access and transfer files to and from another host over a network. On the Internet, FTP refers to a tool for accessing linked files.

Full Autohold (on idle line)
When this feature is on, if you select an available line and then do something that selects another line, the first line is put on hold. Full Autohold is enabled under Trunk/Line data in Lines programming.

Full Handsfree
See Handsfree.

Fully Qualified Domain Name (FQDN)
The combination of host name and domain name. For example mycomputer.nortelnetworks.com is Fully Qualified Domain Name.

G

Ground Start trunk
Ground start trunks offer the same features as loop start trunks, but are used when the local service provider does not support disconnect supervision for the digital loop start trunks. By configuring lines as ground start, the system will be able to recognize when a call is released at the far end. Ground start trunks are provided only by a Digital Trunk Interface (Dti).

Group Listening
A feature that allows you to have others in your office hear a caller through your telephone speaker. The caller hears you only when you speak into the receiver and cannot hear other people in the office.
You can cancel Group Listen for the current call. Group Listen is cancelled automatically when you hang up the Group Listen call.

**H**

**H.323**

H.323 is the standard for using IP to send voice and video within intranets and on the public Internet.

**Handsfree**

A feature you can use to make calls without using the telephone receiver. Full Handsfree is activated under Capabilities in Terminals and Sets programming. When it is activated, a Handsfree/Mute button is automatically assigned to the telephone.

**Handsfree (HF) Answerback**

When activated, this feature automatically turns on the microphone at a telephone receiving a Voice Call so that the person receiving the call can respond without lifting the receiver. It is activated under Capabilities in Terminals and Sets programming.

**Handsfree/Mute button**

See Handsfree.

**Hard disk drive**

A data storage device that uses nonremovable, rigid magnetic platters. Hard disk drives work faster and store more data than disk drives do for diskettes.

**Hardware**

The physical components of the BCM system.

**HDLC**

See High-level Data Link Control.

**Headset**

A head-mounted or ear-mounted telephone receiver that is used instead of the handheld receiver. Headsets are not BCM system components and must be supplied by the customer.
**Held (Line) Reminder**
A telephone rings and displays the message “On hold: LINENAM” when an external call has been placed on hold for a certain period of time. The Held Line Reminder feature and Remind delay are set under Feature settings in System programming.

**HF Answerback**
See Handsfree Answerback.

**High-Level Data Link Control (HDLC)**
HDLC is a group of protocols or rules for transmitting data between network points or nodes. Data is organized into a unit, called a frame and sent across a network to a destination that verifies its successful arrival. The HDLC protocol also manages the flow or pacing at which data is sent. HDLC is one of the most commonly used protocols in Layer 2 of the industry communication reference model, Open Systems Interconnection (OSI).

**Hold button**
This button is used to suspend calls so that the person using the telephone can perform another task without disconnecting the caller.

**Hook Switch Flash**
See Link time.

**Host Name**
In networking, the name of a computer that primarily provides services, such as database access, to other computers or Business Communications Manager servers in the domain. The host name is associated with a unique IP address. Since the Business Communications Manager server has a unique IP address, it qualifies as a host.

**Host system signaling**
Also referred to as end-to-end signaling. Telephones can access a remote system or dial a number on an alternate carrier by means of host feature activation, such as Link, Pause and Run/Stop.

**Hotline**
This feature automatically calls a preassigned number when the telephone’s receiver is lifted or the Handsfree/Mute button is pressed. A Hotline number can be an internal or external number. Hotline is assigned under Capabilities in Terminals and Sets programming.

**HTTP**
The Hypertext Transfer Protocol (HTTP) is the set of rules for exchanging text, graphic images, sound, video and other multimedia files on the World Wide Web.
HTTP proxy
See Web proxy.

Hz (hertz)
A unit of measure for indicating frequency in cycles per second.

I

ICMP
ICMP is a message control and error-reporting protocol between a host server and a gateway to the Internet. ICMP uses IP datagrams, however the messages are processed by the TCP/IP software and are not directly apparent to the application user.

I/C
An abbreviation of intercom.

IETF
See Internet Engineering Task Force.

Initialization
The steps required to prepare hardware or software for operation.

Install
To set up for operation. For example, hardware is installed by attaching it to the appropriate connectors or sockets either inside or outside the Business Communication Manager server.

Integrated Services Digital Network (ISDN)
A digital telephone service that allows for a combination voice and data connection over a single, high-speed connection. ISDN service can operate over the same copper twisted-pair telephone line as analog telephone service.

intercom button
A button that provides access to internal lines used for calls within a BCM system and access to external lines through a line pool or external code. A telephone may be assigned zero to eight Intercom buttons. This is done under Line Access in Terminals and Sets programming.

intercom keys
See Intercom button.
Interface
An information interchange path that allows communication between computer parts.

internal line
A line on your telephone dedicated to making calls to destinations inside your system. An internal line may still connect you with an external caller if you use it to access a line pool or to pick up a call using the call handling features such as Call Park or Call Pickup Directed.

internal number
A number (also referred to as a Directory Number or DN) that identifies a telephone or device.

internal user
Someone using a BCM telephone within the system.

Internet
A global TCP/IP network linking millions of computers for communications purposes.

Internet Engineering Task Force (IETF)
The IETF is the committee that defines standard Internet operating protocols such as TCP/IP. The IETF is supervised by the Internet Society’s Internet Architecture Board (IAB).

Internet-Standard Network Management Framework
Device configuration and monitoring via SNMP.

IP
The Internet Protocol (IP) is the protocol that supports data being sent from one computer to another on the Internet. Each computer on the Internet has at least one address that uniquely identifies it from all other computers on the Internet. When you send or receive data, the message gets divided into units called packets. Each of these packets contains both the sender’s Internet address and the receiver’s address. IP is a connectionless protocol, which means that there is no established connection between the endpoints that are communicating. Each packet that travels through the Internet is treated as an independent unit of data without any relation to any other unit of data. In the Open Systems Interconnection (OSI) communication model, IP is in Layer 3, the Networking Layer.

IP address
The Internet Protocol address is a unique identifier that allows communication over the Internet to be directed to the appropriate destination. Every computer on the Internet must have a unique IP address. IP addresses are allocated by an ISP in the following format: nnn.nnn.nnn.nnn, where nnn is a numeric value from 0 to 255. IP addressing might be referred to as being static (fixed) or dynamic.
**IRQ (Interrupt Request)**
A signal sent by a hardware device to the microprocessor requesting its immediate attention. For example, each communications port has an Interrupt Request line for notifying the microprocessor when data has been received or transmitted.

**IRQ Conflict**
Two hardware devices are vying for the same IRQ. On installation of a device where an IRQ conflict occurs, the user may have to manually configure the IRQ settings to resolve the conflict.

**ISDN**
See Integrated Services Digital Network.

**ISDN DN**
A directory number (DN) used by ISDN terminal equipment connected to the system. The BCM system uses a maximum of thirty ISDN DNs.

**K**

**Kbyte**
The abbreviation for kilobyte. A kilobyte is equal to 1024 bytes.

**L**

**LAN**
An LAN is a network of interconnected workstations sharing the resources of a single processor or server within a relatively small geographic area.

**Last Number Redial**
A feature that allows you to redial the last external number you dialed.

**Least cost routing**
See Routing service.

**line**
The complete path of a voice or data connection between one telephone (or other device) and another.
Lines
A programming section that lets you assign settings to each trunk and external line.

Line number
A number that identifies an external line. The total number of lines depends on the number and type of trunk Media Bay Modules installed.

Line Pool
A group of lines used for making external calls. Line pools provide an efficient way of giving a telephone access to external lines without taking up many line buttons. A line is assigned to be part of a line pool under Trunk/Line data in Lines programming.

Line Redirection
A feature that allows you to redirect all calls on an incoming line to a destination outside the system. Once a line is redirected it cannot be answered within the system. The system may be set up to give a brief ring when a call comes in on a redirected line, under Capabilities in Terminals and Sets programming. This feature differs from Call Forward in two ways. It redirects only external calls (not internal calls), and it redirects calls to destinations outside the system. Call Forward redirects calls only to destinations inside the system. See Call Forward.

Link
If your BCM system is connected to a Private Branch Exchange (PBX), you can use a Link signal to access special features. The Link signal can also be included as part of a longer stored sequence on an External Autodial button or in a Speed Dial code. The Link symbol uses two of the 24 spaces in a dialing sequence.

Local Area Network (LAN)
A group of computers or Business Communications Manager servers physically connected in a manner that lets them communicate and interact with each other.

Long Tones
A feature that lets you control the length of a tone so that you can signal devices such as fax or answering machines which require tones longer than the standard 120 milliseconds.

Lunch schedule
See Schedules and Services.
MAC
The Media Access Control (MAC) is a physical address that is the portion of the data-link layer in 802.x networks that controls addressing information of the packet and enables data to be sent and received across a local area network.

Maintenance
A type of programming that is used to diagnose and repair problems in the BCM system. Maintenance requires no programmable settings.

Mailbox
A storage place for voice messages on BCM Voice Messaging.

Communication Server 1000 ISDN Primary Rate Interface
A protocol used between members of Nortel Meridian family of Private Telecommunication Network Exchanges. The signaling information is carried via time slot 16 of a 2.048 Mbit/s digital transmission system.

message
A feature that allows you to send a message to another system user. The Message feature also lets you know if you have any messages waiting and maintains a Message Waiting List to keep a record of your internal messages and your (external) messages.

MHz
The abbreviation for megahertz. This is a unit of measure indicating frequency in millions of cycles per second.

microprocessor
A chip that is the center of all activity inside the Business Communications Manager server. The microprocessor controls all logical and arithmetic operations for the computer and is responsible for executing program commands. It is also referred to as the Central Processing Unit (CPU).

Modem
A communications device that allows data to be exchanged between computers over telephone lines. The exchange is done by electronic processes called modulation and demodulation. The modem changes (modulates) the data into tones to send to another modem and also converts (demodulates) tones when receiving from another modem.

Move Line buttons
A feature that allows you to move external lines to different buttons on your telephone.
Multilink PPP
Multilink PPP is an extension to the PPP protocol that enables you to group a set of links into a bundle for more bandwidth. The links in the bundle can operate at different speeds. Typical links can be ISDN B channels, dialup connections and leased lines.

Music source
A radio or other source of music can be connected to the system to provide music for the Music on Hold and Background Music features. A music source is not part of the BCM system and must be supplied by the customer.

Names
Names can be assigned to System Speed Dial numbers, external lines, telephones, mailboxes, ACD Queues and service schedules. This is done in programming. You can use up to sixteen characters to name a System Speed Dial number, 13 characters for mailbox and ACD Queue names and seven characters to name a telephone, line or schedule. If a name has not been assigned, the line number or DN will appear on the display instead of a name.

name server
A name server provides the means of translating readable host computer names into actual IP addresses so you do not have to remember long numbers in order to access other computers and destinations on the Internet. For example, DNS servers and WINS servers are name servers.

NetBIOS
The Network Basic Input/Output System (NetBIOS) is an interface and upper-level protocol developed by IBM for use with a proprietary adapter for its PC network product. NetBIOS provides a standard interface to the lower networking layers. The protocol provides higher-level programs with access to the network. Windows NT systems use NetBIOS.

Network
Two or more computers linked together electronically to share programs and exchange data. Joining computers over a network requires adding specialized hardware and software to each computer.

network device
A network device is a hardware entity characterized by its use as a communications component within a networking infrastructure.
Network DN

A number supplied by the ISDN network service provider for ISDN terminal equipment.

Network Interface Card (NIC)

An adapter card containing the hardware necessary to connect a Business Communications Manager server to a local area network.

NIC

A network interface card (NIC) is a computer circuit board or card that is installed in a computer so that it can be connected to a network.

Personal computers and workstations on local area networks (LANs) typically contain a network interface card specifically designed for the LAN transmission technology, such as Ethernet or Token Ring. Network interface cards provide a dedicated, full-time connection to a network.

Night schedule

See Schedules, Services.

O

On hold

A setting, programmed under Feature settings in System programming, that controls whether external callers hear music, periodic tones or silence when they are placed on hold.

Operating system

The disk-based software that manages the operation of the Business Communication Manager server. An operating system controls the flow of information between the computer hardware. Windows NT is the operating system that manages the Business Communication Manager server.

Option

A Business Communications Manager server choice that is given to a user through display prompts.

OPX

Off-premise extension.

Out-of-Band

Generally this means “not in the speech path.” For example, when you are on a call with a wireline set and you press a digit on the keypad, the phone does not generate the DTMF. That would be in-band. The phone
sends an out-of-band keypress indication to the BCM system which then produces the tone that is heard by the callers, in-band.

**Overflow**

A setting in Routing Service that allows users to decide what path an outgoing call will take if all the lines used in a particular route are in use when the call is made.

** Overrides**

One component of a restriction filter. Overrides are numbers you can dial even if they are forbidden by a more general restriction. See Restrictions.

**P**

**Packet**

A packet is the unit of data that is routed between an origin and a destination on the Internet or any other packet-switched network. When any file (email message, HTML file, GIF file, URL request and so forth) is sent from one place to another on the Internet, the Transmission Control Protocol (TCP) layer of TCP/IP divides the file into pieces of an efficient size for routing. Each of these packets is separately numbered and includes the Internet address of the destination. The individual packets for a given file may travel different routes through the Internet. When the packets have all arrived, they are reassembled into the original file. A packet-switching scheme is an efficient way to handle transmissions on a connectionless network such as the Internet. An alternative scheme, circuit-switching, is used for networks allocated for voice connections. In circuit-switching, lines in the network are shared among many users as with packet-switching, but each connection requires the dedication of a particular path for the duration of the connection. Packet and datagram are similar in meaning. A protocol similar to TCP, the User Datagram Protocol (UDP) uses the term datagram.

**Page**

A feature you can use to make announcements over the BCM system. You can make page announcements over the telephone speakers and/or external speakers.

**Page Time-Out**

A setting that controls how long a Page Announcement can last. It can be assigned under Feature settings in System programming.

**Page zone**

An area in the office that receives internal page announcements independently of the rest of the office.
Each page zone is identified by a number. Telephones are assigned to page zones under Capabilities in Terminals and Sets programming.

**PAP**
The Password Authentication Protocol (PAP) is a procedure used by PPP servers to validate a connection request. PAP works as follows: After the link is established, the requestor sends a password and an ID to the server. The server either validates the request and sends back an acknowledgment, terminates the connection or offers the requestor another chance. Passwords are sent without security and the originator can make repeated attempts to gain access. For these reasons, a server that supports CHAP will offer to use that protocol before using PAP.

**Parallel port**
A port that transfers data through multiple wires so that eight bits are transmitted simultaneously. Parallel ports usually use a 25-pin interface that transmits and receives data one byte at a time using a separate data line for each bit.

**Park prefix**
See Call park prefix.

**Park timeout**
The time before an unanswered parked call is routed back to the telephone that parked it. Park timeout is under Feature settings in System programming.

**Password**
A four-digit to eight-digit number that is entered using the dial pad. A password is used to open mailboxes or perform configuration tasks.

**Pause**
A feature that enters a 1.5 second delay in a dialing sequence on an external line. This is often required for signaling remote devices, such as answering machines, or when reaching through to PBX features or host systems. The Pause symbol uses one of the 24 spaces in a dialing sequence.

**PBX**
See Private Branch Exchange.

**Permanent virtual circuit (PVC)**
The PVC is an end-to-end virtual connection in frame relay networks.

**Peripheral Component Interconnect (PCI) Slot**
Socket on the Business Communication Manager server main board that connect to the BCM cards.
Personal Speed Dial
Two-digit codes (71-94) can be programmed to dial external telephone numbers. Personal Speed Dial numbers are programmed for each telephone and can be used only at the telephone on which they are programmed.

Pickup Group
A telephone can be placed into one of nine call pickup groups. A call ringing at a telephone within a pickup group can be picked up at any other telephone within the same pickup group. A telephone is assigned to a pickup group under Capabilities in Terminals and Sets programming.

Pin-1
The first pin in a multiple-pin connector, or chip designated as such, to help you properly orient the component when attaching or installing it.

Point-to-point protocol (PPP)
PPP is a protocol for communication between two computers using a serial interface; typically a personal computer connects to a server by a phone line. For example, your Internet server provider may provide you with a PPP connection so that the provider’s server can respond to your requests, pass them on to the Internet and forward your requested Internet responses back to you.
PPP is a full-duplex protocol that can be used on various physical media, including twisted-pair or fiber optic lines or satellite transmission. It uses a variation of High Speed Data Link Control (HDLC) for packet encapsulation.
PPP can process synchronous as well as asynchronous communication. PPP can share a line with other users and it has error detection.

Pool
See Line pool.

Port
A connector on the Business Communication Manager server that allows data exchange with other devices, such as a printer or mouse.

portable telephone
See Digital Mobility portable telephone.

PPP
See Point-to-Point protocol.
Predial
A feature that allows you to enter a number and check it on your telephone display before it is actually dialed. If the number is incorrect, you can edit it. The number is dialed only when you pick up the receiver or select a line.

Primary Rate Interface (PRI)
An ISDN interface which uses 23 B channels and a D channel (23B+D).

Prime Line
The line on your telephone that is automatically selected when you lift the receiver, press the Handsfree/Mute button or use an external dialing feature. A Prime Line is assigned to a telephone under Line access in Terminals and Sets programming.

Prime Set (prime telephone)
A telephone that provides backup answering for incoming calls on external lines. The prime telephone for a line will ring for any unanswered calls on that line. A prime telephone is assigned to a line under Trunk/Line data in Lines programming.

Priority Call
If you get a busy signal when you call someone in your office, you can interrupt them for an urgent call. This feature is enabled for a telephone under Capabilities in Terminals and Sets programming.

Privacy
This feature determines whether a system user may select a line in use at another telephone and join an established call. Privacy is enabled under Trunk/Line data in Lines programming, but can be turned on and off by users during individual calls.

Private branch exchange (PBX)
A PBX is a telephone system within an enterprise that switches calls between enterprise users on local lines while allowing all users to share a certain number of external phone lines. The main purpose of a PBX is to save the cost of requiring a line for each user to the telephone company’s central office. The PBX is owned and operated by the enterprise rather than the telephone company.

Private line
See Private to.

Private network
A telephone network consisting of owned or leased telephone lines used to connect different offices of an organization independently of the public network.
Private to
Let's you select the telephone that will use the line exclusively. The line cannot appear on any other telephone, except the prime telephone for that line. Private lines cannot be placed into line pools. Private lines are assigned under Trunk/Line data in Lines programming.

programming
Setting the way the BCM system will work. Programming includes system-wide settings and individual telephone and line settings.

Protocol
A set of rules and procedures for exchanging data between computers or Business Communications Manager servers on a network or through the Internet.

Proxy
A proxy is a server that acts on behalf of another.

public line
An external line that can be assigned to any telephone and to many telephones. A line is assigned as Public under Trunk/Line data in Lines programming.

public network
The regular telephone network that connects most homes and businesses.

pulse/tone dialing
An external line setting for pulse or tone dialing. Pulse is the traditional method of dialing used by rotary-dial or push-button single-line telephones. Tone dialing allows telephones to communicate with other devices such as answering machines. Tone dialing is required to access the features that PBX systems may offer or to use another system remotely.

PVC
See Permanent virtual circuit.

Q
Quality of Service (QoS)

On the Internet and in other networks, QoS is the idea that transmission rates, error rates and other characteristics can be measured, improved and, to some extent, guaranteed in advance. QoS is of particular concern for the continuous transmission of high-bandwidth video and multimedia information. Using the Internet’s Resource Reservation Protocol (RSVP), packets passing through a gateway host can be expedited based on policy and reservation criteria arranged in advance. Using ATM, which also lets a company or user preselect a level of quality in terms of service, QoS can be measured and guaranteed in terms of the average delay at a gateway, the variation in delay in a group of cells (cells are 53-byte transmission units), cell losses and the transmission error rate.

In BCM, QoS is provided over IP. QoS is guaranteed for outgoing traffic until it reaches the next hop.

QoS

See Quality of Service.

Q reference point signaling (QSIG)

QSIG is an ETSI standard signaling for multivendor peer-to-peer communications between PBXs and/or central offices.

R

RAM (Random Access Memory)

Computer memory that stores data temporarily. RAM stores the data used by the microprocessor as it executes instructions. The contents of RAM are erased each time the Business Communication Manager server is turned off or restarted.

recall

See Link time.

receiver

The handset of a telephone.

Regression Code

Restores the previous system security number so that previously applied UTAM Activation Codes and Portable Credit Codes can be reentered to restore full system operation. Also required in cases of system recovery. This code cannot be reused.
**Relaying**
Relaying is the process of moving data along a path determined by a routing process. The data is relayed between a source and a destination.

**Remind delay**
A feature that causes a telephone to beep and display the message On hold: LINENAM when a call has been on hold for a programmable period of time. This period is the Remind delay.

**Remote access**
The ability to dial into a BCM system from outside the system and make use of selected features. The lines, features and dialing capabilities available to a remote user are determined by the Class of Service.

**remote access dial restriction**
See Remote restriction.

**Remote access service (RAS)**
The RAS is the ability to get access to a computer or a network from a remote distance. In corporations, people at branch offices, telecommuters and people who are traveling may need access to the corporation’s network. Home users get access to the Internet through remote access to an Internet service provider (ISP).
A remote access server is the computer and associated software that is set up to handle users seeking access to network remotely. Sometimes called a communication server, a remote access server, usually includes or is associated with a firewall server to ensure security, and a router that can forward the remote access request to another part of the corporate network.

**Remote capability**
A subset of BCM features that are available to users connected through remote access.

**Remote device**
A remote device is any network device that is accessible only by means of communication over a digital or analog (dialup) network.

**Remote monitoring**
A feature that lets an off-site technician with a PC call in and troubleshoot your system through the built-in modem.

**Remote paging**
This feature allows remote users to use the system paging feature. Access to this feature is governed by the Class of Service for the call. See Remote access and Class of Service.
**Remote restriction**
A restriction filter applied to a line in order to control which digits can be dialed during an incoming remote access call. It is the equivalent of a set filter for a remote user.

**remote user**
Someone who calls into a BCM system from a telephone outside that system and uses system features or lines. See Remote access.

**Restriction filter**
Through a combination of restrictions and overrides, restriction filters prevent certain telephone numbers or feature codes from being dialed. Restriction filters can be applied to lines, sets, specific lines on a set and to Class of Service passwords.

**Restriction service**
A Services section that allows you to assign alternate dialing filters to lines, telephones, lines on a particular telephone and alternate remote filters to lines at specified times of the day and on specified days.

**Restrictions**
One component of a Dialing filter. Restrictions are numbers you cannot dial when that dialing filter is in effect. See Exceptions.

**Ring Again**
A feature that can be used when you cannot get through to someone on your system because their telephone is busy or there is no answer. Ring Again instructs the system to inform you when they hang up or next use their telephone.

**Ring group**
A setting under Services that lets you assign a number of different telephones to ring during one of the schedules. Up to 20 ring groups can be programmed by an installer or a system coordinator.

**Ring type**
A feature that allows you to select one of four distinctive rings for your telephone.

**Ring volume**
A feature that allows you to set the volume at which your telephone rings.

**Ringing service**
A Services section that allows you to make additional telephones ring at specified times of the day and on specified days.
RIP
See Routing Information Protocol.

Rls button
Ends a call in the same way that hanging up the receiver does.

ROM (Read Only Memory)
Memory that stores data permanently. ROM contains instructions that the Business Communications Manager server needs to operate. The instructions stored in ROM cannot be changed and are used by the Business Communications Manager server each time it is turned on or restarted.

Router
A router is a device that forwards traffic between networks, based on network layer information and routing tables. A router decides which path network traffic follows using routing protocols to gain information about the network and algorithms to choose the best route based on a routing matrix.

Routing
The path a message takes from its point of origin to its destination on a network or the Internet.

Routing Information Protocol (RIP)
RIP enables routers in the same autonomous system to exchange routing information by means of periodic updates. RIP is a widely-used protocol for managing routing information within a self-contained network such as a corporate local area network (LAN) or an interconnected group of such LANs.

Using RIP, a gateway host (with a router) sends its entire routing table (which lists all the other hosts it has on record) to its closest neighbor host every 30 seconds. The neighbor host passes the information to its next neighbor and so on until all hosts within the network have the same routing path information, a state known as network convergence. RIP uses a hop count as a way to determine network distance. Each host with a router in the network uses the routing table information to determine the next host to route a packet to for a specified destination.

RIP is considered an effective solution for small homogeneous networks. For larger, more complicated networks, RIP’s transmission of the entire routing table every 30 seconds may put a heavy amount of extra traffic in the network.

The major alternative to RIP is the Open Shortest Path First Protocol (OSPF).

Routing service
A programming section that allows outgoing calls to be directed automatically based on the numbers a caller dials. For BCM servers linked in a network, routing can create a transparent or coordinated dialing plan. It can also be used to direct calls to the least expensive lines according to a Services schedule (sometimes called least cost routing).
Run/Stop
A feature that creates a break point in a programmed external dialing sequence. When you press a programmed key, the system dials the number up to the run/stop. When you press it again, the system dials the digits following the run/stop.

S

SAPS
See Station Auxiliary Power Supply.

Saved Number Redial
A feature that allows you to save the number of the external call you are on (providing you dialed the call) so that you can call it again later.

Schedules
Any of six different sets of services that can be applied to your system. See Services.

Selective line redirection
See Line Redirection.

Serial port
A port that sends and receives data one bit at a time. This port can be used to connect the Business Communications Manager server to a printer, external modem or mouse. Serial port connector has nine pins that are designated by software with the letters COM and a single digit, such as CO Communication Sever 1000.

Service modes
See Services.

Service profile identifier (SPID)
A number that identifies the services ordered with your ISDN BRI line. Each ISDN BRI line has two phone numbers. Each of these phone numbers has a SPID.

Services
A programming section that lets you assign which telephones ring, which restrictions apply and which call routing is used during any of six different schedules. There are three services: Ringing service, Restriction service and Routing service, all found in Services programming.
Set
A telephone.

Set Copy
A programming section that allows you to copy programmable settings from one telephone to another of the same type. Set Copy provides two options: duplicating System Data and User Data, or duplicating System Data only. Set Copy does not provide the same copy capability as copy, which is more selective of the settings that can be duplicated.

Set filter
See Restriction filter.

Set lock (telephone lock)
This feature allows you to limit the number of features that may be used at a telephone. Full set lock allows very few changes or features, Partial set lock allows some changes and features and No set lock allows any change to be made and any feature to be used. Set lock is assigned under Capabilities in Terminals and Sets programming.

Set relocation
See Automatic Telephone Relocation.

SIMM
Single In-line Memory Module. The Business Communication Manager server is equipped with one SIMM that provides 64 MB of SDRAM. The memory can be increased with the addition of more SIMMs.

Simple Network Management Protocol (SNMP)
SNMP is the protocol governing network management and the monitoring of network devices and their functions.

SNMP

Software Keycodes
All BCM Applications are loaded onto the system when it is shipped. Some of the BCM applications are standard and work immediately after the system is installed. Other applications are optional and must be enabled using software Keycodes which the customer must purchase in order to upgrade to those features.

Startup programming
When a BCM system is first installed and powered up, Startup programming must be performed before any programming can be done. Startup initializes the system programming to defaults.
Station
An individual telephone.

Station Auxiliary Power Supply (SAPS)
A device which provides power to a telephone that is connected more than 300 m (975 ft.) and less than 1200 m (3900 ft.) from the server, or to a CAP module.

Station Media Bay Module
A computer module which provides access to telephone lines. The 16-port Digital Station Media Bay Module (DSM 16) allows the connection of 16 digital telephone sets to the system. The 32-port Digital Station Media Bay Module (DSM 32) allows the connection of 32 digital telephone sets to the system. The Analog Station Media Bay Module (ASM 8) allows the connection of analog station sets to the system.

Station Set Test
A series of diagnostic tests for these components of a telephone: display, buttons, handset, speaker and power.

subnet mask
A value used to route packets on TCP/IP networks. When the IP layer has to deliver a packet through an interface, it uses the destination address contained in the packet, together with the subnet mask of the interface, to select an interface and the next hop in that subnet.

synchronous
A synchronous signal is sourced from the same timing reference. A synchronous signal causes the interval between successive bits, characters or events to remain constant or locked in to a specific clock frequency.

system data
An option in the Set Copy function. System data refers to the programmable system settings that apply to all telephones and lines.

System programming
A programming section that lets you assign and maintain certain settings on the BCM system.

System speed dial code
A two-digit code (01 to 70) that can be programmed to dial a telephone number up to 24 digits long. System speed dial codes are programmed for the entire system under the System Speed programming heading.

System Startup
See Startup programming.
T

T-1
Digital carrier system or line that carries data at 1.544 Mbps.

TAPI
See Telephony Application Program Interface.

Target lines
Lines used to answer incoming calls only. A target line routes a call according to digits it receives from an incoming trunk. They are referred to by line numbers in the same way as physical lines.

TCP/IP

TE
See Terminal equipment.

TEI
See Terminal Endpoint Identifier.

Telco features
A programming section that lets you specify the external telephone numbers that are dialed by the Message feature to retrieve voice messages, or to set up CLASS (CMS) services for lines and telephone. Telco features are accessed by an installer or a system coordinator.

Telephony Application Program Interface (TAPI)
The Telephony Application Program Interface (TAPI) is a standard program interface that lets you and your computer communicate over telephones or videophones to people or phone-connected resources elsewhere in the world.

Telnet
Telnet is a service that provides terminal-emulation capabilities for logging into the BCM unit from a remote location.

Terminal Endpoint Identifier (TEI)
A digit used to identify devices that are using an ISDN connection for D-channel packet service.
Terminal equipment (TE)
A generic term for devices that connect to an ISDN network. Examples of ISDN TE are ISDN telephones, computers equipped with ISDN cards and video terminals.

Terminals and Sets
A programming section that lets you assign and change settings that apply to the telephones and other devices connected to the Business Communications Manager server. Terminals and Sets programming is performed by an installer or a system coordinator.

time and date
A programming section that lets you manually change time or date.

Token-Ring
A token-ring is a network topology and data signaling scheme where a special data packet (called a token) is passed from one station to another along an electrical ring. A transmitting station takes possession of the token, transmits the data, then frees the token after the data has made a complete circuit of the electrical ring.

Tone dial telephone
A push-button telephone that emits DTMF tones.

TOS
See Type of Service.

Transfer
A feature that lets you redirect a call to another telephone in your BCM system, over a network or outside your system.

Transfer Callback
If a transferred call is not answered after a specific number of rings, the call will return to the telephone that made the transfer. The number of rings is assigned under Feature settings in System programming. Transfer Callback does not apply to calls transferred externally.

Transmission Control Protocol/Internet Protocol (TCP/IP)
A language governing communication among all computers on the Internet. TCP protocol checks packets of information for errors, submits requests for re-transmission in the event of errors and returns multiple packets of a message into the proper original sequence when the message reaches its destination.
IP dictates how packets are sent out over networks and has a packet addressing method that lets any computer on the Internet forward a packet to any other computer that is a step or more closer to the packet’s recipient.
Trunk
The physical connection between the BCM system and the outside world, using either the public telephone system or a private network.

Trunk Answer
A feature that allows users to answer a call on any line that has an active Ringing Service Mode, even if the line does not appear on their telephone. Trunk Answer is enabled in Services programming.

Trunk Media Bay Module
A computer module that provides access to telecommunications trunks. The Digital Trunk Media Bay Module (DTM) provides the connection between a standard digital PSTN T-1 or PRI line and the Enterprise Edge system. The Caller ID Trunk Media Bay Module (CTM) provides the ability to access four analog Caller ID PSTN lines. The Basic Rate Interface Media Bay Module (EE-BRIM S/T) connects up to four BRI S/T ISDN lines to the BCM system.

Type of Service (TOS)
The TOS field is located in the IP packet header and is used in DiffServ processing.

UDP
See User Datagram Protocol.

Unsupervised line
A line on which Disconnect Supervision is disabled. If an external caller hangs up, the system does not detect the disconnection and does not hang up its line. See Disconnect Supervision.

User Data
An option in the Set Copy feature. It refers to the personal settings that are unique to an individual telephone and are not programmed for the system. User Data is programmed at each telephone. These settings, for example, include Personal Speed Dial and the assignment of programmable memory buttons.

User Datagram Protocol (UDP)
A protocol that offers a limited amount of service when messages are exchanged between computers in a network that uses IP. UDP is an alternative to the Transmission Control Protocol (TCP) and, together with IP, is sometimes referred to as UDP/IP.
Like the Transmission Control Protocol, UDP uses IP to actually transfer a data unit (called a datagram) from one computer to another. Unlike TCP, however, UDP does not provide the service of dividing a message into packets (datagrams) and reassembling it at the other end. Specifically, UDP does not provide sequencing of the packets that the data arrives in. This means that the application program that uses UDP must be able to make sure that the entire message has arrived and is in the right order. Network applications that want to save processing time because they have very small data units to exchange (and therefore very little message reassembling to do) may prefer UDP to TCP. The Trivial File Transfer Protocol (TFTP) uses UDP instead of TCP.

**User Filter**
See Restriction filter.

**User Preferences**
A programming section that lets users assign autodialers, user speed dial codes, display contrast and other settings to a specific telephone or person. Users do not have to program these settings at the person’s telephone. User preferences are assigned in Terminals and Sets programming.

**User Speed Dial**
Users can program two-digit codes (71-94) to dial external telephone numbers. User Speed Dial numbers are programmed for each telephone and can be used only at the telephone on which they are programmed.

**Voice Call**
A feature users can deploy to make an announcement or begin a conversation through the speaker of another telephone in the system. The telephone a user calls will not ring. Instead, the person he or she calls will hear a beep followed by the caller’s voice. The user’s telephone will beep periodically to remind him or her that the microphone is open.

**Voice Call deny**
A feature that prevents your telephone from receiving Voice Calls.

**Voice message center**
If users have subscribed to Call Display services, they can receive visual Voice Message Waiting indication provided their telephone has a display. If users have Voice Message Waiting Indication, they can program the telephone numbers required to access up to five different Voice Message Centers. They can also program which of the five centers each specific line can access.
Voice over IP (VoIP)
A set of facilities for managing the delivery of voice information using the Internet Protocol (IP). In general, this means sending voice information in digital form in discrete packets rather than in the traditional circuit-committed protocols of the public switched telephone network (PSTN). VoIP and Internet telephony avoid the tolls charged by ordinary telephone service.
In addition to IP, VoIP uses real-time protocol (RTP) to help ensure that packets get delivered in a timely way. Quality of Service (QoS) is difficult to guarantee with public networks.
Indication, users can program the telephone numbers required to access up to five different Voice Message Centers. You can also program which of the five Centers is to be accessed by each specific line.
Using VoIP, an enterprise positions a VoIP device at a gateway. The gateway receives packetized voice transmissions from users within the company and then routes them to other parts of its intranet (local area or wide area network) or, using a T-1 or E-1 interface, sends them over the public switched telephone network.

W

Wait for Dial Tone
A feature that causes a sequence of numbers to pause until dial tone is present on the line before it continues to dial. The Wait for dial tone symbol (‡) uses two of the 24 spaces in a dialing sequence.

Web cache
A server or collection of servers that store copies of Internet content. The Web cache server can be either located on the LAN, where the clients it serves are also located, or it can be embedded within the enterprise WAN or at the client’s Internet Service Provider (ISP).

Web proxy
A server that acts on behalf of the requestor of pages from an HTTP server and the Internet.

Weighted Fair Queuing (WFQ)
A queuing method that allows prioritization of low volume traffic such as telnet. Interactive traffic receives higher priority than batch transfers. High bandwidth usage traffic, such as batch file transfer traffic and other high bandwidth use traffic are prioritized equally.

Wide Area Network (WAN)
A collection of computers or BCM servers connected or networked to each other over long-distances, typically using common carrier facilities.
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