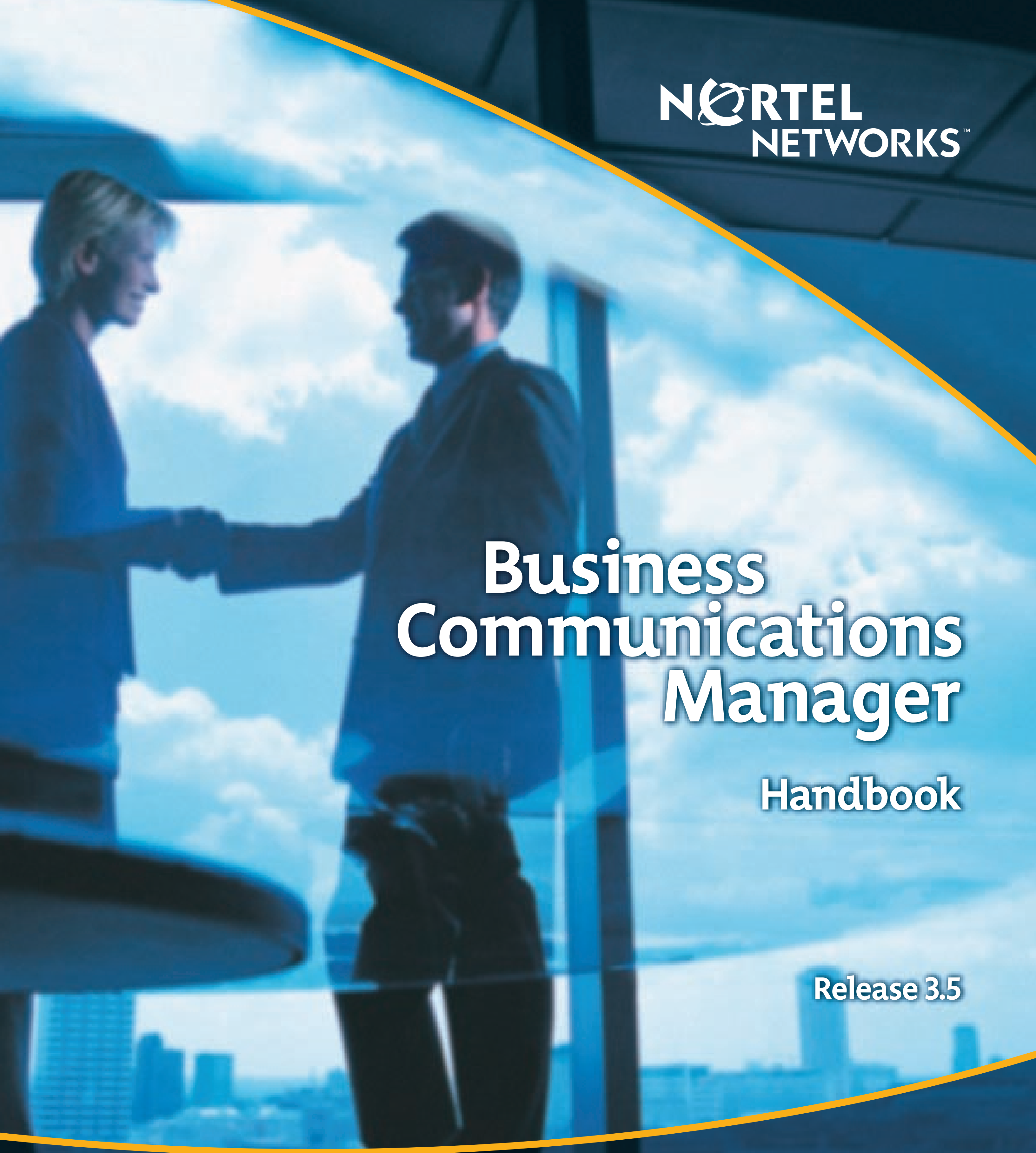

# Business Communications Manager

## Handbook

Release 3.5

**Note:** To view the Business Communications Manager Handbook with chapter bookmarks, go to the Adobe Acrobat Menu bar – under View, select “Bookmarks and Page.”

# Business Communications Manager

**Handbook is published by Nortel Networks. This handbook is intended as a reference guide for Sales Representatives, Telemarketers, and others who support the Nortel Networks Business Series product portfolio.**



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# Introduction

# Introduction

## Chapter Highlights

- IP Telephony – supports powerful new ebusiness applications that level the playing field with larger competitors, extend network services to remote workers, increase portability, simplify moves and changes and eliminate toll charges on site-to-site calls.
- Over 150 Powerful Applications – are preinstalled on the Business Communications Manager* (BCM).
- Call Center Applications– combine the reach of the Web with personalized agent interaction and customer support.
- Universal Internet Access – for all connected users and workstations, including access to corporate intranets, support for intrasite virtual private networks (VPNs) and remote connectivity for mobile or home users.
- Simplified Network Infrastructure – cuts costs by connecting IP phones over the local area network (LAN) wiring system, seamlessly extending features to multiple sites through IP connectivity and streamlining network management.
- Wireless eMobility* Solutions – break the chains that tie users to their workstations, giving call center agents hands-on access to samples and supporting wireless scanners for efficient inventory procedures.
- Browser-based Management – simplifies installations and provides an intuitive, wizards-based method of managing the network from any Web-enabled workstation.
- Independent Call Processing – means that if an operating system (OS) malfunctions, call processing remains unaffected, unlike many IP systems.

# Overview

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

The *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 is intended for reference purposes only. Please consult the latest product catalog or ordering source for detailed specifications and ordering information. The configurations and applications mentioned in this handbook may not be standard offerings with your company. Before talking to a customer about information in this book, please check for availability, pricing, current distribution and support policies.

# The Business Communications Manager

Nortel Networks 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 Networks enterprise communications solutions is the BCM.

Originally introduced to the market in early 2000, Nortel Networks 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 Networks 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.

# Business Communications Manager 3.0

Business Communications Manager 3.0 builds upon the capabilities delivered in previous releases to raise the bar as the standard for IP telephony solutions for the enterprise small site. Release 3.0 dramatically increases the value of the BCM as a branch office solution in small or large corporate networks with the addition of interactive voice response (IVR) support, which can drive cost reductions and revenues and multisite management to lower the overall cost of ownership. The increased TDM station capacity, when considered alone or in conjunction with the IP station capacity, allows the needs of sites with between 16-200 users per system to be met while preserving the choice of a TDM, IP or hybrid telephony strategy. Additionally, Release 3.0 incorporates a number of "Design for Serviceability" aspects that makes the BCM easier to install, support and service.

BCM 3.0 release introduces significant improvements to the platform's functionality, including:

- **Interactive Voice Response (IVR)** – is a self-service application designed to allow businesses to be accessible to their customers 24 hours a day, 365 days a year. Businesses can supply callers with access to a broad range of information simply by responding to a series of prompts via their touchtone phones. BCM 3.0 provides an integrated IVR runtime engine (RTE) that sits on top of Voice Applications Layer similar to Voice Mail or Call Center and interprets the IVR applications. IVR applications developed using Portal Solutions PeriProducer, or PeriPro Application Builder environment and PeriStudio Audio Development Tool, can be installed on the BCM in the field and can run on the BCM-IVR.
- **Network Configuration Manager (NCM) 2.0** – is a multisite management feature that provides centralized configuration and system management capabilities for a number of BCMs in a network. NCM delivers a management capability for multisite BCM customers and channel partners, enabling them to significantly reduce the total cost of ownership of their BCM systems.
- **Increased TDM station capacity** – means that the number of digital station sets has been increased. The number of station sets supported on the BCM is a critical capability for meeting the needs of larger sites. Through a more efficient use of existing resources, the number of digital sets supported has been increased to a theoretical maximum of 192. In practical solutions with a mixture of trunks and stations, BCM 3.0 supports from 128 to 160 digital sets, effectively doubling the previous digital station capacity.
- **Hardware platform improvements** – offer ease of installation, service and use. BCM 3.0 introduces a number of key improvements to the hardware platform. First, the number of Media Bay Modules supported on the base system has been increased to four, allowing more

sophisticated solutions at a lower cost to the end user. Second, the number of versions of the base platform has been reduced to two, a standard platform and a redundancy version, reducing stock requirements and order complexity. Third, the platform now incorporates "Design for Serviceability" features that make the BCM easier to install, set up and service. In addition, the capability for field installation of a WAN card has been introduced and the Media Services Card is field replaceable. Finally, BCM 3.0 offers a new, lower-cost hardware platform. This platform has two bays for Media Bay Modules and meets the needs of smaller, more price-sensitive sites.

- **BCM 3.0 Upgrade** – is the upgrade mechanism for BCM; it has been redesigned for BCM 3.0 to reduce the time it takes to upgrade previous releases. The upgrade supports field upgrade from either BCM 2.5 or 2.5 FP1 releases to Release 3.0.
- **Other features** – add value to Release 3.0. Telephony routing enhancements allow routing on 12 digits dialed and multiple level (≥2) automatic route selection. Call Detail Recording (CDR) has been enhanced to allow pull transfer of CDR data. Silent Monitor for Call Center, serving more advanced call center needs and Silent Monitor of Hunt Groups for less formal call centers, provide ability for a supervisor to silently monitor calls. An administrative edition of Desktop Assistant Pro is a downloadable client that allows a system administrator to configure any set on the system remotely. A client diagnostic tool for the i2050 Software Phone provides diagnostic information. Improvements to Unified Manager include a redesigned login page, a new network update wizard with network-loaded templates and button programming for the Add User wizard.

# The Business Case for Business Communications Manager

## A Convergence System

Convergence is best discussed by looking at three distinct dimensions of the IT environment:

- Infrastructure
- Management
- Applications

### At the Infrastructure Level

Since IT managers are increasingly measured on time-to-market for new applications, in addition to more traditional metrics tied to budget and trouble resolution times, it is critical for businesses to have a multiservice corporate networking infrastructure that is application ready. With the growing diversity of applications, in terms of both their performance needs and business criticality, the networking infrastructure needs to deliver rock-solid reliability.

Convergence is the means to the end. In this case, convergence means achieving the highest levels of price/performance and the highest levels of affordable network reliability. This end should be addressed holistically, at the switch and network level, with proactive network management supporting it.

### At the Management Level

People costs make up a significant portion of a business' IT budget – as much as 40 percent in many cases. Converged voice/data IT organizations can better manage the transition to unified applications and infrastructures, which can go a long way towards justifying cross-training of telephony and data resources, with their strengths in understanding the end user and network, respectively. Operational procedures have to reflect the growing business criticality of certain applications. Comprehensive management systems with uniform graphical user interfaces (GUIs) incorporating network, policy and service management not only improve overall network performance but also permit better use of scarce human resources.

### At the Applications Level

The growing popularity of the Internet and the standardization of Internet Protocol (IP) are inspiring companies to combine all communications, allowing them to simplify voice and data architecture and ultimately save money.

At the application level, it is widely accepted that new applications, whatever their form, will be developed on IP and that existing applications should be migrated to IP as best suits a business' specific needs. These applications need to be blended seamlessly with unified messaging supporting voice, email and fax communications; however, the real-time nature of these applications introduces some new infrastructure demands for time-critical performance and reliability.

IP telephony means lower costs through network simplification, higher employee productivity, better customer service, more revenue and greater profitability. Nortel Networks BCM and its proven Voice over Internet Protocol (VoIP) technology can boost business performance and accelerate business success today.

The bottom line: BCM merges voice and data services in a unified system that meets business demands at the infrastructure, management and the application level – delivering business-critical availability, application-optimized performance and lower life-cycle costs, including operations, capital per bit of capacity and bandwidth costs.

## An Internet Telephony Solution

To forward-thinking CEOs, convergence suggests new ways to accomplish these established goals:

- Set new standards in customer loyalty and satisfaction
- Rapidly deliver new products to existing and emerging markets
- Retain and attract skilled resources
- Enhance employee satisfaction and productivity
- Make efficient use of IT resources.

To attain these goals, CEOs have emphasized investments in supplier and customer integration. Such investments lead to integrated business systems, which constitute the kernel of an ebusiness model.

While CEOs may define business needs, satisfying these needs falls to another category of executive: the chief information officer (CIO). And, in fulfilling their assigned role, CIOs rely on information technology. This progression has been pronounced in certain industries, such as finance. Now, however, companies of all types are building their businesses around their networks and networked application environments. CIOs recognize that the emerging ebusiness environment is driving the need for quicker time-to-market for new applications, improved price/performance, increased reliability and scalability and management simplification.

With the BCM IP telephony offerings, customers can experience the benefits of IP while retaining the applications, reliability, features and functionality they have come to expect from their traditional voice networks. IP telephony is now a key factor in differentiating business operations and customer service strategies, offering companies a key competitive edge in a global economy where the nearest competitor is now only a click away.

BCM delivers a business-building suite of advanced applications to enable the delivery of new services and dramatically increase revenue opportunities. For businesses and organizations of all sizes, this includes multimedia contact centers and such high-powered business productivity tools as unified messaging and management, computer telephony integration (CTI) and mobility.

Customers can also save time and money through simplification because the BCM approach to IP telephony literally transforms multiple networks into a single, multiservice network, while driving simplicity to the desktop. Importantly, it protects a customer's investment in their current network. No forklift is required! The result is a powerful network that is reliable, rich in features and low in latency that offers economical and efficient solutions.

# Target Markets

Nortel Networks has identified the following three primary customer profiles as key target markets:

- Large Enterprises with branch locations
- Medium-sized, multisite businesses
- Standalone, single-site customers (up to 200 people).

## Large Enterprises with Branch Locations

Companies deploying a number of similar configurations in their small-site networks improve network efficiencies with the BCM. These businesses can use the BCM to:

- Standardize all their company's communications needs with a single product
- Network multiple locations that use their existing wide area network technology
- Simplify their business processes by using a single vendor
- Manage their telephony and data computing with remote management software tools.

Examples of large enterprises with branch locations include:

- Financial institutions
- Insurance agencies

- Retail chains
- Discount stores.

### Medium-sized, Multisite Businesses

The BCM provides the integrated voice and data solutions that larger retail outlets, such as grocery, general retailing and home improvements, may be considering. Furthermore, with its high level of reliability and ease of use, the BCM accomplishes this integration while protecting the customer’s investment for the future.

### Standalone, Single-site Customers

Companies seeking the competitive advantage of advanced customer service, and ecommerce options, appreciate the robust and future-proof performance BCM offers with its sophisticated voice telephony and Internet access.

Any company with 16 to 200 users in a single location can use this product to optimize their telecommunications and data computing activities.

Examples of medium-sized and small business customers include:

- Law offices
- Real estate firms
- Travel agencies
- Schools
- Small manufacturing companies.

## Vertical Markets

BCM offers a best-in-class, complete, converged voice/data solution to enterprise small sites across all industries. There are, however, key features and applications that make BCM an attractive solution for the following specific industries:

- **Education** – The built-in capability of BCM is very appealing to the K-12 market with the drive to “wire” classrooms for Internet access. And, as they wire for Ethernet, they can use the IP station support of BCM to provide voice connectivity to the classroom as well. The importance of a telephone set in every classroom has increased dramatically.

- **Retail** – In addition to IP telephony, the powerful call center capabilities available with BCM, including the Multimedia Call Center and *ip*View Softboard, can help retail customers deliver the top-flight customer service that leads to success. The IVR capability, available with Release 3.0, enables users in environments, such as pharmacies, to implement applications that help to offload repetitive tasks.
- **Government** – In this market, BCM reduces costs by creating a converged data/telephony network architecture. Advanced telephony features such as Call Transfer between branches, conference calling, call centers and wireless telephones streamline internal operations and increase efficiency.
- **Finance –** BCM provides financial institutions with the capabilities to evolve their branch telephony communications to IP telephony, while offering the benefits of lower wide area costs, operational simplification of moves and changes and security.
- **Professional services** – Businesses in this market, particularly those with branch offices, can realize cost savings by using an IP network to support telephony between sites. Further, in an environment where a Meridian* 1 PBX is already installed at the central site, centralized voicemail and four-digit dialing can be extended to all branch locations across the wide area network (WAN).
- **Restaurants** – VoIP and call center solutions offered by BCM allow businesses in this market to save on communications costs. Linking phone systems and the Internet means that a call to any restaurant in the chain moves as IP packets over a data network to the call center at the main location. Restaurants can achieve more uniform and consistent levels of customer service across all their branches.

## Business Communications Manager Key Benefits

As businesses look to balance their business goals and objectives with the evolution of telecommunications and computing, they are finding that convergence products must meet the following key requirements:

- Reliability
- Availability
- Streamlined costs

- Increased revenues, expanded customer base and improved customer service
- Simplified administration
- Future-proofed networks.

## Reliability

Customers are searching for a voice/data solution that provides the ultimate in reliability. The heartbeat of any business is its ability to send and receive information (through phone calls, data, images) to and from customers and suppliers. Dial tone is not an option and companies need to be able to rely on their network to transmit the required information as quickly and reliably as possible. Our customers are not in the business of managing technology. They require highly reliable solutions that allow them to focus on their core business.

The BCM Media Services Card (MSC) provides reliable call processing and media processing of voice channels and contains media services processor expansion cards that provide digital signal processing (DSP) resource control for applications. MSC call processing operates independently of the BCM platform's Microsoft NT Embedded
v. 4.0 operating system (OS). If the OS malfunctions, call processing remains unaffected.

## Availability

With the expanding use of the Internet to connect businesses together with their customers and other businesses, office hours are increasing to 24 hours a day, 7 days a week, 365 days a year for more and more companies. Converged systems not only have to operate reliably – they have to be available reliably to match these hours. Business applications and voice/data solutions allow unattended systems to carry on a company's business when all staff members at that location have gone home for the day.

Built in with the design of the BCM are features that increase availability. The DSP resources of the Media Services Card mentioned above, along with the system's central processing unit (CPU), provide redundant processing capability. A number of strategies can be used with the BCM to increase system availability even further. The Redundancy Feature option provides dual, hot swappable power supplies along with dual chassis cooling fans and data protection with RAID hard disk drive mirroring. External uninterruptible power supplies connected to the BCM can be used to handle power interruptions. And, the hardware platform's unique Design for Serviceability features minimize downtime for servicing of the BCM.

## Streamlined Costs

The ultimate goal of any business is to optimize shareholder value. An obvious and necessary first step towards reaching that goal is streamlining costs while boosting productivity. Our customers are looking for solutions that allow them to reduce their overall cost of ownership (up-front equipment costs, network service costs and maintenance and management costs) and increase employee productivity.

The BCM integrates voice and data components (CPE and DTE) into a single platform that can reduce costs by replacing individual components such as voicemail systems, automated attendant capabilities, computer telephony and remote access servers, multiplexer or CSU/DSU, router, modem and DHCP server. This next-generation platform integrates PBX/KSU, IP router, VoIP Gateway and IP services as well as voice applications and management software and servers, to provide a complete, unified, value-added solution. These powerful services and business applications multiply the productivity of employees and help to manage human resources costs for the company.

## Increased Revenues: Expanded Customer Base and Improved Customer Service

The second step in maximizing shareholder value is increasing revenue. Businesses are looking for solutions that will allow them to take advantage of market opportunities, compete more effectively and drive revenues. Expanding market reach and increasing the number of customers works directly toward increasing revenues. In today’s highly competitive and rapidly changing marketplace, most businesses understand that customers have choices and demand superior customer service. Businesses are seeking solutions that enable them to deliver outstanding customer service.

The BCM has more than 150 powerful software applications preinstalled. Some applications work immediately after the system is installed, while others can be enabled and downloaded locally or remotely for implementation as business requirements evolve.

These applications include:

- **Multimedia Call Center** – permits businesses to tap into the power of the Internet to expand into and reach new customers and to increase company awareness and revenue.
- **Voice Messaging** – offers a choice of Norstar* or CallPilot* Voice Mail interface that allows for at least 200 hours of message storage and up to 1000 voice mailboxes. Voice Messaging is available as a try-and-buy option, which can be activated for 60 days.
- **Message networking –** links CallPilot with other voicemail systems and allows the exchange of voice messages between users at different sites on a network connected by TCP/IP or MCDN networking.
- **Unified Messaging** – allows users to manage voice, fax and email messages directly from their multimedia-equipped PC or laptop.
- **Call Center, Professional Call Center and Call Center Reporting** – offer dynamic call handling and reporting applications that have been steadily enhanced with each new release. For instance, the number of queues, or skillsets, has been increased to 50 in Professional Call Center and improvements in agent display information have been made to include dialed number identification service (DNIS), allowing the person in the queue to see the number of an incoming call.
- **Custom Call Routing (CCR**) – ensures that callers reach the right department or person on the first try.
- **Attendant Console –** allows telephone attendants to monitor phone calls from their computer screen and answer and route them with a simple point and click of a mouse. Attendant Console runs on industry-standard Windows® 95/98/2000/NT or XP PCs and operates in a multitasking environment that lets attendants use their PCs for other work when not actively handling calls. Attendant Console is available as a try-and-buy option (one seat), which can be activated for 60 days.
- **Fax Messaging** – allows the user to receive, send and forward faxes in the same fashion as voice messages. Fax Overflow prevents customers from missing faxes by sending overflow faxes to a Fax Overflow mailbox, which stores the faxes until the fax machine is able to print them. Fax on Demand allows a user to retrieve documents stored in special mailboxes. Fax Suite provides Fax Messaging, Fax Overflow and Fax On Demand as a bundle. Fax Suite is available as a try-and-buy option that can be activated for 60 days.

- **LAN CTE** – allows customers to use the system as a TAPI Server. This capability means that any TAPI-compliant application running on a Client PC can control telephones on the BCM system via TAPI. LAN CTE Server is available as a try-and-buy option that can be activated for 60 days.
- **Personal Call Manager (PCM)** – is the award-winning telephony application available on BCM. It is designed for users on the Windows 95/98/2000/NT/XP operating system and brings much of the feature-rich telephony user interface to the desktop computer.
- **eMobility** – provides 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. The roaming feature allows a portable user to make and receive calls and access business features anywhere within a coverage area.
- **Unified Manager** – is a software tool that comes standard on every system and is used to manage the BCM at a single site. Unified Manager is a Java-enabled Web browser that provides a series of windows and menus that allow the user to navigate through the different areas of the application and program the system.

Tailored to the business processes of the company using the BCM, these applications can work to increase revenues through allowing receipt of information that might not otherwise get through. For example, in a restaurant using the BCM, a well-executed messaging application can ensure reservations are received and confirmed while staff members are busy with other duties.

### Simplified Administration

There is a fundamental fear faced by some considering IP telephony that convergence will exponentially increase the complexity of their network and that the resulting administration will be unmanageable.

With Unified Manager, a Web-based management tool, the BCM can be configured and managed from any PC on the network. Administration is further eased by the strategic use of Meridian Customer Defined Networking (MCDN) and Q.SIG (Europe) voice networking to support networking between BCM nodes and a Meridian 1 hub system, for centralized PSTN trunking, a coordinated dialing plan and name and number ID delivery between network locations. Also, the use of IP telephones greatly reduces the effort and cost previously associated with user moves, adds and changes.

BCM is compatible with Meridian 1 Internet Telephony Gateway and can share the Meridian Mail or CallPilot* Voice Mail of a Meridian 1. A consistent interface is provided for all voicemail users and the system uses the Unified Messaging client from CallPilot, which has been enhanced to support other popular third-party email packages. This adds additional strength to the overall simplification of an extremely powerful product.

Multiple BCM systems within a network can be efficiently managed using the BCM Network Configuration Manager (NCM) tool. This powerful server/client software running at a central operations center allows programming changes to be distributed to all BCM systems or to groups of BCM systems within the network. Programming changes can be scheduled for distribution at off-peak hours. Changes made to the BCM systems within a network include auto attendant and call center greetings, auto attendant schedules, custom call routes, IVR scripts and keycodes.

### Future-Proofed Networks

Potential customers are concerned about investment protection and their ability to strategically place themselves in a solid, viable position for the technologies of tomorrow. The BCM has evolved to support IP telephones and to integrate with leading-edge networking technologies such as virtual private networks (VPN). BCM is aligned with Nortel Networks Succession Strategy, guaranteeing a smooth migration path to pure IP solutions. When this migration strategy and vision towards IP is combined with the BCM architecture and its ability to support numerous PSTN connections points (BRI, PRI, T-1, E-1 or analog), it is not difficult to demonstrate that this product is intended for investment protection and long-term product and service viability.

## Data Capabilities

BCM provides a combination of routing services. IP routing is provided via static routing, RIP, RIP2 and open shortest path first (OSPF). BCM supports both basic and stateful packet filtering to provide security when passing packets. Five proxy firewall systems (HTTP, SOCKS, IPX-WINSOCK, NAT, PAT and DNS) are supported.

BCM also provides a number of services that enhance IP routing and IPX support via the local area network (LAN) segment. Network Address Translation(NAT)  (NAT) allows address allocation that provides for routing stability and network scalability. DHCP provides automatic assignment of IP addresses and BCM has a built-in DHCP Relay Agent to allow pass-through of DHCP traffic to and from LAN connected devices. DNS maps easy-to-remember names to IP addresses and Web caching allows multiple users to share information downloaded from the Internet.

## Platform Services

The BCM system can be accessed remotely via a WAN/Internet connection or a dialup connection. The dialup connection is established via a built-in V.90 modem, which creates an IP connection that allows the IP-based management tools to function.

BCM also provides a tool called BRU that allows administrators to back up and restore all customer configuration and user data.





# Hardware

# Hardware

## Chapter Highlights

- BCM 3.0 Hardware Platform Features – build on the features of previous releases and offer significant benefits; BCM is now easier to install, support and service, while offering significant benefits.
- Upgrade Support – allows users to upgrade their BCM base systems to 3.0, while preserving user programming and data. The upgrade kit includes installation instructions.
- Two Hardware Platforms – include BCM400 and BCM200. With four bays for Media Bay Modules, BCM400 simplifies configurations involving four Media Bay Modules, eliminating the need for the Expansion Cabinet. BCM200 with two bays for Media Bay Modules provides a lower cost solution for smaller, price-sensitive customers.
- Media Services Card (MSC) – is now a field replaceable unit (FRU) to allow replacement of this card in the field as opposed to sending the entire system in for repair.
- Removable Trays – improve access to the hardware for service and support. The lower tray (common to both 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 (BCM400 only) provides access to the hard drive and RAID card (if so equipped).
- Desktop Assistant Pro Administration Edition – is a client application available for system administrators that can be downloaded from the BCM and used on a desktop.
- i2050 Software Phone Diagnostic Tool – is a client application that can be downloaded from the BCM and used on the desktop in conjunction with an i2050 Software Phone client on that desktop.

# Overview

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

- Base units
- Media Bay Modules (MBM)
- Media Services Card (MSC)
- Two 10/100 Base T Ethernet Ports
- Business Series Terminals (BST)
- Legacy Terminals
- IP Terminals.

***Figure 2-1***

**BCM400**

**BCM200**

The chassis comes fully equipped with a Pentium® III 700-megahertz (BCM400) or a Pentium Celeron™ 850-megahertz (BCM200) 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 Embedded operating system.

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

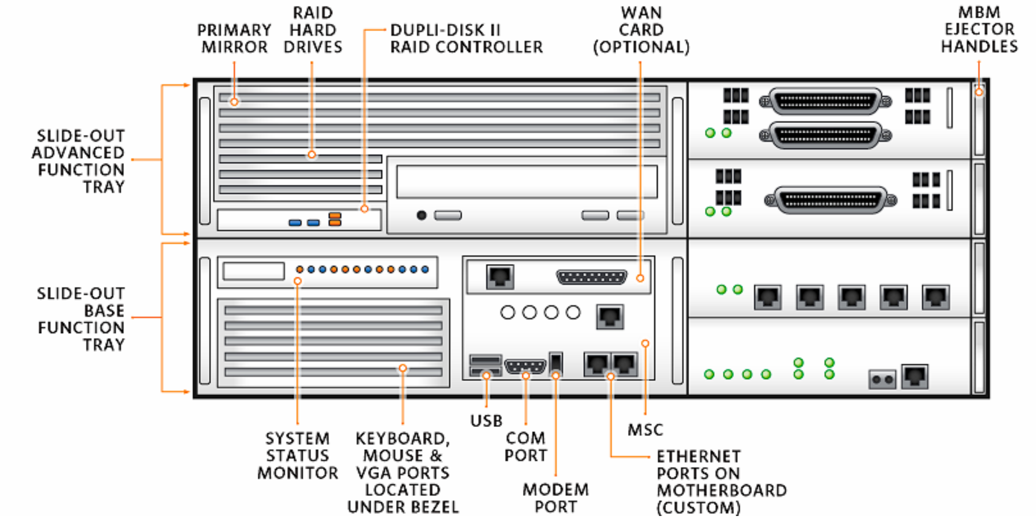
- Intel® Pentium® III 700 MHz (BCM400) or a Pentium Celeron 850MHz (BCM200), CPU
- 256 MB SDRAM
- 20 GB hard drive
- 2 10/100 BaseT Ethernet ports (on-board)
- 1 V.90 embedded modem (North America units only)
- 2 PCI slots (one used by the Media Services Card and one for adding a WAN interface card)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

The BCM base unit controls all tasks, including call processing, voice messaging and data routing. The base unit also contains telephony hardware and data networking hardware components.

Making and receiving calls is crucial to any business. The call processing capability of BCM has been designed to process calls even when the Windows NTE operating system is out of service.

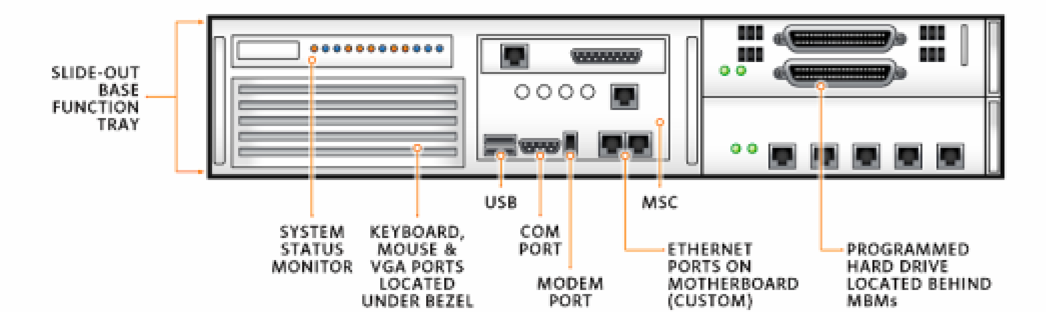
The BCM has been designed for flexibility and scalability, with support for pluggable and interchangeable Media Bay Modules. Release 3.0 now offers the BCM400 platform with four bays for Media Bay Modules. This addition will simplify configurations and eliminate the need for the Expansion Cabinet. It 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; however, 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.

***Figure 2-2***




BCM 3.0 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.

***Figure 2-3***



BCM 3.0 offers a number of new hardware platform features including:

- Upgrade support
- An additional Media Bay Module on BCM400
- A lower cost BCM200 platform for smaller sites
- Network interface changes
- Serviceability improvements.

## Emerging Trends

As the market migrates to IP telephony or a converged solution, service providers need to develop products that will differentiate themselves from the competition. By bundling voice and data services, a service provider can offer a “single stop” for a business’ communications needs and position itself to deliver a managed services offering either today or in the future. Installation and maintenance of various systems is also important, as is product reliability and build/design on industry standards for ease of operability.

## Benefits

The hardware platform changes to BCM 3.0 offer significant benefits. An important aspect of the hardware platform changes is total cost of ownership. A key change is the increase of Media Bay Modules for TDM telephony on the BCM400 base platform from three to four. While approximately 30% of BCMs to date have been installed with an Expansion Cabinet, two-thirds of the cabinets contained only one Media Bay Module. The new hardware platform will reduce the cost of an installation without sacrificing the functionality and at the same time simplify the installation and operation of the system.

The lower cost BCM200 version, with two bays for Media Bay Modules, meets the needs of price-sensitive smaller sites. For customers with 10 to 24 users per system who do not expect to grow past 32 users per system, this platform will have a very attractive price-to-value relationship. It delivers all the other features of BCM 3.0 with the same high level of reliability and serviceability.

Changes to the internal hardware will improve the reliability of the BCM while reducing the number of spares needed for support. For example, by making use of a computer motherboard with two on-board 10/100 Base T Ethernet ports and an on-board modem (North America only), these individual cards have been eliminated. The WAN card is now field installable, meaning that the base platform has been simplified and only those customers needing the WAN interface have it in the system.

Release 3.0 also incorporates a number of "Design for Serviceability" aspects that make the BCM easier to install, support and service. The improvements, including the hardware design with slide out front-accessible shelves, the field replaceable Media Services Card (MSC) and the provision of both a standard and a full redundancy version of the base system, have been designed to contribute to ROI. The upgrade from Release 2.5 or FP 1 to release 3.0 has been redesigned to reduce the time it takes to complete an upgrade while making the upgrade procedure easier and more robust. In addition, its design will facilitate easier upgrades to future releases of the BCM, making installed base support and migration an opportunity for ROI contribution.

# Business Communications Manager Components

## Connection Ports

### Serial Port

The BCM is equipped with one serial port that supports asynchronous serial data communication. 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 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 BCM is configured with ten LEDs mounted on the front panel. These LEDs are assigned the following functionality:

***Table 2-1***

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

# 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)
- Station Media Bay Modules
- Trunk Media Bay Modules
- Fiber Expansion Media Bay Module (BCM-FEM).

## Media Services Card

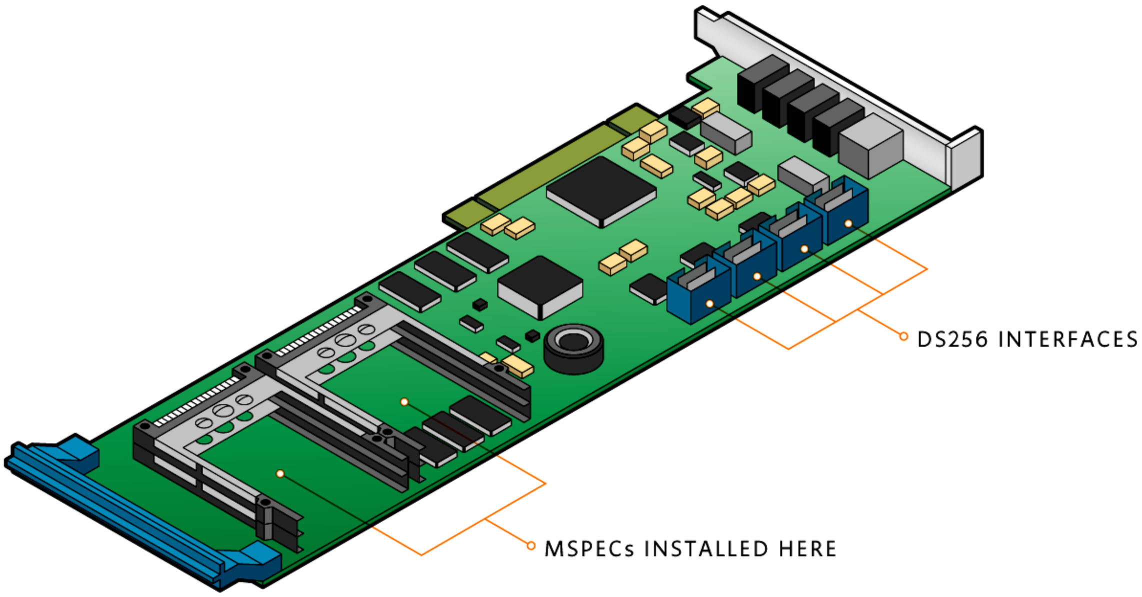
With BCM Release 3.0, the Media Services Card (MSC) is now a field replaceable unit (FRU) to allow businesses or users to replace this card in the field, as opposed to sending the entire system in for repair. The MSC performs call processing and media processing of the voice channels. It also provides the processing power for telephony functions independent of the Windows NTE 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 Voice Messaging will not function. The MSC also provides the processing power for the voice channels and compression utilities for Voice Mail, Call Center and IP telephony.

Installed on the MSC are two to four 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 voicemail, call center and IP telephony. All of these voice applications can share the DSP resources on one BCM platform.

The BCM 3.0 platform comes equipped with two PEC IIIs on the BCM400 and one PEC III on the BCM200. Depending on applications requirement, the BCM400 can be optionally equipped with four PEC IIIs and the BCM200 can be optionally equipped with two PEC IIIs.

***Figure 2-4***




The MSC also provides the following functions:

- Connection between the MSC and the pluggable Media Bay Modules

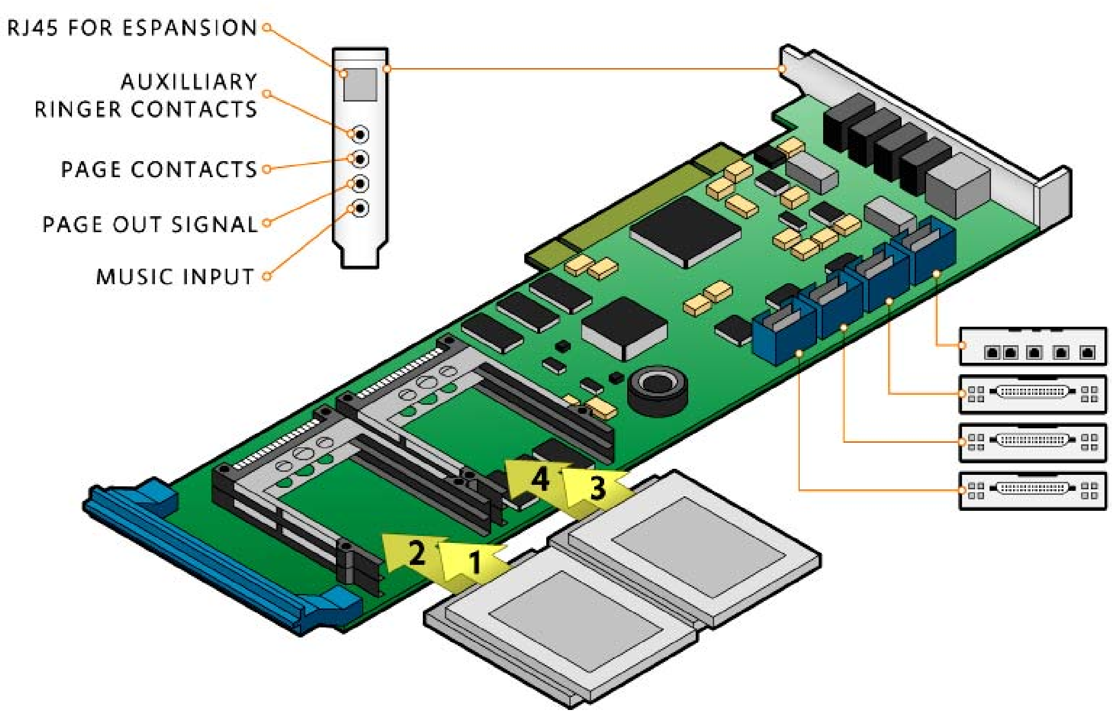
  The MSC has four DS256 interfaces used to connect to the Media Bay Modules installed in the BCM. The DS256 connectors are 2 x 5 pin headers located along the top edge of the MSC. A 10-conductor ribbon cable connects the MSC to the pluggable Media Bay Modules.

- 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 2-5***




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.

- 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 2-6***

### 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 2-7***

### Analog Station Media Bay Module (BCM-ASM 8)

The Analog Station Module (ASM) provides connectivity to eight analog stations. Analog support includes terminals, fax machines, answering machines and modems up to a 28.8 speed.

The BCM-ASM 8 has two LEDs on the faceplate labeled as follows:

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

***Figure 2-8***

## 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 future releases (CALA).

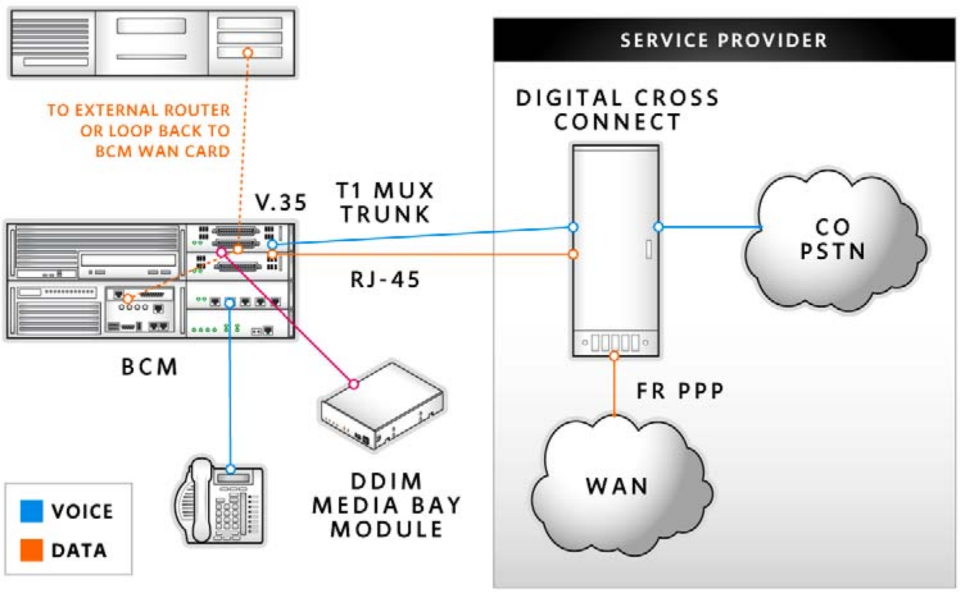
The front faceplate of the BCM-DTM has a number of LEDs that indicate power status and any ongoing tests and alarms that the module is undergoing. The faceplate also has an RJ-48C connector that connects the BCM-DTM to the service provider’s connection point and a set of loopback connectors used to run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

***Figure 2-9***

# Digital Drop and Insert Mux (DDIM) Module

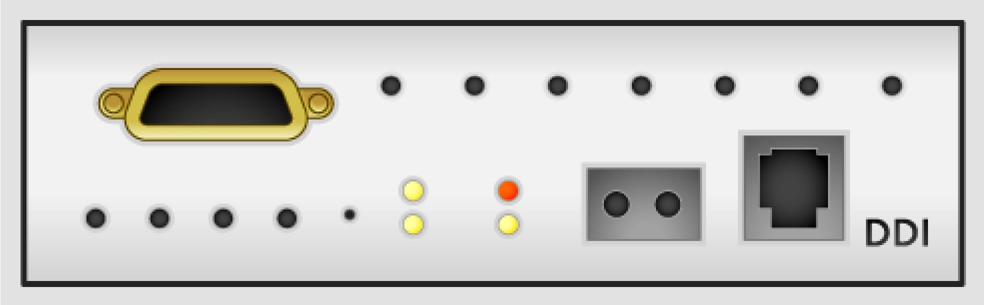
The Digital Drop and Insert Mux (DDIM) Module is a module that fits into the BCM and combines the functionality of the standard BCM Digital Trunk Module with a built- in Drop and Insert CSU. It accepts a T-1 with both voice and data from a service provider and splits off the channels carrying data and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

***Figure 2-10***




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 2-11*




The DDIM is supported in the main BCM 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 Networks 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 Networks 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.

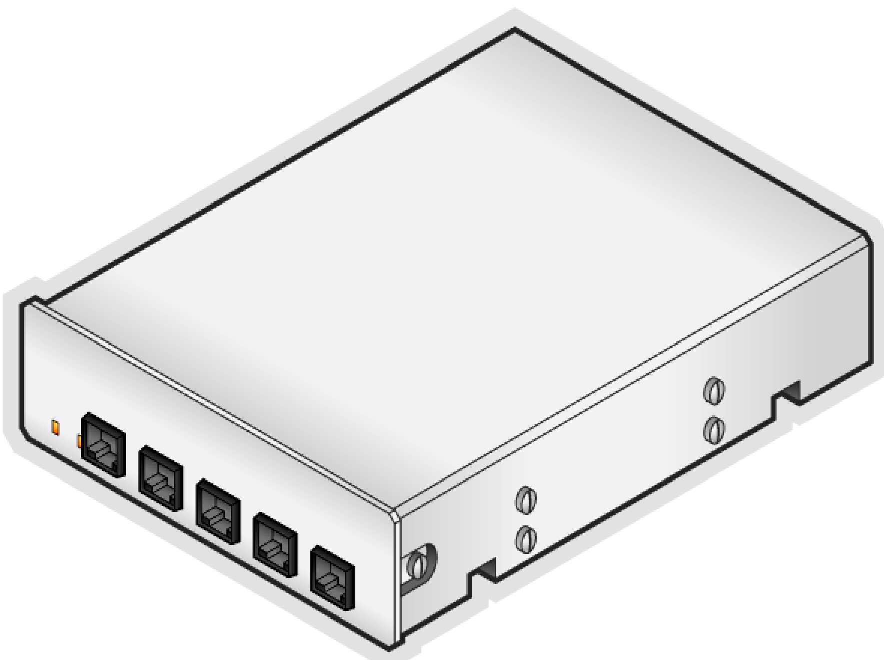
### Caller ID Trunk Media Bay Module 4 (BCM-CTM 4)

The BCM-CTM 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. The front faceplate of the BCM-CTM has four RJ-11 connectors that connect the CTM to the service provider's connection point.

The BCM-CTM faceplate also has two LEDs:

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

***Figure 2-12***



### Caller ID Trunk Media Bay Module 8 (BCM-CTM 8)

The BCM-CTM 8 connects up to eight analog CLID PSTN lines to the BCM system. Two auxiliary ports permit the connection of a 33.6+ Kbps modem, fax machine or single line analog telephone to line 1 or 5. When the auxiliary device is using line 1 or 5, 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. The front faceplate of the BCM-CTM has eight RJ-11 connectors that connect the CTM to the service provider's connection point.

The BCM-CTM faceplate also has two LEDs:

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

***Figure 2-13***

### 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 2-14***

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

Fiber Expansion Media Bay Modules connect Norstar Fiber Station and Trunk modules to the BCM 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 2-15*

**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 CTM8 and CTM.

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 2-16***

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 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:

- CTM 4/8
- DSM 16+/32+
- 4 x 16 (Combo)
- ASM

- 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.

***Figure 2-17***

**BCM Expansion Chassis**

# Data Networking Hardware Components

The data networking components connect the BCM 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.

## V.90 Modem

The V.90 embedded modem 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 3.0 has replaced the two 10/100 Base T Ethernet cards, available with 2.5, with two on-board 10/100 Base T Ethernet ports, eliminating the need for LAN card spares.

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
- LAN traffic smoothing via interrupt modulation and increased buffer size
- Point-to-Point Protocol over Ethernet (PPPOE)
- DiffServ queuing
- IPX support via LAN segment
- Supports IEEE 802.3 format
- Utilizes CSMA/CD for physical media access.

## WAN Interface Card

The WAN interface card is used to connect the BCM system to the wide area network. It is now available with BCM 3.0 as a field replaceable unit (FRU) instead of a factory-ordered system, reducing the number of hardware products to be stocked and simplifying the configuration choices. All customers will have two Ethernet ports available and will be able to add a WAN if desired. The WAN interface card has a T-1 interface port, a built-in CSU and a serial sync port. 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 opening the system. 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 (BCM400 FRU only).

## Upgrade Support for Installed Base

The BCM 3.0 upgrade kit provides a CD-ROM with which to upgrade BCM base systems in the field to Release 3.0. The upgrade allows users with BCM 2.5 and BCM 2.5 FP1 to upgrade their systems to release 3.0. 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 Voice Mail, Call Center (ACD), 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. For headsets, visit www.sencomm.com.
- 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 2-18***

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 2-19***

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 T7316

***Figure 2-20***

The T7316 telephone is part of the Business Series Terminals portfolio. It has 24 fully programmable buttons. Ten of the programmable memory buttons are supported by LCD indicators (for one-button access to a combination of lines, features and autodial numbers).

Six of the programmable memory buttons are supported by LCD indicators (for 1 button access to features and internal extensions).

Eight of the programmable memory buttons are used for one-button access to a combination of autodial numbers and features (these buttons are not supported by LCD indicators).

The T7316 supports the following features:

- External ringer interface
- Headset interface
- Handsfree interface
- Message waiting indicator/visual ringing lamp
- LCD with tilt Display – two-line by 16-character display is included on the T7316 telephone. This module is used to display call progress information, as well as to provide the legends for the three display (soft key) buttons on the module. 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 T7316 buttons are as follows:

- Twelve dial pad buttons

- Volume control (rocker type)
- Hold button
- Release button
- Feature button
- Three display buttons (soft keys)
- Ten programmable buttons with indicators (for lines, features and autodial numbers)
- Six programmable buttons without indicators (for features and internal extensions)
- Eight programmable buttons without indicators (for autodial numbers and features).

# 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 Networks does not support the connection of Headsets to the T7208 or T7316 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
- The existing M7324 uses a dual function Handsfree/Mute button, whereas the T7208 and T7316 sets have a separate Mute button
- 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.)

## M7324 Telephone

***Figure 2-21***

The M7324 Telephone has the following features:

- Two line, 16 characters each LCD display
- 24 Memory buttons with LCD indicators
- Three soft keys to activate visual display prompts on the LCD
- Headset jack
- Wall mountable
- Quick feature list
- Feature access button
- Release button
- Hold button
- Supports up to two Answering Position (CAP) modules.

### Description

The M7324 offers 14 programmable buttons with an LCD for one touch access to any combination of lines, features and autodial numbers, making it an excellent choice for a Central Answering Position (CAP).

Up to two CAP modules, also referred to as Key Lamp Modules, can be attached to an M7324 telephone – adding 48 buttons per module. This additional key count creates comprehensive coverage for numerous lines or to accommodate additional BCM features:

***Table 2-2***

| | |
|---|---|
| • Two Line, 16 Character each LCD display | • Feature access button |
| • 24 Memory buttons with LCD indicators (for one button access to a combination of lines, features and autodial numbers) | • Release button |
| • Three soft keys – to assist using the visual display prompts on the LCD | • Hold button |
| • Headset jack | • Supports up to two Answering Position (CAP) modules (adding 48 buttons per module) |
| • Wall Mountable | • |

## Nortel Networks Business Series Terminals

***Table 2-3***

| Feature List | T7100 | T7208 | T7316 | M7324 |
|---|---|---|---|---|
| LCD display | √ | √ | √ | √ |
| Integrated tilt display | √ | √ | √ | |
| Deployment on Business Communications Manager and Norstar (all releases) | √ | √ | √ | √ |
| Memory | | | | |
| • buttons with LCD indicator | 0 | 8 | 16 | 24 |
| • buttons without LCD indicators | 1 | 0 | 8 | 0 |
| # of line appearances | 0 | 8 | 10 | 24 |
| # of programmable autodial buttons | 1 | 8 | 24 | 24 |
| # of fixed buttons | 5 | 7 | 10 | 8 |
| # of soft key buttons | 0 | 0 | 3 | 3 |
| Handsfree | | √ | √ | √ |
| Visual ringing indicator | √ | √ | √ | √ |
| Audio control center (dedicated Headset and Mute buttons) | | √ | √ | √ |
| Volume bar | √ | √ | √ | √ |
| Call log | √ | √ | √ | √ |
| Intercom | √ | √ | √ | √ |
| Selective ringing tones / Discriminating ringing | √ | √ | √ | √ |
| Automatic set relocation | √ | √ | √ | √ |
| Multilingual capability | √ | √ | √ | √ |
| FWD / DND | √ | √ | √ | √ |

| Feature List | T7100 | T7208 | T7316 | M7324 |
|---|---|---|---|---|
| Wall mount capability | √ | √ | √ | √ |
| Support for Central Answering Position (CAP) Module | N/A | N/A | N/A | √ |

## 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 T7316 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 Release 3.0 introduces 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, T7316 only (does not label T7406, or M7XXX Series)
- Available in English, French and Spanish
- Supported on Windows 95/98/2000 and NT4.

Customers can purchase the application on CD (NTAB3320) or download it from 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 2-22***

**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 2-4***

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

## NACU

***Figure 2-23***


### 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.

### Target Audience

The NACU is the ideal teleconferencing solution for small and medium-sized conference rooms and for private offices where individuals frequently engage in teleconferences.

For best performance, persons speaking on the teleconference should be no more than six feet away from the NACU and the room housing the NACU should be no bigger than 10 x 13 feet.

## Central Answering Position (CAP)

***Figure 2-24***

### Description

The Central Answering Position (CAP) is a module connected to an M7324 telephone that provides 48 additional memory buttons, which can be used to show a busy or idle status for up to 48 more sets, or to program system features or autodial numbers. Up to two CAP modules can be added to any M7324 set. One Station Auxiliary Power Supply (SAPS) is required to power every two CAP modules.

A telephone with one or two CAP modules must be assigned enhanced CAP status in order to provide line appearances or access to central office lines.

Note that CAP positions can have one or two CAP modules attached, but what counts is the total number of positions as opposed to modules. BCM systems can support up to five enhanced CAP positions.

### Target Audience

Attendants and receptionists with M7324 sets responsible for call coverage for many lines will find the CAP very useful. Administrators, managers and others providing backup answering services will also improve their productivity using a CAP module, as will people requiring high feature usage or line access capabilities, such as telemarketing center managers.

## Station Auxiliary Power Supply (SAPS)

***Figure 2-25***

#### 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 M7324 set) at 1,000 feet. One SAPS is required to power every two CAP modules.

#### Target Audience

The Station Auxiliary Power Supply is ideal for BCM installations in large facilities, such as shopping centers, warehouses, airports and manufacturing floors.

# 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 Networks offers two superior desktop models, along with an innovative software-based solution that brings VoIP to a user's desktop laptop or PC. The three Nortel Networks Internet telephones include:

- **i2002 Internet Telephone –** 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.

- **i2004 Internet Telephone –** 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.
- **i2050 Software Phone** – 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 Networks Internet telephones provide support for a wide range of today's high-value ebusiness applications, including CallPilot Unified Messaging and Symposium* Call Center services.

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.

### i2050 Software Phone Diagnostic Tool

In addition to the three Internet telephones, BCM 3.0 introduces the i2050 Software Phone 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 i2050 Software Phone client on that desktop. It facilitates the quick resolution of any issues with the i2050 Software Phone, providing both IP and multimedia information.

# Desktop Solutions

## Nortel Networks i2002 and i2004 Internet Telephones

***Figure 2-26***

**i2002** **i2004**

The i2002 and i2004 Internet Telephones 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 i2002/i2004 Telephones 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.

In mission-critical environments, the Nortel Networks 24-Port Power Over LAN Hub functions as an in-line unit at the wiring closet to supply Nortel Networks desktop Internet telephones with power over the network cabling system. The i2002 and i2004 Internet telephones provide the following unique features:

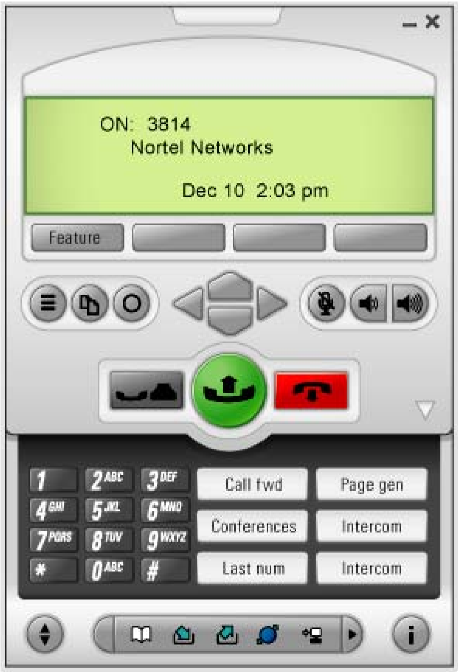
- High-fidelity full-duplex speakerphone facilitates group conference calls and delivers crystal-clear, handsfree communication
- A pixel-based LCD display provides a window into a full range of personal productivity tools

- 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 3.0 provides easier access to more features on the i2002 and i2004 Internet Telephones 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.

# Software Solutions

## Nortel Networks i2050 Software Phone

***Figure 2-27***




Ideal for mobile users, the Nortel Networks i2050 Software Phone is a software-based solution that loads directly onto your laptop or desktop PC. Once a Nortel Networks headset is connected to the USB port, the i2050 phone delivers virtually identical functionality as the Nortel Networks i2002 and i2004 desktop phones. The i2050 phone 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 Nortel Networks i2050 Internet Phone provides the following unique features:

- Macro functions transform lengthy operations into a single-digit action
- Nortel Networks 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.

## Internet Telephones Benefits

- Support connectivity to any Nortel Networks VoIP enabled platform, including BCM, IP-enabled Meridian* 1 and Meridian SL-100 PBX systems and Succession* Communication Servers
- 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 Networks Internet telephones are supported by multiple Nortel Networks communication systems, including BCM, IP-enabled Meridian 1 and Meridian SL-100 systems (with Internet Telephony Gateway cards installed) and Succession Communication Servers. This industry-leading platform interoperability facilitates growth and offers seamless migrations across customer premises and carrier-based solutions. All three 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 Nortel Networks i2002 Internet Telephones 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. The i2004 can be equipped with an optional 10/100 three-port switch that accomplishes the same thing. By 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 Nortel Networks i2002 and i2004 Internet Telephones can receive their power over the network cabling. The Nortel Networks 24-Port Power over LAN Hub 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 24-Port Power over LAN Hub. The unit fits into a standard 19-inch wiring closet rack and provides a cost-effective way to centralize power to the Internet telephones. This approach delivers carrier-grade reliability by enabling redundant power resources located at the wiring closets to provide emergency backup power to Nortel Networks Internet telephones located across the network.

### Dynamic Host Configuration Protocol (DHCP) Addressing

Easy to set up and configure, the Nortel Networks Internet telephones 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 Networks Business Policy Switch, 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 Networks is the industry leader in telephony solutions. All three of the Nortel Networks Internet telephones 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 CallPilot 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 Networks actively participates in defining standards-based solutions that support the broad deployment of VoIP across enterprise environments. The Nortel Networks i2002 and i2004 Internet Telephones 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 Nortel Networks Internet telephones behave like standards-compliant MGCP and H.323 devices, enabling Nortel Networks platforms to be distributed across the network while seamlessly interworking with similar standards-based gateways.

## IP Set Features and Programming

Release 3.0 introduces some new uses of the large LCD display of the IP sets. The display will now show a list of telephony features that users can highlight and then activate from the display. This capability means a user will not have to refer to the feature code list or memorize the feature codes as a means of activating features. BCM 3.0 will provide a default list of the most common features, which can be displayed on the i2004 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 the i2004, i2002 and i2050 sets.

BCM 3.0 allows the user to program additional buttons on the IP set. On the i2004 and i2050 sets, six additional buttons are programmable for a total of 12 programmable buttons. On the i2002 set, five extra buttons for a total of 9 buttons are programmable.

BCM 3.0 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.

# Telephony

## Chapter Highlights

- IP Telephony on BCM 3.0 – supports powerful new ebusiness applications that level the competitive playing field with larger competitors, extend network services to remote workers and eliminate toll charges on site-to-site voice calls.
- TDM Station Capacity – has been increased with BCM Release 3.0, which now supports up to a maximum of 192 digital sets.
- Analog Station Capacity – has been increased with BCM Release 3.0, meaning that the Analog Station Modules make more efficient use of DS-30 channel resources. In addition, Release 3.0 can support more analog sets.
- Destination Code Routing – has been enhanced to allow routing on up to 12 dialed digits as opposed to the previous maximum of 7 dialed digits on BCM 2.5.
- Call Detail Recording (CDR) – records and reports call activity and allows companies to record information about all incoming and outgoing calls.
- CDR Pull Mechanism – allows a central client machine, on its own schedule and up to its own capacity, to contact any number of BCMs and retrieve call data.

## Overview

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

- PBX/Key system functionality
- Fully integrated voicemail
- Automated Attendant with Custom Call Routing (CCR)
- Computer Telephony Integration (CTI).

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 only)
- 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 (Caller ID).

BCM Release 3.0 offers significant telephony enhancements, including:

- Increased TDM (digital) station capacity
- Telephony routing enhancements
- Call Detail Recording enhancements
- Increased analog station capacity.

## Emerging Trends

With IP telephony, all moves, adds and changes can be administered with the same tools used to control the network, saving time in hours and dramatically cutting down service costs. Moreover, with the dynamic allocation of IP addresses, personnel can simply plug in a phone at a new location and receive the full suite of functionality for which their phone had been originally configured. This type of savings is particularly important to dynamic organizations, such as engineering or consulting companies, where people are structured into project teams.

## Benefits

BCM provides the most complete telephony offering for the small site with hundreds of features businesses can use as they require them. With all the TDM telephony capabilities of Norstar, BCM allows Norstar partners to leverage their knowledge, skills and installed base in supporting customers who fall into these segments and can take advantage of the additional voice and data networking capabilities offered by BCM.

BCM 3.0 offers significant benefits, including increased station capacity to meet the needs of customers in all markets who have a higher number of users on their system. It is as applicable to the standalone and multisite small to medium-sized businesses as it is to the enterprise business with replicated, loosely-coupled or highly integrated branches. BCM now accommodates sites with 16-200 users per system. In addition, companies with a lower number of users per system, but which desire the room for growth as their business evolves, will be confident that the BCM is the right choice for them.

## 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 voicemail 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 server 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. With the introduction of BCM 3.0, the BCM will also allow 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 Release 3.0 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 3.0 allows the use of FTP and other TCP/IP based standard tools to set up and transfer the CDR files and maintains platform independence.

## 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 3-1 is an example of an account code list.

*Table 3-1*

| Account code | Description |
| --- | --- |
| 11127 | Patricia |
| 37 | Field Support |
| 239 | Joe |
| 45 | Modern Ways Limited |
| 100 | Long-distance |

## 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 and Benefits

This section describes BCM telephony features and some of the possible benefits customers can realize.

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

*Table 3-2*

| Features | Description | Benefits |
|---|---|---|
| Access Control to Link, LNR (Last Number Redial), SNR (Stored Number Redial) | System security enhancement that allows system administrator the option to remove access to Link, LNR and/or SNR on a set-by-set basis | Prevents toll fraud by forcing set to adhere to BCM call restrictions |
| Accidental Disconnect Protection | If the receiver is accidentally dropped back into the cradle when answering a call, it can be retrieved within one second | Prevents calls from being lost and thereby provides more professional call handling |
| Administration & Configuration Tree | | |
| Alternate Restrictions | Alters which calls can be made by changing dialing restrictions according to both time of day and day of week | Helps to control unauthorized calling |
| Analog Station Module Recognition | | |
| 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<br>Lets users answer calls at the monitoring set by pressing the active button | Users can monitor or answer all calls at another phone by simply pressing one button |
| Auto-Answer | Used with DID, DISA and E&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 | Improves customer service by offering more efficient incoming call handling |
| Auto Bump On/Off | 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<br>When "Off," the Call Log will not log new calls | Call management at the set level means better efficiency and customer service<br>Ensures that users answer their latest calls |
| Auto Dial (Internal and External) | Allows users to program internal or external numbers onto memory buttons for one-button dialing access | Saves time by providing direct access to another person in or outside the office with no need to remember the frequently dialed numbers |

| Features | Description | Benefits |
|---|---|---|
| Automatic Callback (AC)* | Automatically redials the last outgoing number dialed<br>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<br>Can be programmed as an external autodial for one button convenience | Improves productivity as the caller, while waiting, can work or continue to use the telephone<br>With BCM, the special ring is easy to hear and callers are quickly identified with call display |
| Automatic Daylight Savings time | 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.<br>Can be deactivated where not applicable | Eliminates the next business day confusion about what time it is and how to program the change |
| Automatic Line Selection | When answering incoming ringing calls, BCM automatically selects the longest ringing line first<br>Ringing incoming calls are automatically connected by lifting the receiver, pressing Handsfree, or using Call Queuing | Users do not need to know which line is ringing or which call has been ringing the longest |
| Automatic Number Identification (ANI) | Delivers the calling line number (T-1 Specific) | Improves customer service when used to pull customer info from a database before speaking with customer or for call screening when users are waiting for an important call |
| Automatic Recall (AR) * | Works the same as Automatic Callback (AC)<br>Applies to the last incoming call received | Saves time and money during employee moves<br>Employees can keep the same extension number |
| Automatic Route Selection (ARS) | Automatically selects the preprogrammed long-distance carrier based on the dialed digits, time of day and day of week | Lowers costs by insuring that the cheapest available long-distance routes are being used |
| Automatic Set Relocation | Sets moved to a different location retain all custom programming | Saves time and money during employee moves |

| Features | Description | Benefits |
| --- | --- | --- |
| Auxiliary Ringing | A set's headset jack can send ringing tones via an amplifier to an external loud ringer connected to the set | Users can hear a ringing set in a noisy environment. |
| Background Music | Users can listen to music (customer supplied) through the set's speaker when the set is idle | If supplied, it delivers music to a user's set to enhance their work environment |
| Busy Lamp Indication | Indicates that a user's set is busy | Users can see call status |
| Button (Key) Inquiry | Allows users to check the programming on memory buttons | Ensures current programming matches the button labeling |
| Call Display when Busy | Call display shows the name of a calling party while the user is on a call | Users can decide whether or not to interrupt their current call and take the incoming call<br>Ensures users don't miss important calls |
| Call Duration timer | Temporarily displays the length of the last or current call so a user can record it. | Users can track time spent on calls; useful for account billing |
| Call Forward All Calls | Sends all calls to another set | Improves customer service by ensuring all calls are answered |
| Call Forward No Answer | Transfers an unanswered call to another DN after a preset number of rings | Ensures a user's phone is answered if the user is unable to answer a call or has forgotten to activate Call Forward |
| Call Forward On Busy | If a set is busy, sends calls immediately to another DN | When a set is busy, the user is not disturbed and incoming calls are routed promptly |
| Call Forward Override | When a set is on Call Forward, the "Forward To" set can still call the "Forwarded" set to relay important messages | Improves office communications; users can inform others of important messages or ask for forwarding to be cancelled |
| Call Forward – Selective | Allows users to transfer a call to the Prime set by pressing the "Do Not Disturb" button when a central office line is ringing | Improves time management for the user, who can view the caller's name and then choose to accept or reject it |

| Features | Description | Benefits |
|---|---|---|
| Call Information | Displays information about incoming calls<br>For external calls, CMS is required and it displays the caller's name, telephone number and the line name<br>For internal calls, it displays the name and the internal number | Users can obtain information about ringing, answering, or held calls |
| Call Log | Allows users to enter Call Log to view stored information including:<br>Caller's name and/or number (if delivered from the central office)<br>Date and time<br>Answered Call Indication<br>Repeat Call Counter | Increases business opportunities and improves customer service by capturing caller's number and/or name even if the call goes unanswered<br>Improves productivity by allowing a user to redial the caller's phone number |
| Call Log – Optional Password | Allows users to enable password protection of their call log<br>*Caller Log: 600* | Provides the user with security for their call log |
| Calling Name & Number Display | Allows users to view the name and number of the incoming caller before answering and during the call<br>The calling number is also stored in the Call Log (Requires CLASS/CMS, ISDN-BRI or ISDN-PRI) | Improves customer service as users know who is calling<br>In conjunction with Call Log, knowing who called increases business opportunities<br>The caller's number can be passed to an associated PC application to improve customer service; for example, by triggering automatic customer profile retrieval |
| Calling Name & Number Display Blocking | Allows users to prevent delivery of calling name and/or number when placing a call<br>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)<br>On incoming calls, presents users with "Private Name" and/or "Private Number" when receiving a "blocked" call | Maintains privacy and security for outgoing calls<br>Identifies callers who have blocked their number and/or name for inbound calls |

| Features | Description | Benefits |
|---|---|---|
| Call Park – Linear/Round Robin | Allows system administrators to choose linear or round robin call park codes<br>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<br>Round robin call park codes are assigned sequentially until the maximum number of codes is reached before starting again at the first code<br>*Number of Call Park Codes: 25* | Provides system configuration flexibility and customization to customer needs |
| Call Park (with Callback) | Automatically places an active call on hold and assigns it another code so it can be retrieved from another set | More convenient and efficient, parked calls can be retrieved from any telephone in the system |
| Call Pickup Directed | Allows users to answer a ringing call at any other set by dialing the ringing set's intercom number | Simple and convenient to answer calls ringing at another set |
| Call Pickup Group | Allows users to answer any call ringing at another set within the pickup group | Within a defined group can provide more efficient call coverage for incoming calls ringing at another set within the group |
| Call Pickup Trunk Answer from any station | Allows users to answer a ringing external call at any other set | Improves after-hours communications, as calls can be heard and answered from any station |
| Call Queuing | Answers the next available call, but gives priority to the longest waiting external call | Improves customer service, because calls are answered promptly and efficiently |
| Callback | Automatically returns unanswered parked or transferred calls to the originating set after a preset number of seconds | Improves customer service as all calls will be answered |

| Features | Description | Benefits |
| --- | --- | --- |
| 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 | Allows more calls than a set has lines to wait at a set; alerting tones or illuminated LCD notify the user of a waiting call |
| Central Answering Position (CAP) | Also known as a Key Lamp Module, a CAP module connected to a M7324 set may be assigned enhanced CAP status<br>Allows lines assigned to this DN to be moved to the CAP module | Offers efficient system control and management |
| 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 | Offers efficient system control and management |
| Class of Service (COS) Password | 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<br>Is used when DISA access is controlled with passwords<br>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<br>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<br>*Number of COS Passwords: 100* | Helps to maintain the security of the system by limiting access to authorized users and limiting those users to the features they require<br>Minimizes unauthorized system access |
| Compression of Feature Codes | Reach-through codes (run/stop, programmed release, pause) can be compressed to use less digit space in Autodial or Speed Dial programming modes | Increases for longer numbers, features or access codes in Autodial or Speed Dial |

| Features | Description | Benefits |
|---|---|---|
| Conference (Three-Person) | Creates a three-person call with two other internal or external parties<br>Is easily set up with LCD prompts; Automatic Hold protects the first call from being accidentally cut off<br>Flash hook (switch hook flash) during a conference call has been added in BCM 3.0<br>TAPI application dialing and flash hook during a conference has been added in BCM 3.0 | Allows users to hold a meeting over the phone, reducing meeting expenses |
| Consultation Hold | 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 | Users can obtain additional information without having to terminate and later re-establish the call, thereby saving time and effort |
| Coordinated Dialing Plan | Allows administrator to program calls to route over a network based on destination codes | Network calls can be programmed in a manner consistent with local calls |
| Customer Originated Trace * | 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<br>Note: The user does not receive the number of the caller. Security procedures will vary with different telephone companies | Helps eliminate malicious and nuisance calls |
| Delayed Ring Transfer | Automatically transfers incoming calls to a "prime set" after a preset number of rings | Improves customer service as all calls are answered |
| Dial "0" Station (Dial "X") | Designated receptionist set that can be reached from any other set in the system by pressing the intercom key followed by the designated digit | A receptionist or control person in the office can be reached quickly and easily |
| Dial Intercom | Allows users to quickly call co-workers internally by pressing the Intercom button and dialing the intercom number | Provides easy access to coworkers, while keeping outside lines free |
| Dialing Filters | A maximum of 100 line, set or line/set filters provide virtually unlimited flexibility in programming dialing restrictions and exceptions | Access to lines can be well controlled |

| Features | Description | Benefits |
|---|---|---|
| Dialing Modes | A user can select from the following dialing mode options:<br>Standard: Lets a user choose a line and dial a call using either the receiver or handsfree<br>Automatic: Pressing a dial pad button will automatically select the set's prime line, thus saving time | Offers a more efficient dialing process<br>A convenient, handsfree dialing option |
| Dialing Mode: Pre-Dial | A person enters, checks and edits a number before selecting a line | Eliminates dialing mistakes |
| Dial Mode for Lines | Temporarily changes set from pulse to tone mode to signal external systems and devices | Improves external communications by letting users activate/access equipment such as voice recording devices |
| DID Template | At System Start, template choices include a DID Template which automatically assigns target lines and received numbers as the set DN<br>When a system is expanded these assignments are preserved (Typically programmed by the installer) | Businesses with a network can easily integrate a new system or expand existing systems by using this template |
| Direct Dial – Flexible Digits | A systemwide digit used to call a direct dial set can be any digit from 0-9 | Enhances internal dialing and communications |
| Direct Dial – Multiple Attendants | 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 | Improves customer service |

| Features | Description | Benefits |
|---|---|---|
| Direct Inward System Access (DISA) | 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 | Provides added security by controlling system access. Remote access can be assigned to specific users who need to use the system's network connections or other services<br>Users can access the system from remote locations using a Class of Service password. Used as a security feature to control remote access into the BCM using central office lines; configured with Loop, DID and E&M Trunks |
| Disconnect Supervision | After an external call disconnects, drops the line immediately | Prevents the system from identifying a line as on hold or busy, thus denying access, when in fact it is really idle (Note: the local telephone company must also support Disconnect Supervision) |
| Discriminating Ringing at Set | Different rings for internal and external calls allow users to easily distinguish between call types | Provides improved call handling and customer service |
| Distinctive Ringing/Call Waiting* | A user hears special ringing or call waiting tones if the caller is included in a user-specified list of numbers | Improves customer service by providing an indication of a special customer's call |
| Do Not Disturb (DND) | 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 | Users can work uninterrupted when necessary |
| Do Not Disturb (On Busy) | Internal and private network callers hear a busy tone instead of ringing while the user is on a call<br>Transfers external callers to the Prime set for answering<br>The line indicator for an external incoming call flashes, but the phone does not ring | Eliminates the distraction of a second line ringing, while ensuring that external callers are routed to an answering position |
| Enhanced Call Restrictions and Overrides | Maximum Number of Dialing Filers: 100 | |

| Features | Description | Benefits |
| --- | --- | --- |
| Enhanced Trunking Connectivity Private Network – E&M (tie) Trunk Connectivity | E&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<br>For each system within the network, the length of directory numbers (DNs), line pools and line pool access codes are the same | Ensures cost-effective efficient internal communications |
| Enhanced Trunking Connectivity Public Network – DID Trunk Connectivity | DID (Direct Inward Dialing) trunks let incoming callers bypass the attendant and be directly routed to a target line | Improves call handling |
| Enhanced Trunking Connectivity Public Network – Remote Access | Remote access (with or without DISA) provides off-site remote access to BCM private or public network facilities, which avoids public network toll costs | Helps to improve cost controls |
| Executive Busy Override (Priority Calls) | Allows users within a BCM to force a voice connection to busy set or one on “Do Not Disturb” anywhere in the system | In the case of a true emergency, the calling party can establish contact. |
| External Calls on Intercom Keys | Program lines to ring on an intercom key | Users can utilize more buttons for features by using intercom keys for external calls |
| External/Network Transfer | Allows the user to transfer calls over the public or a private network | Improves customer service by allowing the user to transfer the caller to the correct party, even if they are not on the BCM |
| External Line Access | Allows users to directly access outside lines by buttons on individual phones or indirectly by a line pool | Users can bypass the receptionist to place outside calls, thereby saving time and money |
| Feature Access Key | Allows users to program any feature code onto a memory button | Offers fast, single-button access to frequently-used features |
| Flexible Call Restrictions and Overrides | Call restrictions and overrides can be applied to individual lines and/or sets, but can be overridden with passwords | Maintains Call Restrictions cost control, yet provides access to selected numbers within the restricted categories |

| Features | Description | Benefits |
|---|---|---|
| Flexible Numbering Plan – Changing DN Length | The length of the Directory Number (internal number) can be from two to seven digits<br>All DNs in a system must be of the same length | Provides user with flexibility in assigning internal numbers<br>Helpful when the system is part of a network and a uniform series of internal numbers is required |
| Group Listening | Allows users to hear an incoming voice on both handset and speaker, while an outgoing voice occurs only through the handset | While a group listens to a call through the speaker, the caller hears more clearly through the receiver, as it eliminates background noise |
| Group Set Copy | Allows the system programmer to copy data from one set to a range of DNs<br>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 T7316 to all T7316s within the range)<br>Copying can be done for a particular subheading of programming or to duplicate all or a portion of programming for a set | Users can answer the telephone and continue working; the caller terminates the conversation |
| Handsfree Answerback | Internal voice calls automatically turn on the set microphone so users can reply without touching the set. (Not available with the T7100) | Allows a user to answer the telephone and continue working; the caller terminates the conversation |
| Handsfree – Automatic | 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) | Saves time by providing more convenient handsfree operation |
| Hold – Automatic | 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 | Prevents internal and external calls from accidental cutoff during transfers |
| Hold – Exclusive | Your call can only be retrieved at the set where it was placed on hold | Ensures privacy |
| Held Line Reminder | External calls on hold play periodic reminder tones over the set speaker until the call is retrieved | Improves customer service, as users will be less likely to forget the call |

| Features | Description | Benefits |
|---|---|---|
| I-Hold/U-Hold Indication | LCD line indicators will flash faster for held calls at the user's own set than for calls on hold at other sets | Users can easily identify calls held at their own set from those held at others' sets |
| Hospitality Feature set | BCM includes a set of three features that are applicable to the hospitality industry:<br>Alarm Feature – Alarm clock operation on Business Series Terminals<br>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.<br>Room Condition – This feature provides setting and querying the serviced condition (service done or service required) for the room | Convenient, direct access to a frequently called location saves time and effort |
| Hot Line | Allows users to program a set to call a specific internal or external number whenever they lift it or press a handsfree button | Convenient, direct access to a frequently called location saves time and effort |
| Hunt Groups | Enable single DN to call a group of sets<br>Three hunting modes are available: broadcast, sequential and rotary. All Business Series Terminals, Companion sets, Attendant Consoles and 2500 analog sets can be assigned to a hunt group<br>Silent Monitor for Hunt Groups has been added in BCM 3.0.<br>*Maximum Number of Groups: 24*<br>*Number of Members per Group: 40* | Improves call answering coverage and allows calls to be directed to specialized knowledge<br>Silent Monitor allows a supervisor (using a digital set) to silently monitor a call based on the hunt groups set up. This allows a business not using one of the call center applications to monitor calls |
| Language Choice | 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) | Provides the user with flexibility to accommodate alternative language requirements |

| Features | Description | Benefits |
|---|---|---|
| Last Number Redial | Allows users to redial the last externally dialed number<br>*Number of Digits: 24* | Saves time |
| Line Assignment (Set) | A maximum of eight lines can be assigned to any of the sets in the system | Informs the user of whose line is ringing |
| Line Names | Names can be programmed for incoming and outgoing lines | Informs the user of whose line is ringing to permit more personalized greeting when answering; users can also quickly identify which outgoing line they are using |
| Line Pool(s) | 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 | When lines are shared among a number of sets, line costs and the number of button appearances are reduced |
| Line Pool(s) – Busy Status | When all lines in a Line Pool are busy, the associated LCD set indicator will turn on | Saves time, because users don't have to keep checking for a free line |
| Line Profile | Line settings programmed in Configuration and Administration will appear on the T7316 or M7324 set display | Programming information is easily verified |
| Line Redirection | Often referred to as Selective Line Redirection<br>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<br>Redirected calls cannot be answered from another set (Not available with T7100) | Users can receive calls at any location |
| Line Selection | Users can press idle or ringing lines manually to override the automatic line selection feature | Lets a user 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 | Allows the BCM to hear a "second dial tone" before accessing Centrex or PBX features |

| Features | Description | Benefits |
|---|---|---|
| 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<br>The call can now be monitored through the speaker | Users can work handsfree while waiting for the caller to return |
| Log Space (CLID) | BCM provides a maximum of 600 Call Log spaces | Users can customize their own set<br>Users can capture the calling information on important calls |
| 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) | Users can customize their own set<br>Users can capture the calling information on important calls |
| LOGIT (Manual Logging) | If calls are not automatically logged, lets users manually log an incoming call after they answer it | Users can capture calling information for only those calls important to them<br>The automatic one-button process eliminates manual system, eliminates errors |
| Long Tones | Sends long DTMF tones to access devices | Users can operate devices requiring long continuous tones |
| Loss Package | Compensates for Loop Start (analog) trunk quality<br>Allows selection of appropriate loss/gain and impedance settings for each line<br>The settings are based on the distance between location of the BCM and the service providing central office | Provides improved QoS for customers with analog lines |
| Message Leave (List) | Display messages (“Message for you”) are sent to other set displays requesting a callback | Saves time as callers can leave a message and do not have to call back or leave a message with the receptionist |
| Message Waiting (List) | Allows users to automatically call back the person who sent “Message for you” to their display; they can cancel the message | Improves internal communications |

| Features | Description | Benefits |
|---|---|---|
| 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) | Users can customize the line appearance on their sets, as well as line answering patterns |
| 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 | Tones or music assure callers they are still on hold and have not been disconnected, thus reducing the number of abandoned calls |
| Network Direct Dial | Allows users to dial one digit to reach a specific destination on either a public or a private network | Saves time and improves productivity |
| Night Service | Outside calls that normally ring at the prime set can also ring at additional, preselected sets during preset times | After-hours calls can be heard or answered anywhere in the system, improving internal communications and customer relations |
| Numbering Plan – Flexible | The length and sequence of digits needed to access other sets or outside lines can be controlled | Users can be given intercom numbers to match other PBX numbers |
| On-Hook Dialing | Allows users to dial directly from the dial pad and speak using the handset or Handsfree button | Convenient for the user |
| Paging – Internal | Allows users to initiate a page or be paged internally through the set speakers<br>It is also easy to make announcements through the telephone speakers to a select group of users or to all sets | Users away from their sets, but within the office, can still receive call notification |
| Paging Feature Enhancement | Page time-out is now programmable and a system-wide parameter can now administer the paging tone to be “on” or “off” | Minimizes the length of time the feature is tied up at one set or left on by accident |
| Paging – External | Allows users to make external paging announcements when BCM is connected to a user-supplied amplifier and speaker | Paging announcements can easily be sent to rooms or areas without telephone sets |
| Paging – External and Internal | Allows users to make announcements using both the telephone speakers and the office’s loudspeaker system | Users can easily make systemwide announcements to everyone within the organization |

| Features | Description | Benefits |
|---|---|---|
| Paging Set Access | Individual sets can be denied the ability to perform paging | Improves security of paging system. Sets in open areas can be denied the ability to page (i.e. classrooms or motel rooms)<br>Ideal when employees share phones or move around an office |
| Password Protection | Allows the coordinator to change the system administration password | Helps to protect system-programming data |
| Preselection/Call Screening | The assigned name of the caller's set or line will appear on the set display | Users can identify the caller or the line being used |
| Prime Line | A line (CO, Intercom or Line Pool) can be assigned to a set as its primary line of use for automatic outgoing line selection | Saves time, as a user can begin dialing immediately without selecting a line |
| Prime Set | 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 | Offers more efficient call handling |
| Priority Call | Can interrupt a conversation on a busy set or override Do Not Disturb (DND)<br>Users have the option to block a Priority Call, but it cannot be ignored | In the case of an emergency, the caller can be reached |
| Privacy – On Lines | Automatically prevents another telephone, which shares a user's line, to access or join the call | Ensures security and confidentiality |
| Privacy Control | The Privacy ON/OFF switch lets a third person join the call | Users can easily turn a two-way call into a conference call |
| Receiver Volume | Allows users to program volume | Individuals within the system can set their own receiver volume |

| Features | Description | Benefits |
|---|---|---|
| Remote System Access | 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)<br>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<br>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 | Selected users are given convenient and efficient access to a restricted set or line |
| Restriction Override Password | Allows users to bypass any call restrictions applied to any set or line | Selected users are given convenient and efficient access to a restricted set or line |
| Ring Again on Busy Set | Alerts a user when a previously busy set becomes available | Saves time, as users are free to continue working while waiting |
| Ring Again on Busy Line Pool | Alerts a user when a line becomes free in a line pool | Saves time and frustration, as users can continue working without monitoring the availability of busy outside lines |
| Ring Again on No Answer | Notifies a user when a set that was not answered is used | Avoids unnecessary redialing. It is an efficient way to know when someone has returned to their office |
| Ringing Line Preference | Automatically places the longest ringing call to the head of the queue when several lines are ringing | Improves customer service as a user can answer calls in the order they were received |

| Features | Description | Benefits |
|---|---|---|
| Ringing Service | Now alternate ringing can be programmed for day of week as well as time of day | Provides the flexibility to change which set rings after hours or on weekends |
| Routing Service/Destination Codes | A programming section that allows outgoing calls to be directed automatically, based on the numbers a caller dials (also called Automatic Route Selection – ARS)<br>For systems linked in a network, routing can create a transparent or coordinated dialing plan<br>To make programming routes easier, digit absorption feature has been added to the Routing feature of BCM. Digit absorption selects the portion of the destination code that is always absorbed by the system and not used in the dialing sequence<br>*Number of Destination Codes: 500*<br>*Number of Destination Routes: 999* | Installer programming time is greatly reduced, in terms of entering the routing codes, and more flexibility is provided for routing codes |
| Saved Number Redial | It saves and later recalls the external telephone number appearing on the display | A person can quickly redial the number, saving time and effort |
| 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" | Provides an effective deterrent to malicious and nuisance calls, especially when combined with Call Trace |
| Selective Line Redirection | See 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 | Improves network control, since an administrator can view programming information on specific set configurations and general administration data |

| Features | Description | Benefits |
|---|---|---|
| Service Modes | Three different service modes can be programmed (i.e., lunch, evening, night) with their own ringing arrangements for automatic or manual activation | Increased control over the system's call handling setup over three periods of time offers greater system flexibility |
| Speed Dial Access – Personal and System | Allows users to access both Personal and System Speed Dial Codes<br>*Number of Digits: 24*<br>*Number of Entries: 70* | Prevents dialing mistakes, saves time and is more convenient |
| Speed Dial Line Selection | For speed dialing, the system will use a specific line as determined in administration | Line selection allows the preprogramming of specific lines for each number |
| Speed Dial – Personal Programming | Allows a user to add or change a Personal Speed Dial number on their set | Fast error-free dialing is provided for a user's personal telephone directory, with increased flexibility |
| Speed Dial System Bypass Restrictions | Allows users to program speed dial numbers to override set and line restrictions | Allows easy access to selected numbers within restricted categories |
| Speed Dial System Names | Allows users to program names instead of numbers on the set display when they access the speed dial code | Keeps the dialed number confidential |
| Start DN Option | Allows user to choose the start DN and DN length, rather than the previous mandatory 221<br>Is typically programmed by the installer under Configuration | Provides flexibility of uniform Directory Numbers over a network |
| Station Set Test | 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<br>Pressing Feature 805 on the set activates Station Set Test. The LCD prompts the user through the testing procedure<br>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 | If users or service representatives suspect something is wrong with a button, the speaker, the display or some part of the telephone hardware, a quick test can be done to see which part is malfunctioning |
| System Wide Call Appearance (SWCA) | | |

| Features | Description | Benefits |
|---|---|---|
| Target Lines | 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 | Enables the optimum use of available resources |
| Telephone Administration Lock | Three settings (Full, Partial, None) can be programmed in Administration | Controls the specific features a user can program/use on their set |
| Time and Date Display | The time and date appear on the LCD display of an idle set | It eliminates a person's need to have a workstation clock |
| Time and Date – Show Time | Temporarily displays (for three seconds) the time and date while on a call | Useful in a busy office environment |
| Timed Release | Signal releases a call from the line, but keeps the line for another call | If users are making consecutive fax or data calls, they don't have to worry about access to that line |
| Transfer Immediate (with Callback) | Allows users to transfer calls directly to another set; if unanswered, callback occurs after a preset number of rings | Improves handling of transferred calls |
| Transfer Using Conference | Allows users to transfer a call to an internal number | Ensures a party is not lost during a transfer; better customer service and more professional call handling result |
| Transfer Using Hold | Allows users to transfer calls using the Hold button | Provides a quick, convenient method for transferring calls among users sharing the same line appearances |
| Transfer with Announce | 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 | Improves efficiency, as the user transferring a call can verify the person is available to receive the transfer |

| Features | Description | Benefits |
|---|---|---|
| Unsupervised Conference | 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 | Two parties can continue a conversation not relevant to the third one without the inconvenience of disconnecting and recalling each other |
| Voice Call | Allows users to make a voice announcement or begin a conversation through the speaker of another telephone | Improves flexibility |
| Voice Call Deny | Prevents a set from receiving Voice Calls | Interruptions from voice calls can be prevented while the user is still alerted of incoming internal calls |
| Wait for Dial Tone | Causes a sequence of numbers to pause until dial tone is present on the line before continuing to dial | |

# Telephony Enhancements

BCM Release 3.0 introduces the following telephony enhancements:

- Increased TDM Station Capacity
- Telephony Routing Enhancements
- Increased Analog Station Capacity.

## Increased TDM Station Capacity

The number of station sets supported on the BCM is a critical capability for meeting the needs of larger sites. Through a more efficient use of existing resources, the number of digital sets supported has been increased to a theoretical maximum of 192. In practical solutions with a mixture of trunks and stations, between 128 and 160 digital sets will be supported, effectively doubling the digital station capacity of the BCM.

BCM 3.0 does not change the number of DS-30 channel resources or the ability to change the IP/TDM ratio of channels. This release allows new digital station modules (DSM 16+ and DSM32+ introduced in fourth quarter 2002) to make more efficient use of the DS-30 channels to increase the number of digital station sets supported on the BCM. This is accomplished by using the B2 DNs as B1 DNs for TDM stations.

Consequently, one DS-30 channel will now support up to 32 stations. Using 100% VoIP trunking (i.e. no TDM trunks) BCM 3.0 will support the following maximum TDM stations:

- 2/6 IP/TDM split – 6 DSM32+ provides ≤192 TDM & ≤58 IP for a total of 250 stations
- 3/5 IP/TDM split – 5 DSM32+ provides ≤160 TDM & ≤90 IP for a total of 250 stations.

**Note:** The maximum number of IP sets is reduced by one for every CallPilot, Call Center or IVR channel configured, up to a maximum of 32 channels. The default CallPilot setting is 6 channels configured, meaning the maximum number of total sets is 244 to start with. The number of IP sets is further reduced by one when IP trunks are implemented.

In solution scenarios for the BCM where the focus is on TDM stations and four DS-30 channels are used for TDM station support (assuming a 2/6 split and the use of the other two channels for trunks), up to 128 TDM stations can be easily supported. (Additional Target Lines are also available to maintain a 1:1 ratio of DNs to Target Lines.) Actual capacity may vary based on customer configuration requirements and BCM engineering guidelines.

## Telephony Routing Enhancements

The BCM destination code routing has been enhanced to allow routing on up to 12 dialed digits as opposed to the previous maximum of 7 dialed digits. This feature improves the routing services. The increase in the number of digits in a destination code will increase the number of distinct destination codes that can be programmed. This provides more flexibility and usability of routing services to the customer. For example, this feature provides call routing on up to 12 dialed digits to accommodate a 1-800 help desk number for which there are special routing requirements.

Another enhancement in Release 3.0 involves providing multiple alternate routes (more than one route) for each service mode. The Least Cost Routing (LCR) feature can be more effectively utilized 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.

## Increased Analog Station Capacity

The changes in the core telephony in Release 3.0 that increase the TDM (digital) station capacity allow for increased analog station capacity as well. This means that the Analog Station Modules (ASM8) make more efficient use of DS-30 channel resources and more analog sets can be supported.





# Data Capabilities

# Data Capabilities

## Chapter Highlights

- Nortel Networks BCM 3.0 – has industry-leading networking capabilities, including native IP packet transfer, VoIP capabilities and embedded firewall and security features.
- Firewall with Basic and Stateful Packet Filtering – enhances security and control by blocking packets from entering or leaving a site.
- Network Address Translation (NAT) – allows a host with a private address to appear on the Internet with a registered address.
- Dynamic Host Configuration Protocol (DHCP) Server – assigns IP addresses automatically for easy administration. DHCP Relay Agent allows pass through of DHCP traffic to and from LAN connected devices.
- Web Cache and Domain Name Service (DNS) Proxy – stores frequently used data and allows faster access and less congestion.
- QoS Support - prioritizes IP traffic and provides an acceptable quality of service to delay and jitter-sensitive applications such as audio and video.
- LAN-to-LAN Fast Path Routing – is an innovative design that significantly speeds up the performance of LAN-to-LAN routing by over three times the rate that is normally achieved.

## Overview

BCM 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
- IPX support on both the LAN and WAN interfaces.

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 Networks Secure Routing technology. This technology stems form the industry leading Contivity line of products and is an excellent means of combining VPN support with security in one device.

# Emerging Trends

Generally, the branch offices of small to medium-sized businesses have mix-and-match systems of independent network components that have been added over time as they became necessary. As a result of having different devices from different vendors – each with its own proprietary interface – maintaining and upgrading telephony and data services at each branch location is a complex and expensive task.

In order to remain competitive and to ensure continued success, businesses with branch locations must continually expand the reach of centralized resources and provide value-enhancing front-line services to each branch office. Thus, many businesses today are increasingly looking to implement communications solutions with built-in data capabilities. The BCM is ideal for small to medium-sized businesses with branch locations because it eliminates the need to purchase additional equipment. BCM provides integrated firewall and security measures like Network Address Translation (NAT), basic and stateful packet filtering, Point-to-Point Tunneling Protocol (PPTP), IPSec VPN support and DNS proxy. In addition, BCM Web caching offers LAN workstations shorter download times by reducing WAN traffic.

# Benefits

BCM meets the data needs of most enterprise branch offices with its built-in capabilities. For small sites, a BCM can provide turnkey access to the Internet and private corporate networks, giving employees access to communication channels that will boost their productivity and ability to communicate. The BCM features integrated routing capabilities and support for numerous LAN and WAN protocols and interfaces, as well as features to enable secure data transmission over public networks. A single BCM simplifies corporate communication networks by eliminating the need for a separate router, multiplexer and remote access server at each branch location.

Businesses can secure their communications costs by using the built-in security features of BCM. Virtual private networking (VPN) and authentication and encryption services allow sensitive corporate information across public WANs instead of leased lines.

# 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.

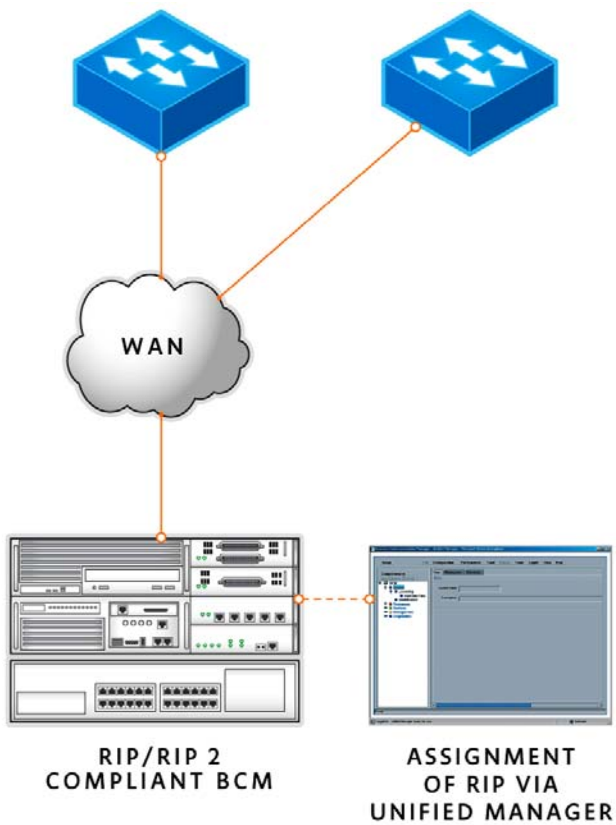
## RIP v2

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.

***Figure 4-1***



## OSPF

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. OPSF 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.

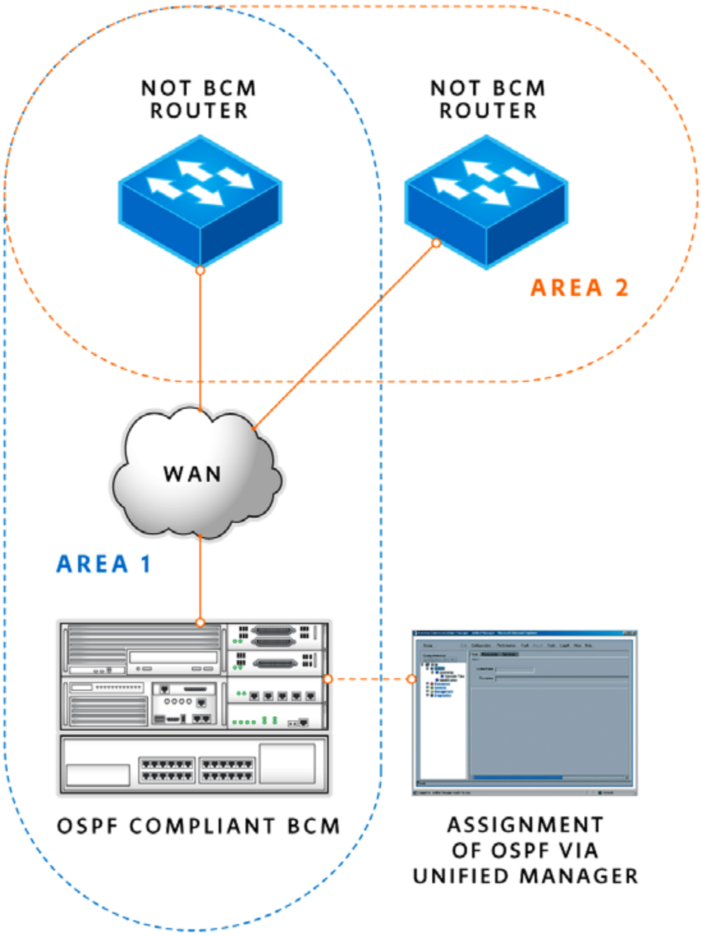
OSPF offers the following advantages:

- 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 4-2***




BCM also supports Static, RIP and SAP routing for IPX networks.

## 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 3.0 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.

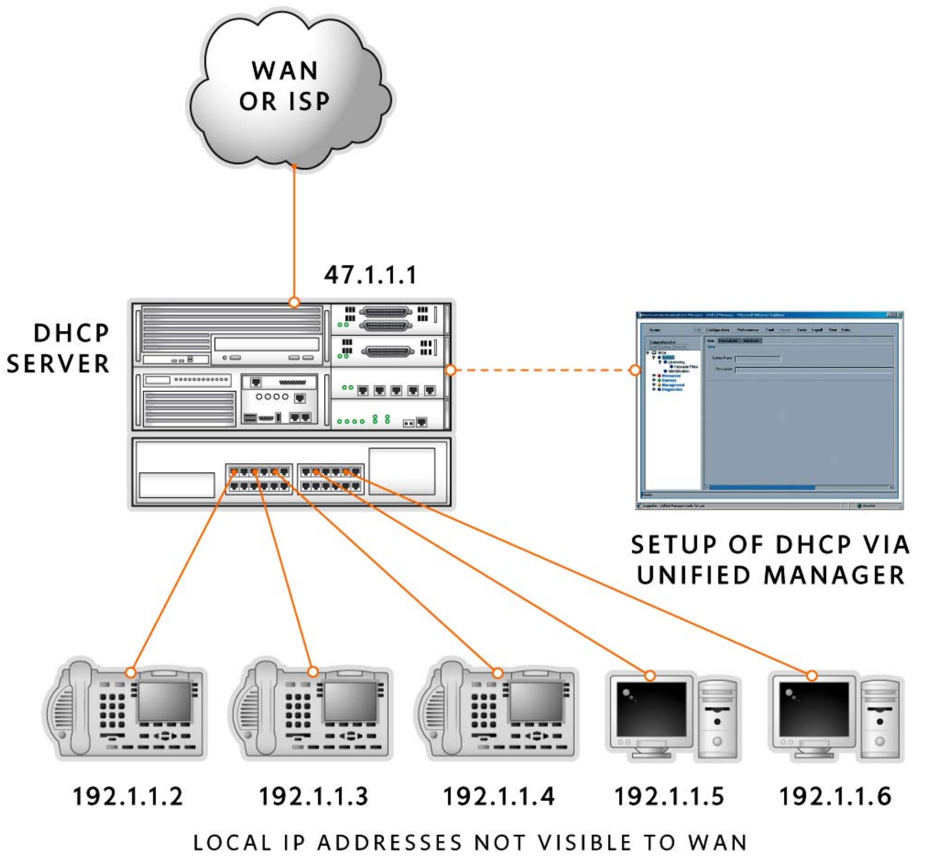
DHCP supports the following attributes:

- 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 i2050 IP telephony 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 4-3*




## 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 Networks 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.

New to BCM 3.0 is the ability to support 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 IPX tunneling
- 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.

# 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.

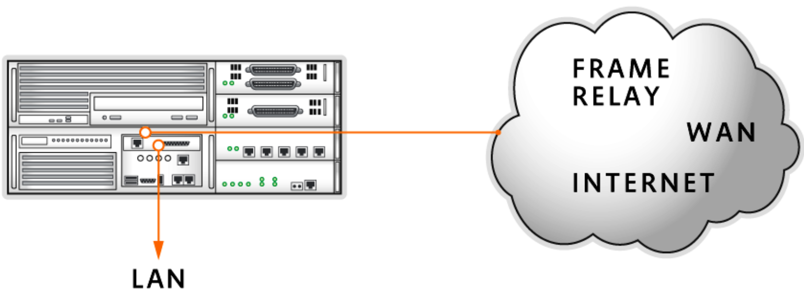
- BCM provides a choice of permanent (T-1, V.35, X.21) or dialup (ISDN dial-on-demand or persistent) WAN connections as well as ISDN or V.90 analog dial backup.
- The primary WAN link is always a permanent link and it is a dedicated network adapter. It runs either Frame Relay or the PPP protocol at the 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.
- 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.

## 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.

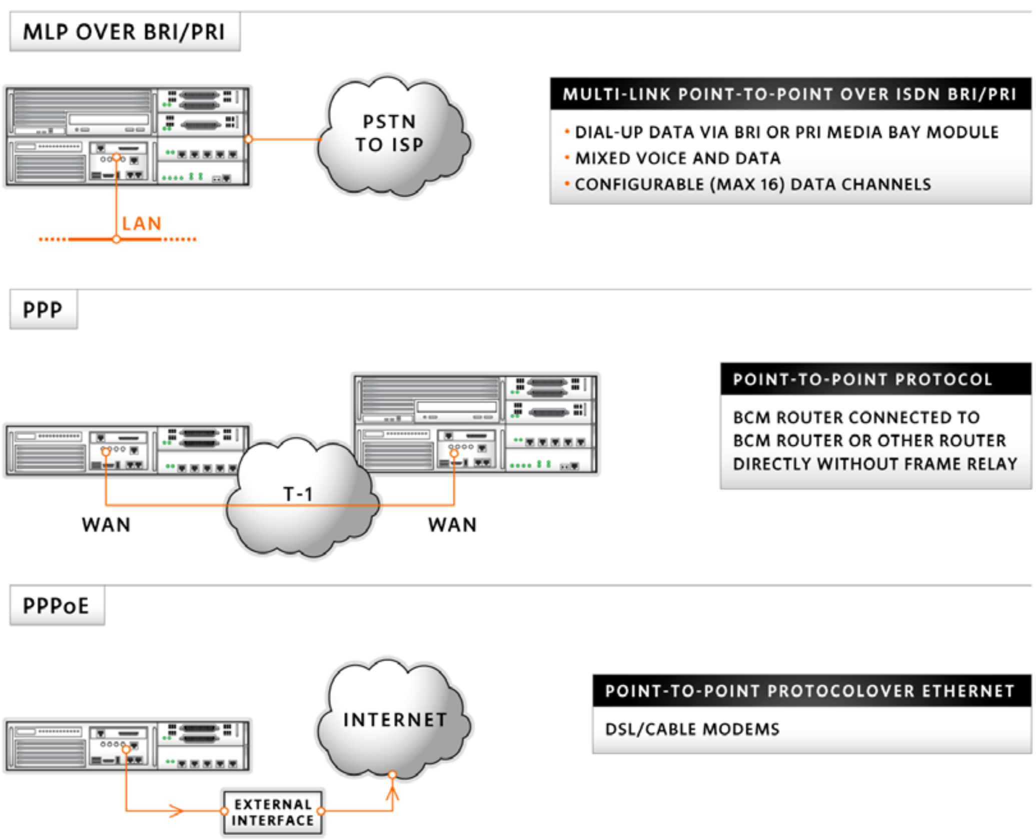
Direct mode enables a single T-1 to support Intranet or corporate WAN access and Internet access, dynamically sharing bandwidth. This eliminates the need for a second T-1, which significantly reduces the recurring monthly cost of network access.

***Figure 4-4***




## Point-to-Point Protocol

***Figure 4-5***



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.

## Common Open Policy Service (COPS)

BCM provides the capability of Common Open Policy Service (COPS), a key ingredient in Nortel Networks 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.





# Messaging

# Messaging

## Chapter Highlights

- Nortel Networks CallPilot Messaging Service – is an application common to BCM, Meridian 1, Norstar, Succession Communications Server for Enterprise, M6500 (Europe, Africa and Middle East only) and Passport platforms. Built on proven messaging experience, CallPilot provides low cost of ownership, ease of use and ensures compatibility with future developments.
- Target Market for CallPilot Messaging Service – consists of businesses of all sizes, including small companies that want to project the image of a larger organization; SMBs that want advanced applications such as Speech Activated Messaging or Unified Messaging; and businesses that have large messaging networks with multiple messaging systems.
- Automated Attendant – works as a voice messaging “receptionist” and ensures that calls are answered 24 hours a day, 7 days a week, allowing callers to direct their calls to the right person.
- Mailboxes – enable companies to order up to 1000 mailboxes that can be used to provide callers with important information or allow callers to leave messages anytime.
- Custom Call Routing – enhances Voice Messaging call routing abilities by allowing incoming calls to be redirected along call paths created by the system administrator.
- Networking – uses various standards including: Audio Messaging Interchange Specification (AMIS) Voice Profile for Internet Mail (VPIM), Meridian Customer Defined Networking (MCDN), Q.SIG and DPNSS, to create seamless integration with existing communications systems such as Meridian Mail, Norstar and others.
- Unified Messaging – lets users merge voice, fax and email messages into a single, PC-based interface.
- Integrated Voice and Fax Messaging – provides the ability to receive, store and process voice and fax messages in the same "multimedia" mailbox, reducing the need for physical fax machines and time- and cost-consuming fax transmissions (particularly international calls).

# Messaging Sales Scenario

## The Challenge

E-creators is a medium-sized software company with 70 full-time employees and six branch offices located across the country. E-creators' versatile range of products caters to Internet users, from major corporations and professionals to general users, helping them to effectively save time and money online. A rapidly expanding business, E-creators recently concluded that to continue to meet its clients' expectations and capitalize on new business opportunities, it needed to rethink its communications strategy.

Like most office workers, employees at E-creators spend much of each business day communicating – making and receiving telephone calls, sending faxes, answering emails and performing these tasks by different means. As with many businesses, E-creators has an attendant who performs call handling and reception duties. Without a messaging solution in place, employees were required to retrieve their own messages from the front desk attendant. Consequently, employees were becoming increasingly frustrated with the amount of time they were wasting, while callers were becoming annoyed because they often had to wait several hours for a return call.

Since E-creators did not have any direct-in-dial phone lines, callers were often unable to reach people after hours or when the attendant had left for the day. And, as software engineers and support personnel often work late, this posed a problem. Communicating between sites was also a challenge, especially in dealing with different phone systems.

Clearly, E-creators needed to control and simplify its communications; it needed a solution that would allow its employees to save time and increase productivity. And, since remote office workers are an integral part of E-creators' business, it needed to ensure that salespeople could keep in touch while on the road.

## The Nortel Networks Solution

For E-creators, the solution was CallPilot on BCM Voice Messaging combined with Unified Messenger. With CallPilot, users can provide callers with specific information that will let callers know when to expect a return call, while eliminating the need for a front-desk attendant to take handwritten messages.

The Auto Attendant functionality allows callers to direct their calls to the right person during after-hours situations or when the attendant is away or busy. The front-desk attendant can focus on more important tasks and be available for callers who press zero for assistance.

Building on the customer-driven functionality of Nortel Networks proven messaging products, Unified Messaging adds highly graphical user interfaces that make system management easy and effective. Unified Messaging combines voice, fax and email into a single point-and-click inbox. It integrates with more email systems than any other messaging product, 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.

Via its email interface, CallPilot enables voice message playback using either a telephone or speakers on a multimedia computer. Users can also store voice messages on the PC for archival purposes and forward .wav files as email attachments. Additionally, users who travel frequently can dial in from remote locations, such as hotels or branch offices and have voice messages streamed to the PC. Voice messages are streamed to the PC to minimize the impact on network bandwidth and enable fast playback.

Through CallPilot, fax management with Unified Messaging is secure and easy. Providing password-protected access to faxes, it enables outbound faxing from any Windows application. This ability reduces the need for additional fax machines or analog lines to each desk to enable fax from the PC.

## Implementation

Easy to install and maintain, CallPilot requires minimal training and start-up costs.

CallPilot reduces cost of ownership through the use of industry standards to help integrate with existing infrastructure as well as ensure compatibility with future developments. And, with its familiar Windows-based interface and online help, users and administrators can learn to use the CallPilot software in a fraction of the time it might take for them to learn other systems.

Unified Messaging works with a variety of networks, including Meridian 1, Succession Communications Server for the Enterprise, SL-100, DMS-100, DMS-500 and Matra as well as Lucent, Mitel and Siemens-Rolm. This flexibility meant E-creators was able to leverage its investment in Nortel Networks by installing CallPilot on their existing Meridian 1 system.

### Business Impact

Auto Attendant has made a significant impact on E-creators' business, specifically with respect to time and productivity. The attendant can now take on more meaningful tasks, as she no longer has to redirect calls. Callers now receive information via ABC employees' personal voice greetings.

Employees can now quickly and easily manage their voicemail and email messages from their PC or laptop, locally or remotely. Road warriors can dial in from any touchtone phone to retrieve their messages or change their greetings at any time. As virtually all of the functions of CallPilot are intuitive, designed around the common operations of most desktop computers, E-creators' employees easily interacted with the new system and required little training.

E-creators has realized cost savings as a result of multimedia messaging. The ability to receive, view, store or route faxes electronically greatly reduces the need for physical fax machines and time- and cost-consuming fax transmissions, particularly when serving E-creators' international customers. The electronic format also means fewer printouts, which reduces expenses on paper, toner and machine maintenance. With Unified Messaging, E-creators has eliminated the need for system management services, since CallPilot administration can happen easily and quickly from a single desktop computer anywhere on the local area network (LAN), using simple and intuitive graphic interfaces.

## Overview

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

Business Communications Manager (BCM) incorporates features that address the needs of small and medium-sized customers. It includes exciting new ebusiness applications, like Multimedia Call Center, that allow the small company to level the competitive playing field and project the image of a much larger organization. BCM also incorporates two new options to increase mobility and accessibility to information.

BCM has always been unique, in that it offers a solution to the customer who wants to take advantage of what technology has to offer, but is not ready to chance their company's success on a risky, featureless IP-only product. BCM gives these customers a future-proof choice, which now includes IP station support to complement their existing digital stations.

## Emerging Trends

Today, many businesses are attempting to solve their communication problems by taking advantage of the advanced applications available on the market and implementing solutions that enhance basic voicemail and auto attendant features. The BCM has been designed to provide multiple application support so customers can reduce their cost of ownership by using one server.

In response to numerous market forces over the last couple of years, a number of unified messaging products have been introduced in the marketplace. BCM Unified Messaging was designed to work with Microsoft Exchange and Outlook email clients. Email support for BCM Unified Messaging has been expanded and is now compatible with:

- Outlook Express
- Netscape Messenger
- Lotus Notes
- Qualcom Eudora Pro
- Novell GroupWise.

As email applications and PC operating systems evolve, so does the BCM. The following table shows the email applications and operating systems supported by BCM:

***Table 5-1***

| **Application** | | |
|---|---|---|
| **Groupware Email Clients** | **BCM 2.5 FP1** | **BCM 3.0** |
| Microsoft Exchange – 4.x | | |
| Microsoft Exchange – 5.x | X | |
| Microsoft Outlook 97 | X | |
| Microsoft Outlook 98 (Corporate Mode) | X | X |
| Microsoft Outlook 2000 | X | X |
| Microsoft Outlook 2002 (XP) | X | X |
| Lotus Notes – 4.5x | | |
| Lotus Notes – 4.6x | | X |
| Lotus Notes – 5.x | X | X |
| Lotus Notes – 6.x | | |
| GroupWise – 5.5x | X | |
| GroupWise – 6.x | X | X |

***Table 5-2***

| **Internet Mail Clients:** | **BCM 2.5 FP1** | **BCM 3.0** |
|---|---|---|
| Microsoft Outlook Express (Internet Explorer 4.0) – 4.x | | |
| Microsoft Outlook Express – 5.x | X | X |
| Microsoft Outlook Express – 6.x | X | X |
| Microsoft Outlook 98 (Internet Mail Mode) | X | X |
| Microsoft Outlook 2000 (Internet Mail Mode) | X | X |
| Netscape Messenger (Netscape Communicator) – 4.05 | | |
| Microsoft Outlook 2002 (XP) (Internet Mail Mode) | X | X |
| Netscape Messenger (Netscape Communicator) – 4.5 | X | |
| Netscape Messenger (Netscape Communicator) – 4.6 | X | |
| Netscape Messenger (Netscape Communicator) – 4.7x | X | X |
| Netscape Messenger (Netscape Communicator) – 6.2x | | X |
| Qualcomm Eudora Pro – 4.02 | | |
| Qualcomm Eudora Pro – 4.2 | | |
| Qualcomm Eudora Pro – 5.x | X | X |

***Table 5-3***

| Operating Systems | BCM 2.5 FP1 | BCM 3.0 |
|---|---|---|
| Windows 95A | X | |
| Windows 95B | X | X |
| Windows 98 | X | |
| Windows 98 SE | X | X |
| Windows ME | X | |
| Windows 2000 Professional | X | X |
| Windows XP | X | X |
| Windows NT 4.0 SP1 | X | |
| Windows NT 4.0 SP2 | X | |
| Windows NT 4.0 SP3 | X | |
| Windows NT 4.0 SP4 | X | |
| Windows NT 4.0 SP5 | X | |
| Windows NT 4.0 SP6a | X | X |

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), voicemail 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:

- CallPilot Voice Messaging
- Personal Mailbox Manager
- Operator Manager (found in Unified Messaging Menu).

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.

# Benefits

## CLASS/CMS Integration

In addition to standard telephone answering and auto attendant call routing benefits, Nortel offers some specific advantages because of the tight integration between the BCM and CallPilot Voice Messaging. When the BCM is equipped with CLASS/CMS or ISDN network features, messaging products provide some very powerful business tools:

- The Automated Attendant can route calls based on up to 100 Calling Line Identification (CLID) numbers to either a specific Custom Call Routing (CCR) tree, extension or mailbox. This routing can also be performed based on area code or prefix. This capability means that calling customers can be routed to their geographic customer service representative based on their calling number.
- Automated Number Identification (ANI)/CLID and the caller's name, if available, are stored in the mailbox with each message and users can call back those numbers using the “CALL” soft key on the telephone, speeding up return calls and simplifying the call return process.

In addition to the primary greeting and the extended absence (alternate) greeting, users can record up to three personalized CLID greetings for specific callers, like a special customer.

## Intelligent Integration

BCM systems were designed to provide superior integration with voice applications. A simple and easy-to-use interface displays visual prompts on the telephone set to guide the user through messaging commands and functions by using the soft keys just below the telephone LCD window. Also, the user receives message notification through a “Message for You” prompt, which appears on the display whenever there is a new voice message in the mailbox.

Other integration advantages include:

- Access to the name directory on the telephone LCD display
- Visual message waiting indication
- Ability to retrieve calls that have forwarded to voicemail by using “Interrupt” with Feature 987, thereby saving the user’s time.
- Ability to route calls and offer specialized greetings by using incoming CLID information

- Ability to transfer to an extension or external number from a CCR tree
- Ability to integrate CLASS/CMS and Integrated Services Digital Network (ISDN) features directly from the BCM system without adding additional hardware devices
- The use of one main telephone number for both voice and fax calls.

In addition to all the user benefits derived from the BCM integration, another significant benefit for the customer is that Nortel provides a single vendor solution with products that meet or exceed industry reliability and quality standards.

# Voice Messaging Overview

## Voice Messaging Components

Voice 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 voicemail 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.

## Fax Messaging

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

## CallPilot DTMF User Interface

Norstar Voice Mail and CallPilot (targeted for larger Meridian 1 customers) are both mature, feature-rich voice messaging applications and are identical at the feature and functionality levels.

They differ in Dual Tone Multifrequency (DTMF) user interface style, flow and feature detail. When a customer has both Meridian 1 with CallPilot and BCM in their network, the CallPilot DTMF user interface of the BCM reduces a caller's confusion by providing a consistent DTMF user interface with the Meridian 1 interface.

Callers learn that they can dial “701” to mark a message urgent, “0 and number” to dial through to an extension and “83” to hang up, regardless of whether the system is a Meridian 1 CallPilot or BCM CallPilot.

### Interface Support

BCM supports two different DTMF interfaces for users and callers:

- Norstar Voice Mail DTMF interface on existing BCM 2.0
- Meridian Mail/CallPilot DTMF interface.

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

### Interface Consistency

BCM provides a consistent Voice Mail interface for both BCM and Meridian 1.

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

### Standard Voice Messaging Software

BCM Voice 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)
- CallPilot Unified Messaging
- CallPilot Voice Mailboxes orderable in seat license increments of 1, 4, 8, 16, 32 and 64, up to 1000
- Fax Suite: provides a packaged software option that includes:
  - Fax Messaging: ability to receive fax messages in the voice mailbox
  - Fax on Demand: provides repetitive information to callers without human intervention
  - 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 chapter 8, *Call Center*.

## Automated Attendant

The Automated Attendant is the voice messaging "receptionist," whose duties are to answer and route incoming calls – 24 hours a day, 7 days a week. 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.

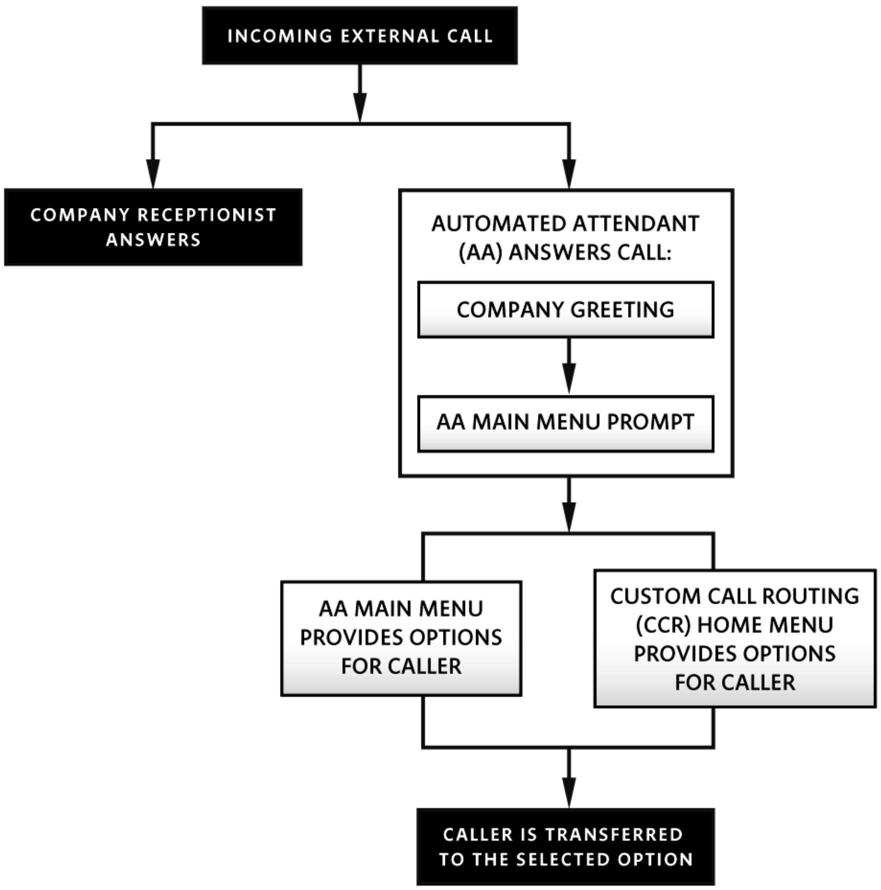
The Automated Attendant provides callers with commands for using each of these options. A caller must press the button associated with the option he or she wants to activate; for example, "To use the Company Directory, press the pound key."

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 5-1 shows an example of call flow with Automated Attendant:

***Figure 5-1***




## 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 Voice 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 voice mailboxes are orderable in seat license increments of 1, 4, 8, 16, 32 and 64, up to 1000 mailboxes. (Up to 200 network delivery mailboxes can be configured on the system without requiring a keycode.)

Types of voice mailboxes are:

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

### Special or Guest Mailboxes

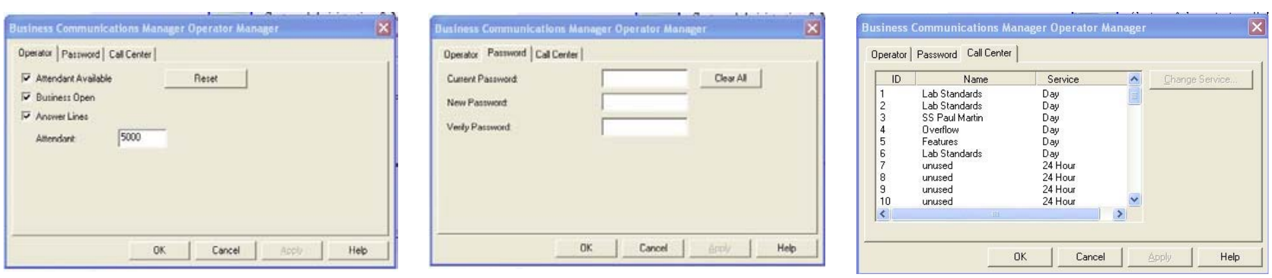
Special Mailboxes are administered by the company's system coordinator.

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 5-2) to administer and program the Voice Mail system.

***Figure 5-2***



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

- General Delivery
- Information
- Guest and Network Delivery.

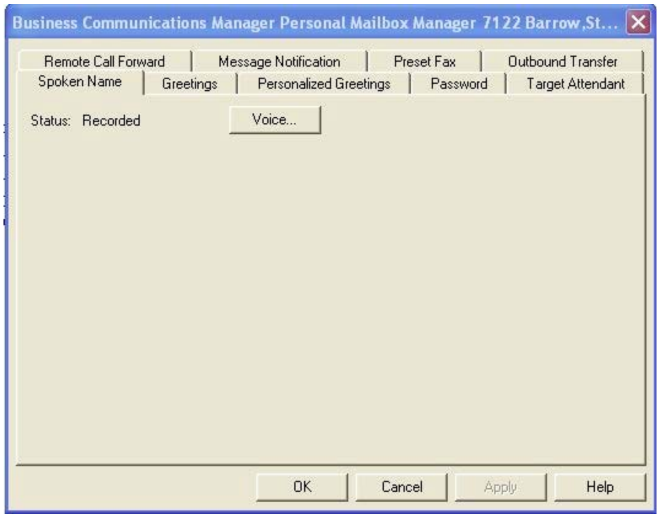
Operator Manager gives the administrator a fast and easy way to manage the Automated Attendant function without ever having to leave his or her desk.

### 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 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 5-3***



- User mailboxes can be assigned to each user who has a Business Series Terminal, M7XXX series set, analog set or a Companion portable 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 voicemail 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 Voice Mail systems and Meridian 1 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, or 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, Voice Messaging answers the call and temporarily stores the fax message in the Fax Overflow mailbox. Later, when the fax machine is ready to print, Voice Messaging sends the stored fax messages to the fax machine.

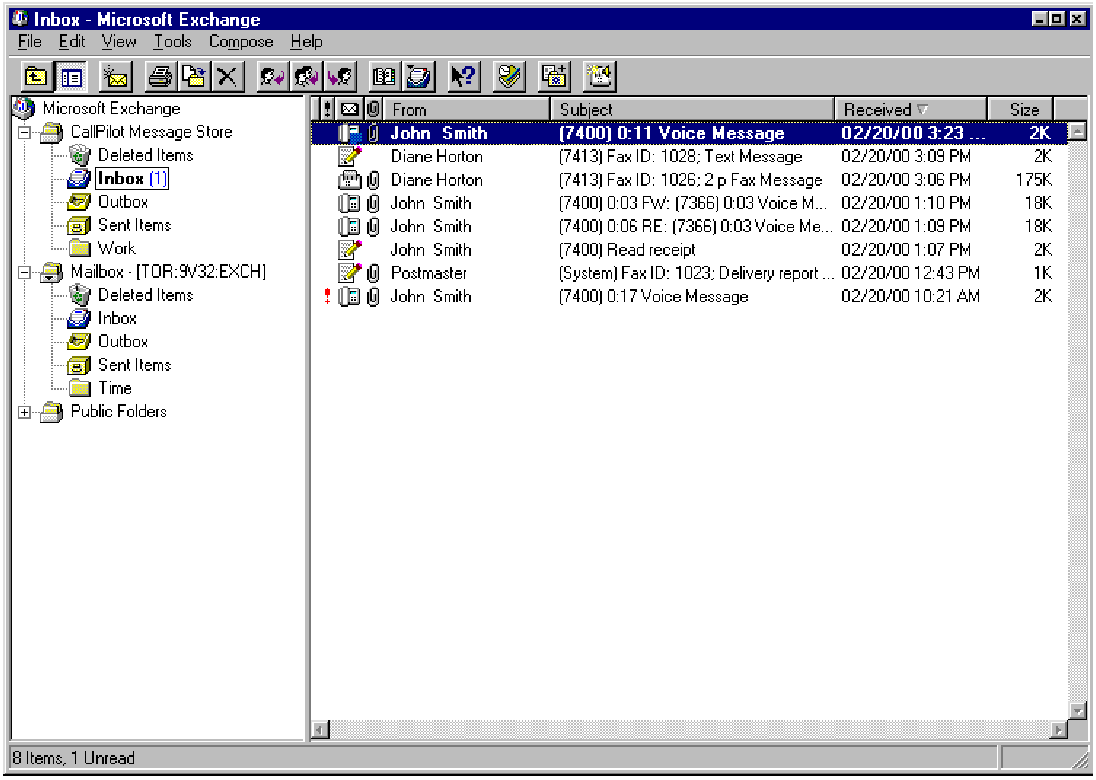
The Fax On Demand mailbox allows a user to retrieve stored documents. It contains a greeting and all the documents the user has stored.

# CallPilot Unified Messaging

CallPilot 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 voice mailbox so they can play their voice messages on their multimedia PCs using speakers or a headset for privacy.

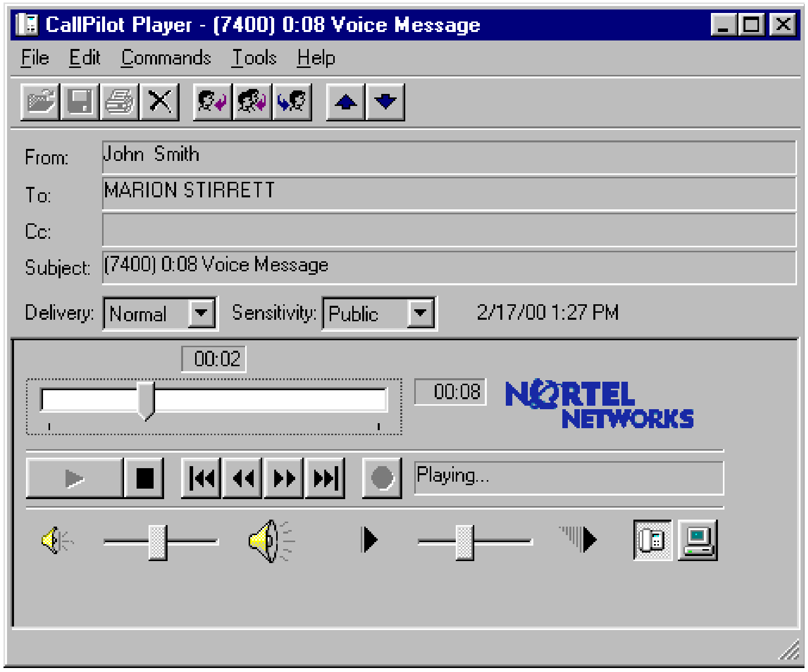
The Voice Message headers are displayed on the PC screen (as shown in Figure 5-4) along with any email and fax message headers. Users can see at a glance how many messages they have, as well as certain message details, especially where Calling Line ID is in use. This means users can choose which messages are more important to them and manage their time based on their priorities. Also, with CallPilot Unified Messaging, users can easily save messages to a folder or file.

***Figure 5-4***



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.

***Figure 5-5***




CallPilot 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.

## CallPilot Unified Messaging User Interface

Previous releases of BCM use the same Unified Messaging client software as Norstar Desktop Messaging, which was limited to working with Microsoft Outlook and Exchange. BCM 3.0 uses the Unified Messaging Client from CallPilot 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 CallPilot Unified Manager installation process is automated and considerably easier for the customer to install and set up.

An additional benefit is that if a customer has a mix of BCMs and Meridian 1s, having a single CallPilot Unified Messaging interface for both systems means that administration is simplified and end user training is reduced.

## Voice Messaging Feature Codes

When using BCM Voice 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 Voice 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 Voice Messaging while a caller is recording a message or listening to the Personal Mailbox Greeting
- Allows mailbox owners to retrieve calls from Voice 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 Voice Mail. 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 Voice Mail (Feature 984)**

- Allows mailbox owners to program a single button to forward all calls automatically to voicemail.

# Features and Benefits

## Administration

***Table 5-4***

| Feature | Description | Benefits |
|---|---|---|
| Backup and Restore | Applies to all system configuration attributes and messages to be backed up<br>Allows users to save system data in the event of operational problems | Preserves data during upgrades and for disaster recovery |
| Create Mailbox Utility | Seeks out all the extensions that do not have mailboxes and creates mailboxes for extensions with the following default characteristics:<br>• Identical mailbox and extension numbers<br>• The mailbox name (set name)<br>• The directory listing, if available<br>• "Yes," for message waiting.<br>• Will not create mailboxes if<br>• A mailbox with the same number already exists<br>• The extension is used by some other mailbox<br>• The extension is identified as a voicemail channel or other "system" extension. | Significantly speeds up mailbox configuration time, making the administrator's job far easier |
| Custom Directory | Allows callers to look up people in the directory by first name, last name or both<br>Search parameters entered by the caller can be changed without re-entering the names | Provides callers with a higher level of service.<br>Saves time and resources for the business. |

## AMIS Networking

***Table 5-5***

| Feature | Description | Benefits |
| --- | --- | --- |
| Direct Addressing | Allows a user to send a message to any mailbox in the network by entering the parameters required by Voice Messaging | Increases productivity<br>Workers can share ideas more frequently |
| Network Delivery Mailbox Addressing | Stores the parameters required by Voice Messaging to reach the destination mailbox | |
| Site-based Addressing | Allows subscribers to send voice messages to remote Voice Mail systems | Maximizes the use of mailboxes on the system versus using the network delivery method that requires a mailbox for every person at the remote site |

## Digital Networking

***Table 5-6***

| Feature | Description | Benefits |
| --- | --- | --- |
| Delivery Options | Network messages can be highlighted as Certified, Urgent or Private | Increases productivity<br>Saves time and money |
| Direct Addressing | Allows user to send a message to any mailbox in the network by entering the appropriate parameters | |
| Forward Network Messages<br>. | The network delivery mailbox stores the parameters required by CallPilot to reach the destination mailbox | Increase employee productivity |
| Non-Delivery Notification | When an error preventing delivery occurs, a network Non Delivery Notification message is generated by the intended recipient system. | Improves communications ensuring users are aware of any message delivery issues. |
| Reply to Network Messages | The person receiving a network message can reply and create an outgoing message that is already addressed to the originator. | Increase employee productivity<br>Simplifies and speeds communications |

| Feature | Description | Benefits |
|---|---|---|
| Site-based Addressing | 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 | Improves resource utilization<br>Maximizes the use of the mailboxes on the system compared to the network delivery mailbox method, which requires a mailbox for every person at the remote site |
| VPIM (Voice Profile for Internet Mail) compatible<br>CallPilot | VPIM is a voice messaging networking standard that allows systems from different vendors, to exchange voice and fax messages, over the internet. The BCM CallPilot is compliant with VPIM version 2.0. As of the end of 2001 the following vendors had VPIM compliant products:<br>Active Voice<br>Alcatel<br>AVT Corporation<br>MITEL /Baypoint Innovations<br>Centigram/ADC<br>Comverse Network Systems<br>Data Connection<br>ELETEL<br>Glenayre<br>Lucent<br>Media Gate<br>Microsoft – Exchange,<br>MJL Korea Ltd.<br>Nortel Networks – CallPilot<br>Nortel Networks – Business Communications Manager<br>Nortel Networks – Norstar Voice Mail<br>Nortel Networks – Meridian Mail<br>Siemens,<br>UniExchange<br>Unisys Corporation<br>Voice Data Systems | Greater flexibility, allowing the interoperability between products from various manufacturers |

## Automated Attendant Features

***Table 5-7***

| Feature | Description | Benefits |
|---|---|---|
| AA Menu Prompt | Can be turned on or off for each greeting table | Provides greater system flexibility |
| Call Transfer – Blind | Transfers calls directly to an extension with ringing starting immediately | Speeds up call processing |
| Call Transfer – Screened | Prompts callers to record their name, which is then played to the destination extension<br>Allows the called party to accept or reject the call without the caller's knowledge | Improves productivity<br>A called party can avoid unnecessary interruptions |
| Caller Display (Call Screening Support on Call Forward) | When call forwarding is enabled, all incoming calls are immediately forwarded to voicemail.<br>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 | Improves efficiency<br>Users can work uninterrupted when they need to but can still take certain essential calls<br>When used in conjunction with message interrupt, the user can see who is calling and if the call goes to their voicemail they can still retrieve the caller |
| Calling Name Display | Stores Calling Name with the message if the BCM is equipped with CLASS/CMS; the telco delivers Name Display | Customers and important callers are given priority |
| CLID Dialing Table Report | Lists all entries in the Call ID table<br>Contains, in each entry, a telephone number, destination type and destination number | Saves time<br>Users can create a list of frequently called numbers and an internal/external autodial directory |
| Dial Extension Number from CCR | Allows callers to dial any extension number from any menu point on a Custom Call Routing tree<br>*CCR levels: 10*<br>*CCR trees: 4* | Increases customer service.<br>Reduces cost to the business |

| Feature | Description | Benefits |
|---|---|---|
| Customized AA Menu Prompt Per Greeting Table | 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. | Increases customer service<br>Reduces cost to the business |
| Dual Language System Support | Allows callers and users to switch between two languages at either the Automated Attendant or Personal Greeting level of system prompts | Companies can use the language of their business (English or French in Canada and English or Spanish in the United States) |
| External Transfer on Centrex | Allows a multisite company to transfer callers between locations | Improves communication<br>Branch location teams can collaborate more often and more effectively |
| External Link Transfer, Single Trunk | (See Miscellaneous – Single Trunk External Link Transfer) | |
| Flexible Business Hours for Company Greetings | 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.<br>Allows businesses to pre-record greetings to match business hours | Improves customer service by eliminating any callers' confusion over business hours |
| Flexible Line Rings for Auto Answer | Transfers calls after a preset number of rings<br>Allows users to customize the system to meet their individual needs | Improves customer service |
| Greeting Tables | Each greeting table can be customized for the line they answer (e.g. sales vs. service, hours of operations). See also Customized AA Menu Prompt per Greeting Table | Improves customer service<br>Ensures customers receive the appropriate time-of-day greeting |
| Multiple Operators<br>. | Each greeting table can have its own attendant | Improves customer service, putting callers through to the appropriate operator |
| Remote Administration Menu | Remotely record company greeting<br>Remotely set business open or | Useful for severe weather or disaster conditions |

| Feature | Description | Benefits |
|---|---|---|
| | closed<br>Allows administrator (using Feature 983) to change any company greeting or remotely set the business status to open or closed | |
| Routing Calls Based on CLID | Allows the system coordinator to assign up to 100 unique telephone numbers to the Calling Line ID table<br>Gives each telephone number a destination type, which determines where the call will be routed<br>Is programmed by area code, exchanges, or individual telephone number<br>**Note**: For the Calling Line ID table to operate, customers must subscribe to these telco Call Display services:<br>Call Line Identification<br>Automatic Number Identification | Speeds up call processing<br>The Automated Attendant can automatically route incoming calls to specific destinations such as a greeting table, mailbox, extension or CCR tree |
| Single Digit Menus (CCR) | Available with Custom Call Routing, allows a caller to select a menu option by pressing a single digit | Improves customer service<br>Ensures customers receive more detailed information very quickly |
| Touch Tone Gate for Auto Attendant/CCR | Allows the system to quickly determine if the caller has DTMF capability and expedite the call if no DTMF is detected<br>Eliminates hold time in areas where rotary phones are common, or where the public network does not provide reliable answer supervision | Decreases hold times<br>Frees up voice channels previously and unnecessarily busied out |
| Transfer Point to an External Number from CCR | Transfers callers to a number outside the BCM system | Ensures callers don't get caught in a CCR loop |
| Transfer (via Feature 986) of an External Caller to a Specific CCR Tree | Directs callers to a specific CCR Tree | Improves customer service<br>Ensures customers reach the correct department in a timely manner |

## Fax Messaging Features

***Table 5-8***

| Feature | Description | Benefits |
|---|---|---|
| Fax | Lets users send and receive faxes | Works with Voice Messaging to provide incoming and outgoing fax capability |
| Fax Messaging | Allows users to store incoming faxes as "fax messages" in their voice mailbox | Improves efficiency<br>Users can view faxes under the same medium as email |
| Fax Answering | Transfers fax calls to a designated fax extension on the BCM via the Automated Attendant | Saves money from having just one main telephone number for both voice and fax calls |
| Fax Overflow | Temporarily stores fax messages if the fax machine cannot answer an incoming call | Ensures users never miss their faxes because a machine has run out of paper |
| Fax On Demand | Allows a user to retrieve documents stored in Voice Messaging in special mailboxes<br>Contains a greeting and all the documents the user has stored | Improves customer service, as businesses can extend their marketing and advertising to their customers easily and smoothly |
| Fax Broadcasting (Group Message) | Allows a user to send a fax to multiple internal or external destinations with through one operation | Improves customer service<br>Improves employee efficiency |

## Group Lists

***Table 5-9***

| Feature | Feature | Benefits |
|---|---|---|
| Group Distribution Lists | Allow users to deliver the same message to a group of users by entering only one address destination or distribution list number<br>*Max. Group Lists: 99*<br>*Max. Members to a Group List: 999* | Saves users time in message preparation, production and delivery |

## Mailbox Features

***Table 5-10***

| Feature | Description | Benefits |
|---|---|---|
| Assigning Target Attendants | Allows mailbox owners to assign an extension as their dial-0 set | Improves customer service, as individuals more familiar with the mailbox owner's schedule answer transferred calls |
| Auto Answer with Personal Greeting | Answers calls with mailbox owner's Personal Greeting after a preset number of rings | Improves customer service<br>Gives callers detailed information regarding the mailbox owner's whereabouts and provides them with options (i.e. leave a message or transfer to a receptionist) |
| Automatic Reply to Internal or External Messages | Enables mailbox owners to automatically reply to a message with one keystroke where CLID and ANI are used | Increases employee efficiency<br>Saves time, as users do not need to look up numbers |
| Broadcast Messages | Enable System Coordinators to record a message and send it to every mailbox<br>Play automatically and are then erased as soon as the subscriber ends the session | Increase employee efficiency<br>Improve internal communications, as every mailbox owner receives the same message |

| Feature | Description | Benefits |
|---|---|---|
| Called Party Cancellation of Off-Site Notification | Allows the party receiving a remote notification call to turn off notification to this destination<br>Is useful when a subscriber enters an incorrect destination telephone number<br>When the called party cancels notification the system removes the number from the subscriber message notification destination list. | Notifies the subscriber when a message has not been sent |
| Cascading Off-Premise Message Notification | Allows mailbox owners to program five internal or external numbers: they can be notified when a message is received in their mailbox.<br>Each number is called in sequence if the preceding number does not answer.<br>Numbers can be designated as a phone, pager or intercom.<br>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 | Improves customer service<br>Users can respond to messages faster<br>Improves external and internal communications, as messages are immediately sent to users wherever they are located |
| Enable or Disable General Delivery Mailbox | Can be disabled or enabled in System Administrator’s Mailbox | |
| Envelope Information | By pressing “7” during or after a message, mailbox owners are notified of its receipt time and date<br>Provides the sender’s name for internal calls | Mailbox owners know exactly when messages were left |

| Feature | Description | Benefits |
|---|---|---|
| Express Internal Messaging | Allows mailbox owners to send internal messages without opening their mailbox<br>Automatically includes the sender’s name and extension | Improves employee efficiency, allowing faster communication |
| Forward Copy with or without Comment | Allows users to forward mailbox messages to other mailboxes with or without comments | Saves time, as users do not have to re-record the message to send it to someone else |
| General Delivery Mailbox | 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<br>When enabled, it allows callers to leave a message. When disabled, allows callers to press zero (“0”) at any time to reach the operator.<br>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. | Improves customer service<br>Ensures calls after hours are always answered by the system and callers can leave a message |
| Guest Mailbox | Is useful for people who do not have an extension number, yet need Voice Mail access<br>Can be provided for a mailbox owner’s favorite customer or supplier | Increases system flexibility<br>Improves communications |
| Information Mailbox | Allows businesses to play frequently requested information only<br>Does not have message-taking capabilities | Eliminates repetition of the same information, such as hours of business, location or the time of performance to multiple callers |

| Feature | Description | Benefits |
|---|---|---|
| Message Delivery Options: Normal, Certified, Private, Urgent | The following four options increase the user's control over message delivery:<br>Normal: the message is delivered automatically (default)<br>Certified: the sender receives confirmation when the message is read<br>Private: messages cannot be forwarded to another mailbox<br>Urgent: a message can be queued to play after broadcast messages, but before "Normal" messages | Increases customer service and employee efficiency<br>Important messages are prioritized and placed at the beginning of the voicemail messages |
| Message Waiting Notification | Displays "Message for You" on the user's set<br>Allows the user to hear the number of new and saved messages upon opening his or her mailbox | Increases employee efficiency, as users do not need to log on to check for new messages |
| Message Waiting Indicator/Visual Ringing Indicator | Red light on the T Series sets indicates that a message is waiting | Improves office productivity<br>Users can answer more calls live and return messages quicker |
| Name Confirmation when Sending | LCD display shows the name and number of the called party or mailbox being contacted | Eliminates delivery errors |
| Name Directory or Extension Accessibility | Allows users to find any system mailbox extension by spelling the called party's last name on the dial pad | Increases productivity<br>Speeds up call processing<br>Improves customer service if the caller only knows the name of the person they are calling and not the extension number |

| Feature | Description | Benefits |
| --- | --- | --- |
| Never Full Mailboxes | Allows external callers to always leave voice messages in a personal mailbox, even if the mailbox is full<br>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<br>Note: The only time an external caller cannot leave a message in a mailbox is when the system is full. | Allows for maximum storage capacity<br>Improves customer access to voicemail users |
| Outbound Transfer from Mailbox | Allows callers to press "7," while listening to a personal greeting, to be transferred to an external number specified by the mailbox owner<br>**Note:** 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. | Provides callers with a means for urgent contact when necessary<br>Improves customer service |
| Personal Greetings Based on CLID | Play to Calling Line ID callers only<br>Allow a mailbox subscriber to program up to three specific telephone numbers, each with its own greeting | Improves customer service |
| Playback Controls | Allow subscribers to move within or between messages without listening to each message entirely | Give users increased control while listening to their messages |
| Personal Mailbox | A mailbox can be assigned to a particular person and extension number for his or her exclusive use | Mailbox owners can receive detailed, confidential messages 24 hours a day |

| Feature | Description | Benefits |
| --- | --- | --- |
| Prerecorded Greetings Storage | Stores up to 40 prerecorded greetings | Saves time, as a system coordinator does not have to record new messages each day<br>Gives businesses a more professional image |
| Primary and Alternate Greetings | Allows mailbox subscribers to switch between prerecorded primary and alternate greetings | Increases flexibility |
| Recovering Deleted Messages | Enables a user to revisit a previously deleted message during a mailbox session and save the message | Users can move quickly through mailbox messages without the risk of accidentally deleting a message that they wish to retain |
| Remote Call Forwarding to Voice Messaging | Allows mailbox owners to turn Call Forwarding to Voice Messaging on or off from a remote location | Improves efficiency<br>Improves customer service<br>If users have forgotten to Call Forward or are unexpectedly away from the office, they can still forward to voicemail so that their customers do not have to wait through multiple rings to be answered |
| Reply Based on CLID | Creates automatic replies to numbers collected from CLASS/CMS or ISDN with ANI<br>Voice Messaging will dial CLID with message by simply pressing the “call” soft key | Improves office productivity<br>Reduces the time users require to return calls |
| Saved Message Queue and Retention Periods | 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 | Mailbox owners can save important messages in case some details are forgotten |
| System Coordinator Mailbox | Enables the system coordinator to send broadcast messages and use the mailbox for administration | Improves system management<br>Ensures calls are not lost |
| Urgent Message Notification | Displays an urgent message with the prompt “This message is urgent”<br>Moves urgent messages ahead of non-urgent messages to the front of the new message queue | Improves time management<br>Mailbox owners can deal with important messages first |

## Miscellaneous Features

***Table 5-11***

| Feature | Description | Benefits |
|---|---|---|
| Call Screening per Set | Requires the caller to enter his or her name<br>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 Voice Mail<br>Is particularly useful where Calling Line Identification (CLID) information is not available, or when the called party has a set without display capabilities<br>Note: This feature is enabled on an individual mailbox basis from Mailbox Administration | Improves productivity<br>Users can choose whether or not to interrupt work to take a call |
| Enable or Disable the Company Directory | Allows the System Coordinator to enable or disable access to the Company Directory for internal and external users | |
| Enable or Disable Voice Messaging Feature | Allows a System Manager to globally enable or disable the Voice Messaging feature<br>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 | |
| External Volume Control | Allows users listening to messages from outside the company to increase the playback volume by pressing the star key ("*") | Eliminates message distortion<br>Ensures users will hear details correctly |
| Interrupt Feature | Allows mailbox users to retrieve calls that have been forwarded to voicemail. While a caller is leaving a message in the user's mailbox, the user can interrupt the message and talk to the caller. | Increases customer service<br>Increases employee efficiency |

| Feature | Description | Benefits |
|---|---|---|
| Multiple Recipients Per Message | Allows users to send messages to multiple recipients with one set of delivery options applied to all recipients<br>Lets users choose to add recipients or delivery options in any order prior to sending the message | Saves time when sending messages to multiple recipients |
| Record a Call | While on a call, the user can activate record a call by dialing Feature 989. The CallPilot will then record the call until the call is disconnected or the user stops recording by dialing Feature 989 a second time.<br>When this feature is activated, both parties will hear “this call is being recorded.”<br>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. | Increases customer and employee efficiency |
| Semi-interruptible Greeting (Extended Absence Greeting) | Allows mailbox owners to inform callers of an extended absence<br>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<br>Follows a special tone that alerts callers that it is in effect | Increases customer satisfaction, as customers can receive up-to-date information about absences |
| Single Button Call Forward to Voice Messaging | 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. | Increases employee efficiency |

| Feature | Description | Benefits |
|---|---|---|
| Single Trunk External Link Transfer | Transfers callers externally out of voicemail without tying up two trunks for the duration of the call | For businesses that use centralized answering for multiple locations, calls can be routed to the appropriate location for products and services, without tying up two trunks to complete the transaction |
| Timed Delivery of Messages | Allows a subscriber to create a message and delay delivery of that message until after a specified date and time<br>Delays message delivery up to the number of days specified in the message retention class of service parameter for a given mailbox<br>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). | Increases flexibility |
| Trunk Answer On/Off | Permits System Coordinators to turn on or off incoming trunk lines programmed for answering by the Auto Attendant | Enhances Voice Mail system control |
| Voice Messaging Option | May be enabled or disabled at any time<br>The default status is enabled<br>When enabled, allows callers to access all mailboxes<br>Transfers callers who reach a busy extension to the extension’s mailbox<br>When disabled, does not allow callers to leave messages in any mailbox unless manually transferred to a mailbox; callers can access Information Mailboxes | When callers hear that the called party is not available, they are transferred to the Automated Attendant voice prompt. At any time, callers can press zero (0) to reach the operator |

## Reports

***Table 5-12***

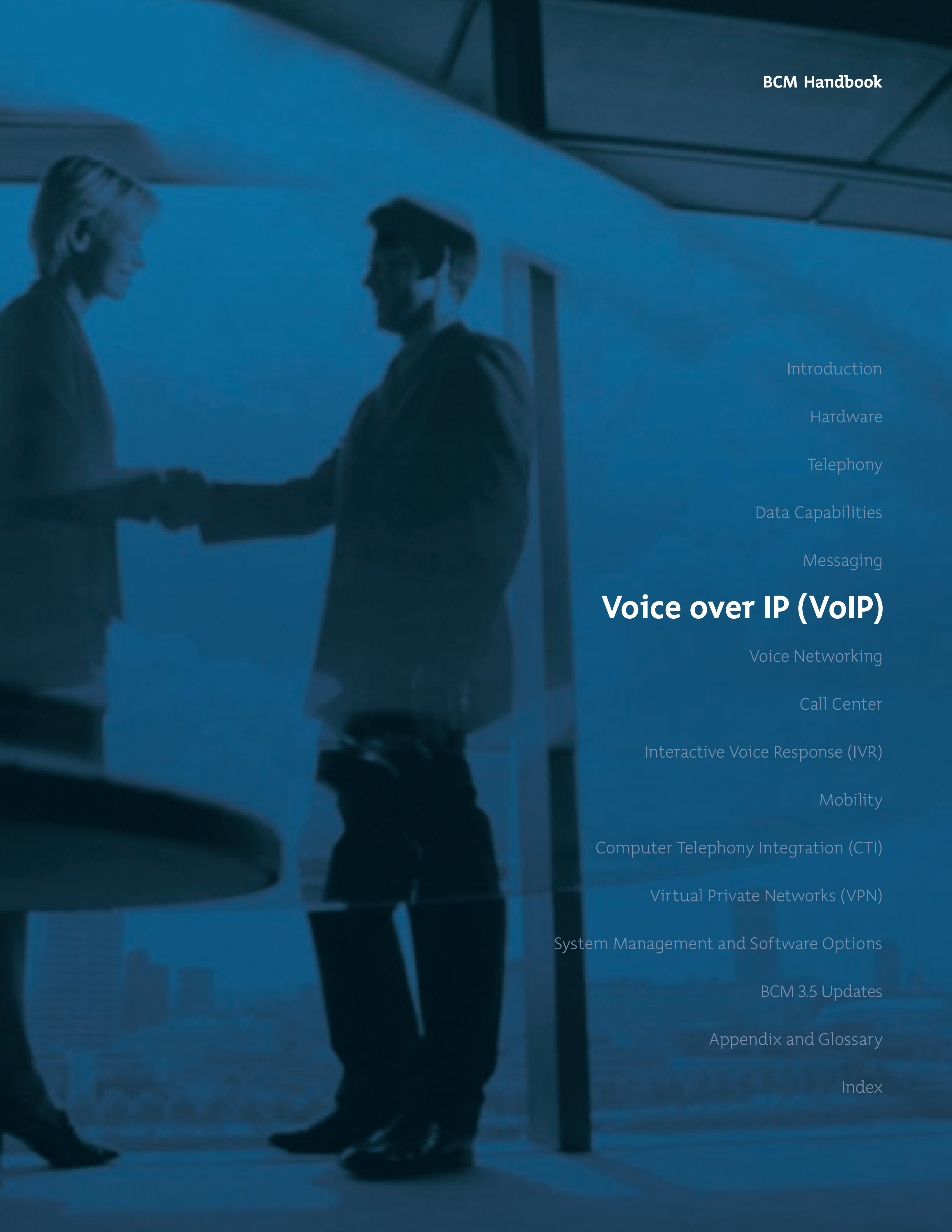
| Feature | Description | Benefits |
|---|---|---|
| Call Handling and Channel Usage Report | Provides traffic statistics on the types of calls handled and the traffic against each port used by BCM | |
| CCR Usage Report | Provides the greeting table from which the CCR tree is currently referenced and a seven-day rolling count of the number of calls received by the tree and the number of times each path is visited | Provides visibility to the CCR tree and the opportunity to fine-tune it as required |
| Numeric Mailbox Information Report | Previously known as the Numeric Subscriber Report<br>Includes more information about the mailbox | Helps to identify potential usage problems and reports on all mailbox types, not just subscriber mailboxes |
| Fax On Demand Usage Report | Lists all the Fax On Demand requests and shows the date and time, item requested, delivery fax number and caller CLID | Provides visibility to in the usage of Fax On Demand and the opportunity to modify it as required |
| System Configuration Report | 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 | |
| Report Generation | Various reports can be printed at the request of the system coordinator | Helps to monitor system programming and simplify updates and customization |

## Security Features

***Table 5-13***

| Feature | Description | Benefits |
|---|---|---|
| Change of Operator Password | A password can be changed from default "OPERATOR" (67372867) to any four to eight-digit sequence | Greater flexibility for the administrator |
| Centrex Transfer Feature Restrictions | 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.<br>The Centrex Transfer restrictions prevent this by specifying dialing sequences that will be denied. | Greater security and cost savings |
| Forced Password Change | 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. | Increased security |
| Double Entry of New Passwords | Prompts mailbox owners to enter a new password twice when changing their password | Increases password accuracy and security |

| Feature | Description | Benefits |
|---|---|---|
| Incorrect Password Detection and Lockout | Tracks, through the Voice Messaging system, the number of incorrect login attempts since the last successful login<br>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. | Provides additional security |
| Internal Set Initialization | 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 | Provides additional security |
| Outbound Calls Restricted to Preset Line/Pool | Allows the administrator to specify which line or pool is to be used for Voice Messaging outgoing calls | Helps to control or prohibit toll fraud<br>Contributes to cost reduction |
| Set Based Restrictions for Outbound Calls<br>CallPilot | Outgoing calls initiated by CallPilot Messaging are subject to set based restrictions, regardless of the line pool selected as the outgoing facility | Cost savings to the business |







# Voice over IP (VoIP)

# Voice over IP (VoIP)

## Chapter Highlights

- VoIP solutions – maximize corporations' profitability by reducing network costs and increasing employee productivity while delivering the unsurpassed reliability our customers have come to expect from traditional voice solutions.
- BCM VoIP – is a keycode-enabled option that supports IP trunks to communicate with other sites. It can also support IP stations such as the i2004, i2002, i2050 and eMobility wireless VoIP solutions.
- Quality of Service (QoS) – is a measure of the caliber of the service provided across the network and is extremely important in a network that handles continuous transmission of real-time sensitive traffic, such as voice and video multimedia traffic.
- Speech Compression – is a technique used to reduce the amount of bandwidth required to transmit voice packets across a network.
- Silence Compression (Suppression) – reduces the amount of bandwidth required to transmit voice packets across a network by not sending packets during times when the caller is not talking.
- VoIP in Greenfield Installations – eliminates multiple drops to the desktop by eliminating the need for the twisted pair drop.
- Real Savings – are easily tracked with VoIP as a fixed expense vs. the variable expenses that are associated with long-distance charges. This represents a large management benefit for the IT manager.

# Voice over IP Sales Scenario

## The Challenge

ClearChannel Communications is a fully integrated marketing communications firm specializing in helping clients build their business through a broad range of services, including communications planning, advertising and communication arts, public relations, database and direct marketing, interactive media, information engineering and technical documentation. Headquartered in Mississauga, Ontario, ClearChannel has just opened a new branch office in Dallas, Texas. Currently, ClearChannel has approximately 100 employees working at head office, and 25 at its branch office.

Moving ahead with the growth of Internet business, and the expansion of its own business, ClearChannel looked to streamline its communications system. The company had to consider a solution that would allow its offices to stay in touch without the worry of spiraling costs. As both offices are in constant communication, ClearChannel required a solution that would IP-enable its smaller branch office, giving it the same capabilities and features as the head office. The solution had to be economical, ready for future features, simple to use and integrate with head office's existing Meridian 1. In addition, ClearChannel wanted to minimize training and make it easy for employees who travel between locations by ensuring that the new phone system was as consistent as possible with the main phone switch. Added to these factors, the ever-expanding demands of a multimedia environment required ClearChannel to seek better ways to handle its bandwidth.

## The Nortel Networks Solution

ClearChannel teamed with Nortel Networks to find a networking solution that would grow along with the business. The solution was to add a VoIP enabled Meridian 1 PBX at the central site, with the Nortel Networks Meridian 1 Option 11C installed to support VoIP. To provide telephony services over the IP network, ClearChannel installed a BCM at the Dallas office. To ensure minimal latency over the entire network, ClearChannel also installed Nortel Networks BayStack 460 to ensure the Quality of Service (QoS) levels it required for clear voice transmissions between locations.

ClearChannel has also implemented i2004 Internet Telephones; doing so saved the company from having to recable the office to replace its previous problematic telephone cabling. The i2004 Internet Telephone sets deliver traditional business communications functionality and value, in a standards-based, open platform. Joining Nortel Networks full range of desktop devices that includes fully featured traditional digital telephones, IP phones and i2050 Soft Clients running on a PC, these sets are designed to meet diverse customer needs and enrich people-to-people communications in unified networks.

Because Internet Telephones are directly connected to the LAN, they allow businesses to capitalize on the economies of a simplified wiring system within the enterprise. Users are provided the features and services they require and the enterprise can balance needs for user functionality with benefits of streamlined management and reduced facilities costs. Now, the Dallas office is ready as ClearChannel looks at deploying and integrating IP telephony between offices.

## Implementation

The VoIP gateway is treated as a completely integrated trunk by the core telephony service within the BCM – utilizing IP telephony is, therefore, completely transparent. The i2004 Internet Telephones are supported on multiple Nortel Networks platforms, including carrier-hosted and customer premise equipment.

With the use of DiffServ methodology, voice traffic is prioritized over data traffic. All voice packets are tagged with a distinct identifier (DiffServ Code Point, or DSCP), allowing DiffServ-enabled elements in the network (i.e. BCM QoS Routing platform and other routers, including BayStack products and the Business Policy Switch) to recognize them. This ensures that those packets receive priority treatment (i.e. they get sent to the head of the line) as they cross ingress or egress points in the network to minimize the delay that is introduced by moving into slower links.

## Business Impact

ClearChannel has realized substantial cost savings through voice enabling its network. The IP phonesets have saved costs and the branch office is now ready for the future. Most importantly, ClearChannel's customers appreciate the system because of its reliability, ease of use and ability to track down the right person quickly.

The value to ClearChannel of improved communication capabilities has been significant. Because VoIP permits the consolidation of separate voice and data networks into a single, unified path for all communications needs, ClearChannel Communications is able to realize lower training and personnel costs and it can easily manage the system remotely through a single interface. The centralized control and simple management make network administration dependable. Additionally, the routing capabilities deliver enhanced security.

## Overview

Simply put, 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 i2004, i2002, i2050 and emobility wireless VoIP solutions.

Subscribers can activate a maximum of 60 simultaneous VoIP trunk gateway sessions on the BCM with the purchase of software keycodes and two additional PEC III DSP processors. 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, 32 and 64. The BCM can support as many as 90 IP stations, depending on the other capabilities planned to be configured.

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.

## Emerging Trends

Businesses of all shapes and sizes are showing tremendous interest in Voice over IP solutions. Industry analysts report that over 50% of all enterprises in North America view VoIP as *the* next generation communications solution. Moreover, these analysts predict that this market will increase significantly in the next few years. Businesses are already realizing the benefits of Voice over IP, and the adoption rate is expected to continue to grow at 26% per year.

VoIP is becoming an increasingly important option. By carrying voice and data over a single Internet connection, VoIP eliminates long-distance toll charges and the hardware necessary for standard PSTN calls. Employees can maximize their company's return on investment in VoIP simply by using it as the primary means of internal communication (e.g. phone conversations, videoconferencing ). Moreover, new VoIP applications and services will provide customers with more choices and greater flexibility, which, in turn, can generate more business and profits for companies. Companies that use VoIP will likely experience greater internal efficiency and also better customer relations.

## VoIP Benefits

BCM provides the ultimate in choice and flexibility, allowing customers to cost effectively migrate to IP and add applications that level the playing field as their business needs evolve.

VoIP solutions from Nortel Networks maximize corporations' profitability by reducing network costs and increasing employee productivity, while delivering the unsurpassed reliability our customers have come to expect from traditional voice solutions.

VoIP solutions enable businesses of all types and sizes to:

- **Innovate** – Businesses are in the midst of changing the way they get and share information, relate to their customers, suppliers and other players, place and accept orders for goods and services and develop and maintain relationships with their marketplace. Voice over IP provides the springboard for the innovations these businesses must make to compete and thrive in the future.
- **Save money** – Voice over IP permits the consolidation of separate voice and data networks into a single, unified path for all communications needs. Consolidation means network efficiency, lower training and personnel costs and remote management through a single interface. It also means flexible and rapid applications deployment at significantly lower costs.
- **Make money** – Voice over IP provides the vital link between a business' ecommerce and customer service objectives by providing a seamless link to products and services, regardless of whether they are phone, email, fax or Web.

# 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 Meridian 1-ITG (Release 3.0) capability
- ITU-H.323v2 compatible gateway
- ITU-H.323v2 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 MERIDIAN 1 ITG and CSE1K.

## VoIP Features

The BCM VoIP Gateway supports ITU-H.323 v2 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.

## VoIP Gateway

The VoIP gateway:

- Allows communication with other supported H.323 v2 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
- 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. 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.
- 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.

## 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 6-1*

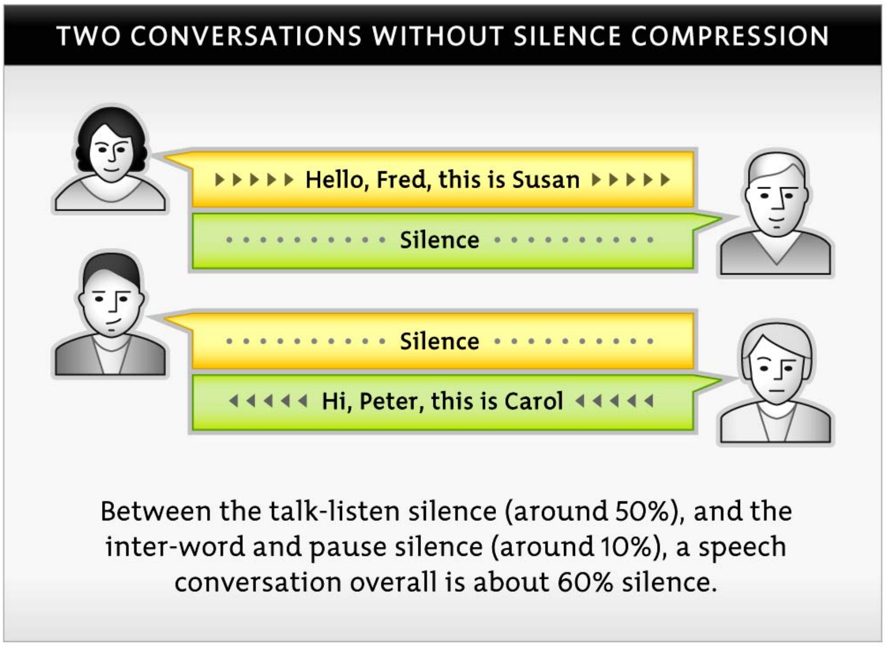
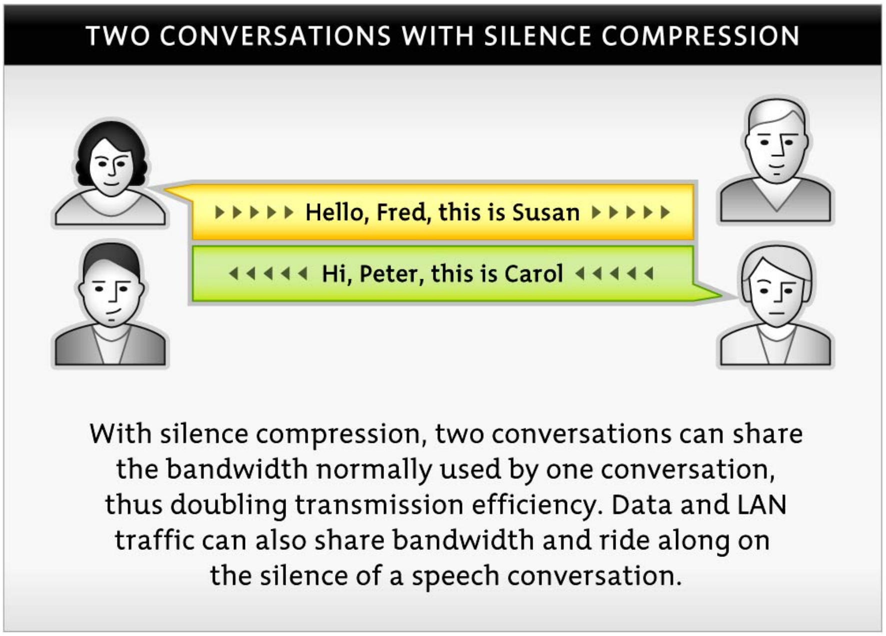



*Figure 6-2*



## 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

BCM introduces the support of IP stations. The IP station portfolio consists of the i2004, the i2002 and the i2050 software-based phone, in addition to other H.323 client devices such as the wireless Voice over IP handset.

The i2004 and i2002 support paging through the set and handsfree intercom. Since the i2050 is a “software phone,” it has no set or speaker and cannot receive pages or hands-free intercom calls. The i2050, however, can originate paging and handsfree intercom calls.

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

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.

## Media Path Management on LAN

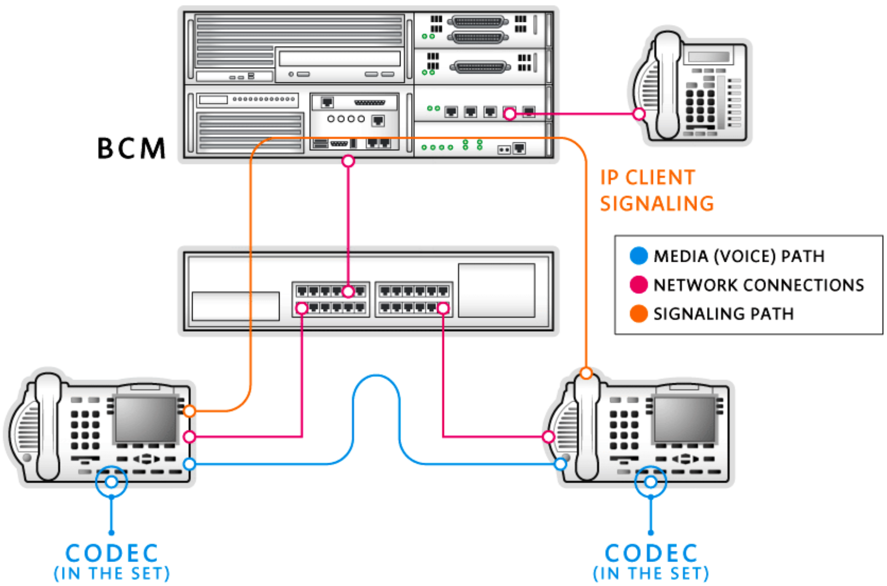
BCM has the ability to handle management between two IP devices, without being converted to TDM and back to IP. During call setup, an IP station such as an i2004 communicates with the BCM Call Server via a signaling protocol called Unistim to determine the destination of the dialed number. If the call 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

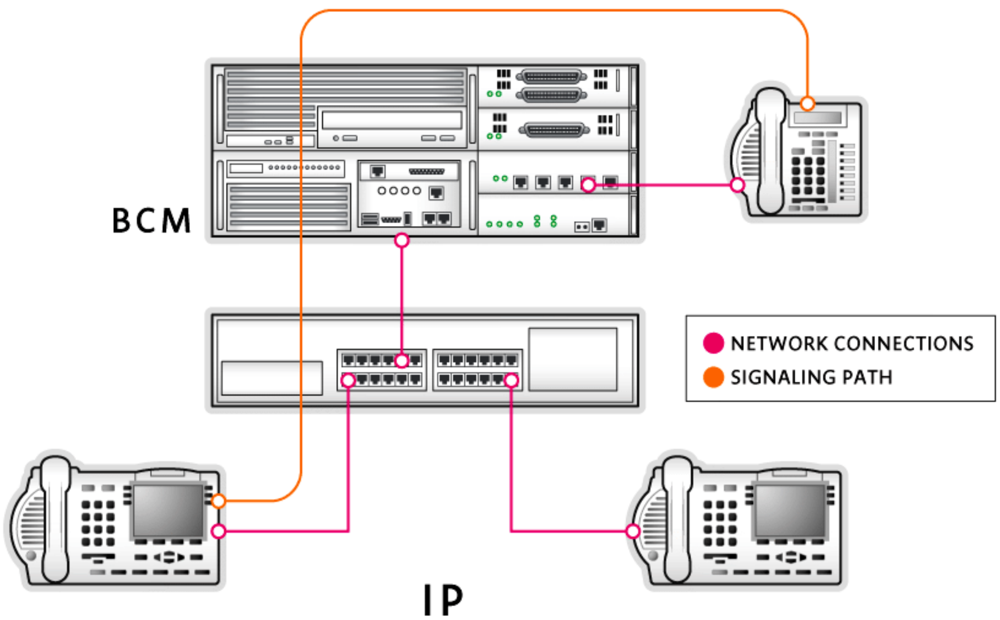
***Figure 6-3***



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/LAN

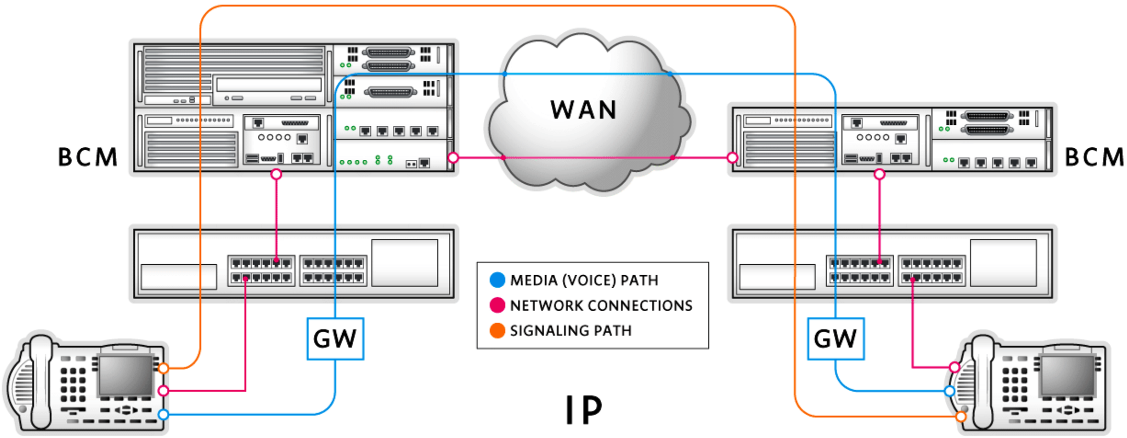
***Figure 6-4***




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.

## Media Path Management/WAN

***Figure 6-5***

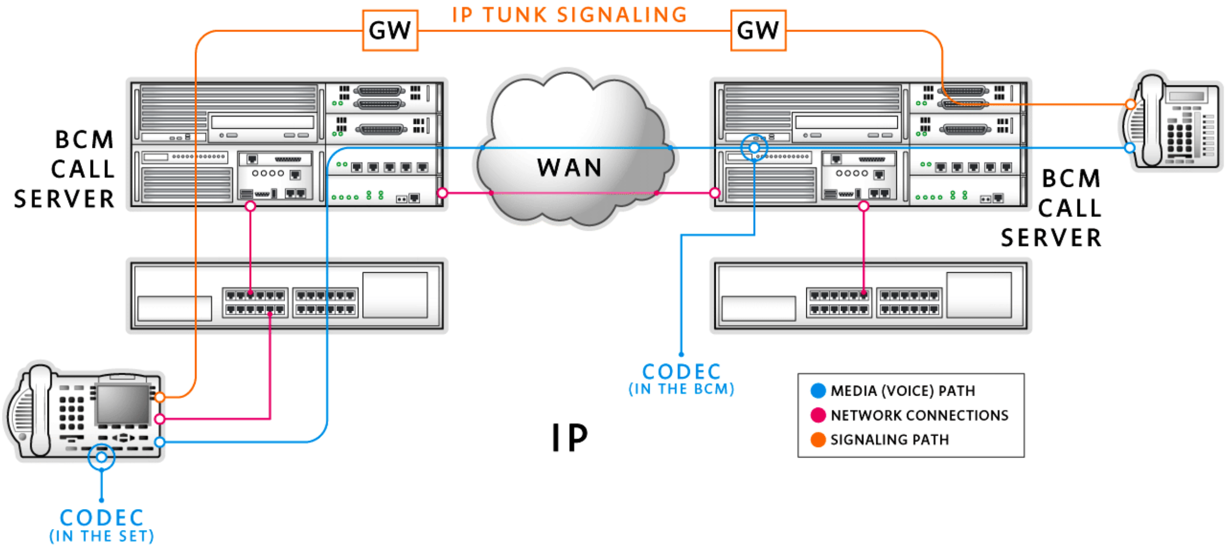



Once again, this is important, as many other solutions convert calls from IP to TDM and back to IP, sometimes several times. Each time voice quality degrades. BCM establishes the call entirely via IP with no degradation or delays. But there are some situations where BCM does convert the call from IP 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 6-6***

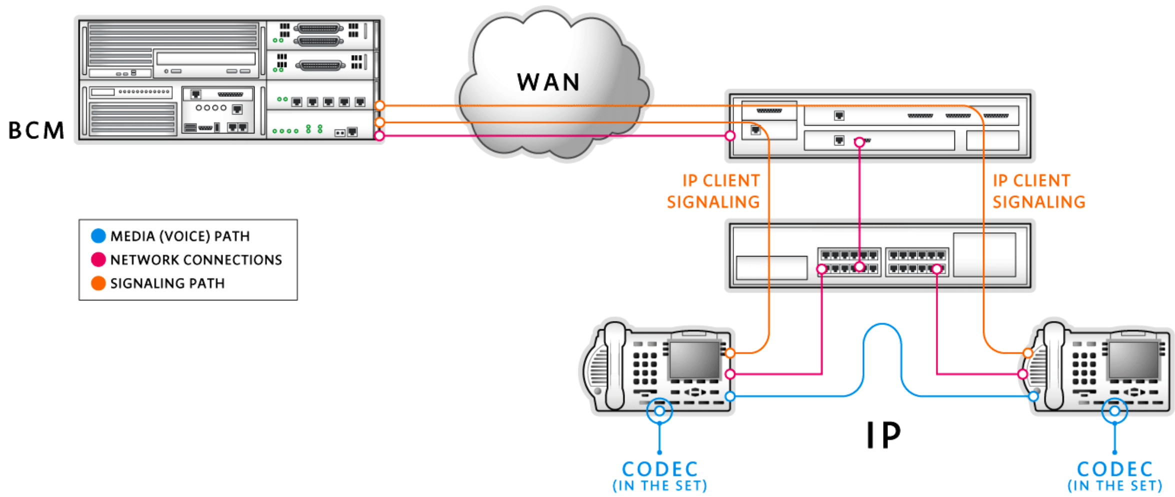



Once again, this is important, as many other solutions convert IP to TDM and back to IP, sometimes several times. Each time there is degradation of voice quality.

## 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.

*Figure 6-7*


BCM
WAN
IP CLIENT SIGNALING
IP CLIENT SIGNALING
MEDIA (VOICE) PATH
NETWORK CONNECTIONS
SIGNALING PATH
IP
CODEC
(IN THE SET)
CODEC
(IN THE SET)





# Call Center





# Voice Networking

# Voice Networking

## Chapter Highlights

- Meridian Customer Defined Networking (MCDN) – is a Nortel Networks proprietary ISDN signaling protocol used to interface a BCM to another BCM, a Meridian 1 system or a Norstar system.
- Basic Rate Interface (BRI) Lines – provide incoming and outgoing access to an ISDN network.
- Digital Private Network Signaling System (DPNSS) – offers significant enhancements to BCM networking capabilities, making it easier to support centralized functionality within private networks.
- Coordinated Dialing Plan – allows a user to reach any set in the private network by dialing an extension; the advantage to this plan is that any node in the network appears to the caller to be on the same node as him or her from a dialing perspective.
- Universal Dialing Plan – allows a user to reach sets on the user's own node by dialing an extension and enables a user to reach any set in the rest of the private network by dialing an access code plus a location code, plus the extension.
- Remote System Access to BCM – allows callers elsewhere on the private or the public network to access BCM by dialing directly without going through an attendant.

## Voice Networking Sales Scenario

### The Challenge

Smith & Co. is a large department store chain, headquartered in Toronto, with over 1000 employees and several stores located across the country. Employees from each location are in constant communication – updating inventories and checking pricing, sales, cash register and other information. As a result, network access fees and long-distance charges incurred by site-to-site communications make up a large proportion of Smith & Co.'s expenditures. Using leased T-1 lines to maintain PRI links between remote sites and the head office, Smith & Co.'s always-on connectivity solution was increasing costs. In addition, with sales employees conducting business on the road, dial-in costs were adding up.

Realizing that there was enough traffic between sites to cost-justify having dedicated lines, Smith & Co. decided to look for a new solution. It wanted to implement a cost-effective dialing plan and find a flexible, future-proof communications solution that would not only address its current infrastructure concerns, but would also grow with its future telecommunications needs.

## The Nortel Networks Solution

VoIP trunking with Internet Telephony Gateway (ITG)-equipped Meridian 1 systems enable economical basic private voice networking between sites and at lower trunk capacities than are required when using PRI connections. Using excess bandwidth on their private data networks to carry voice traffic enables businesses to maximize their bandwidth utilization, allowing them to get the most for their money. It also allows them to bypass the significant long-distance charges they might accrue when communicating with remote locations.

By combining both voice and data on the same T-1, businesses can realize substantial monthly savings. PRI lines provide faster transmission speeds and the addition of a variety of powerful business applications, including remote LAN access, videoconferencing, file transfer and Internet access.

With the Coordinated and Universal dialing plans offered by BCM, dialing between sites is simple and cost-effective. Coordinated dialing plans are networkwide, allowing all telephone numbers to be uniform in length. Universal dialing plans also offer specific benefits, including shorter extension numbers.

The remote access feature offered by BCM allows mobile users to dial in to the system directly, without having to go through an attendant. With remote system access, users are no longer tied to their desks.

## Implementation

As BCM looks and feels very much like Meridian 1, it is easy to integrate small BCM sites into a large corporate voice network. All users interact with the system and use the same familiar menus and tools.

The number of extensions a business currently has drives the decision to implement either a four or five-digit dialing plan. Given Smith & Co.'s specific needs, the company implemented a four-digit dialing plan using the built-in Voice over IP gateway, router and key.

The most important benefit of a voice networking solution is consistency at each site. All voice functions are standard at each business location.

### Business Impact

Now, with the Nortel Networks voice networking solution in place, Smith & Co. can transmit data seamlessly from one location to another. File sharing, email, voicemail and four-digit store-to-store dialing are saving time and improving efficiency, both in-house and on the road.

The company has realized substantial cost savings as a result of the reduced toll charges associated with dedicated lines. Previously, Smith & Co.'s site-to-site communications had accounted for 90% of their long-distance charges.

The company's mobile sales force has also benefited significantly from the remote system access capabilities offered by BCM. Now, they can access the system's resources while away from the office, saving both time and money.

## Overview

In addition to public network connections, BCM can be integrated into an existing private network of BCMs, Norstars and Meridian 1s to form a corporate telecommunications network.

BCM uses enhanced signaling on certain trunk types to join Nortel Networks 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.

## Emerging Trends

While data traffic has been growing over the last few years, voice continues to be an important component of enterprise networks. Voice and telephony still account for the largest volume in today's telecommunications networks, in terms of both traffic and generated revenue. But, as conventional time division multiplexing (TDM), or circuit-switched, networks can be expensive, businesses today are increasingly looking to converged voice and data networks that will allow them to increase their operating efficiencies. The benefits offered by these converged networks allow businesses to cultivate their existing customer base as well as penetrate new accounts. The long-term benefits of convergence go to a company's bottom line while making its Internet infrastructure future-proof and positioned for next-generation applications.

As businesses implement converged voice/data networks and evaluate their infrastructure, they require switches with Quality of Service (QoS) capabilities. QoS is perhaps the most important issue network managers face, with the deployment of converged networks. The radically different nature of voice and data traffic place different demands on the network it is transmitted across. Data traffic requires a significant amount of bandwidth and is tolerant of delays and network latency. Voice traffic requires only modest bandwidth but is very sensitive to network delays and latency. To ensure the highest quality for voice services on the network, service providers must work on developing QoS, as this issue is of paramount importance for businesses looking to converged communications solutions.

## Benefits

The Meridian Customer Defined Networking (MCDN)-based voice networking capabilities of BCM provide a cost-effective branch office solution for corporate networks, particularly those that have other Nortel Networks products such as Meridian 1, CallPilot and Meridian Mail. MCDN via Primary Rate Interface (PRI) provides the opportunity to lower the customer network cost of ownership by supporting centralized voicemail, centralized attendant and centralized trunking to the Meridian 1 Headquarters location. VoIP trunking with Internet Telephony Gateway (ITG) equipped Meridian 1 systems enables basic private voice networking between corporate offices to be economical at lower trunk capacities than would be required when using PRI connections. With the MCDN over IP trunks capability, the centralized functions also become more economical at small BCM equipped sites.

## Meridian Customer Defined Networking (MCDN)

Meridian Customer Defined Networking (MCDN) is a Nortel Networks proprietary ISDN-PRI signaling protocol used to interface a BCM to another BCM, a Meridian 1 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 Meridian 1 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 MBs 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, introduced with BCM, 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-1Trunks 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 kbits/s and a data D-channel that operates at 16 kbits/s. 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.

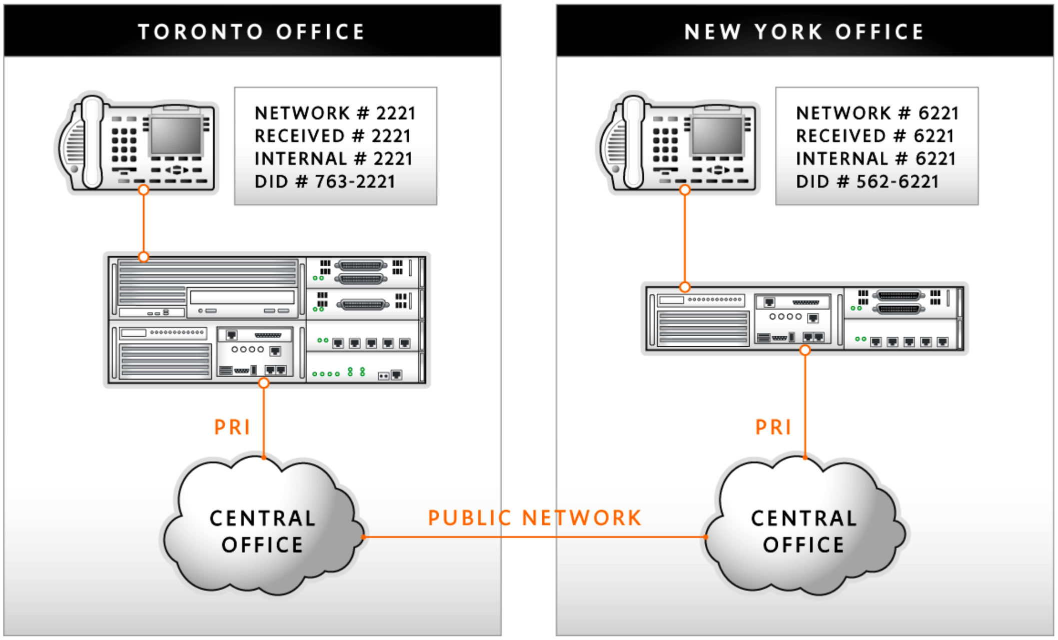
BCM supports remote system access on the following trunk types:

- 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.

***Figure 7-1***




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 Meridian 1 Networking

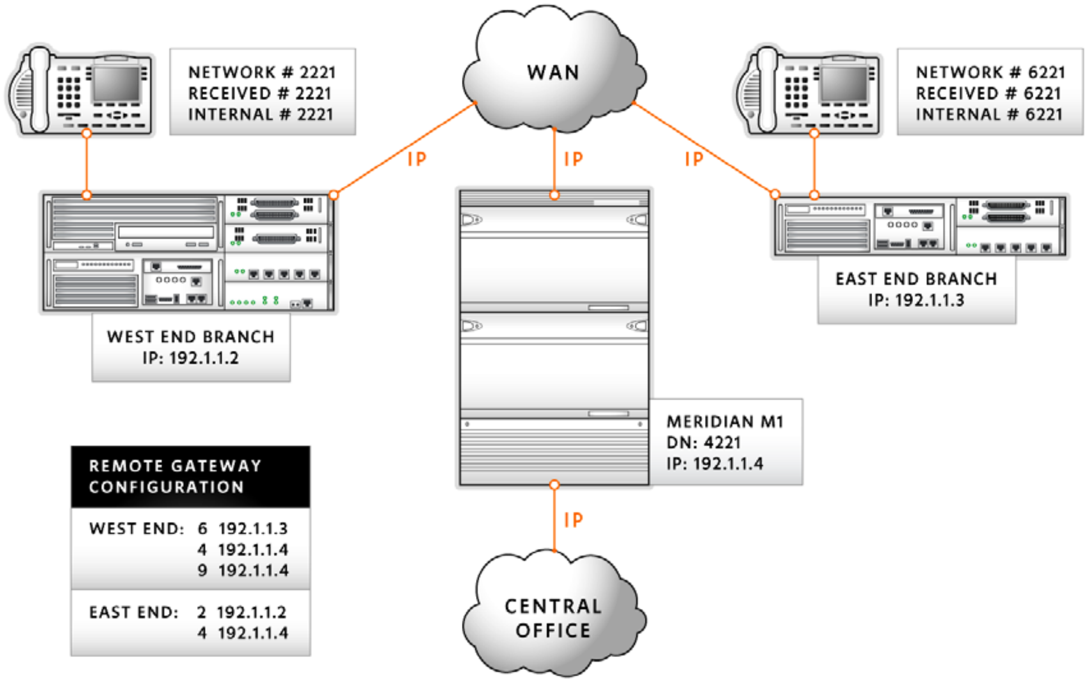
Figure 6-2 shows a private network composed of one central Meridian 1 and two smaller sites, with BCM connected over IP trunks through a corporate IP network.

The network could represent a large head office (with the Meridian 1) 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 Meridian 1. Routing tables at the Meridian 1 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, if a user in the west-end branch wishes to call a user in the east-end branch within the private network, he or she dials 6221.

***Figure 7-2***




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

# 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 Meridian 1 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.

## Toll Bypass with VoIP Gateway

Figure 6-3 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.

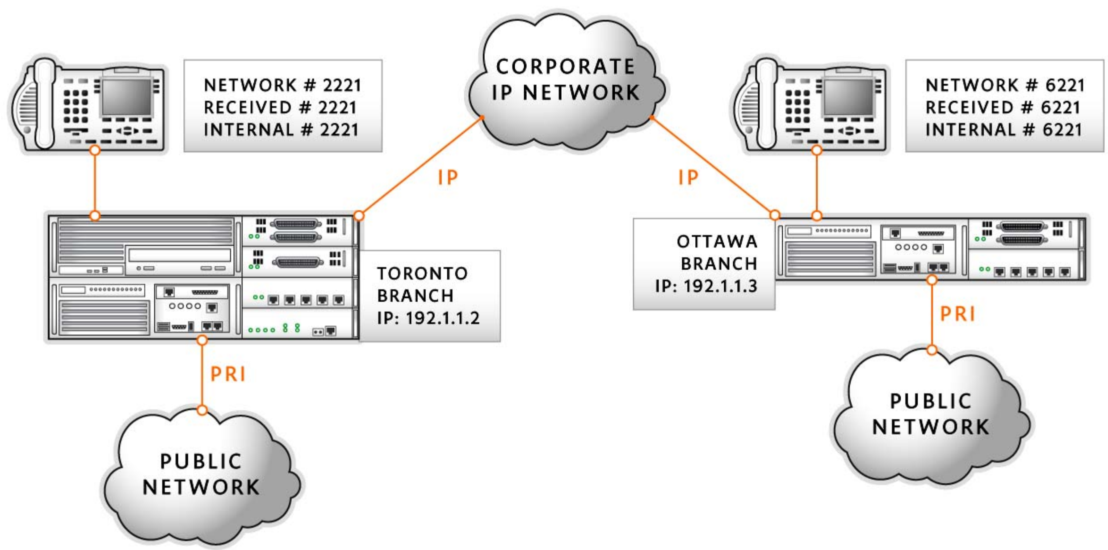
When a call gets rerouted over the PSTN due to congestion, the user will hear a tone and see a prompt indicating that the line has been redirected. The warning indicates that toll charges may be applied to this call.

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 7-3*




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 Meridian 1 networking on either PRI or IP trunks by taking advantage of centralized messaging, centralized trunking and centralized attendant.

Table 7-1 shows features of MCDN and their primary purposes:

***Table 7-1***

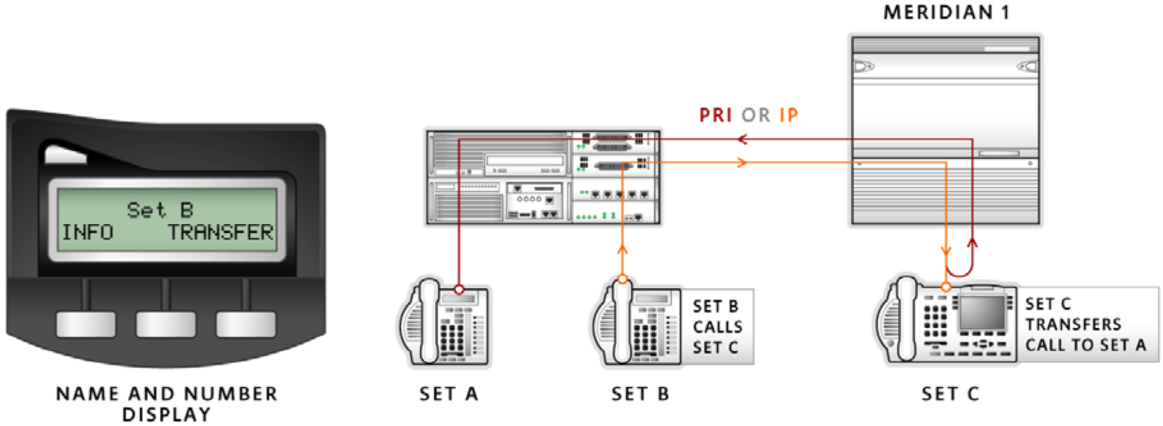
| Features | Purpose |
|---|---|
| NCT/NCRI/MWI | Centralized Messaging |
| ICCL/TRO/TAT | Centralized Trunking |
| Camp-On/Break-In | Centralized Attendant |

### Network Call Transfer

Network Call Transfer allows the transferee 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 7-4 shows CLID on transferred calls.

***Figure 7-4***

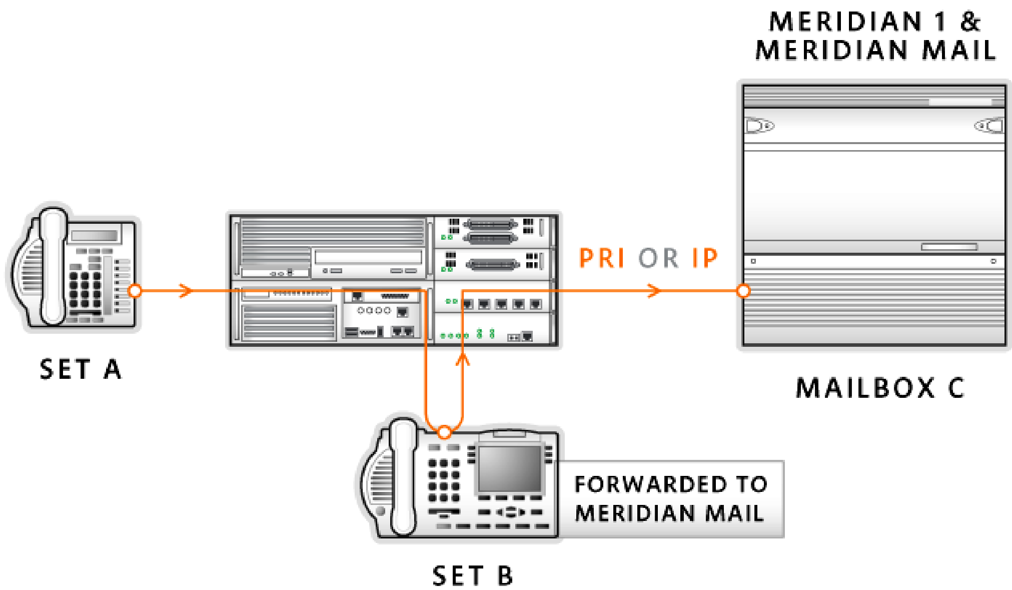


### Network Call Redirection Information (NCRI)

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 voicemail point.

Figure 7-5 shows Set B using NCRI designating the correct mailbox.

***Figure 7-5***




### Message Waiting Indication (MWI)

The Message Waiting Indication (MWI) feature complements centralized voicemail 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 voicemail system. This capability is supported between BCM and Meridian 1 systems only and is not supported between BCM systems.

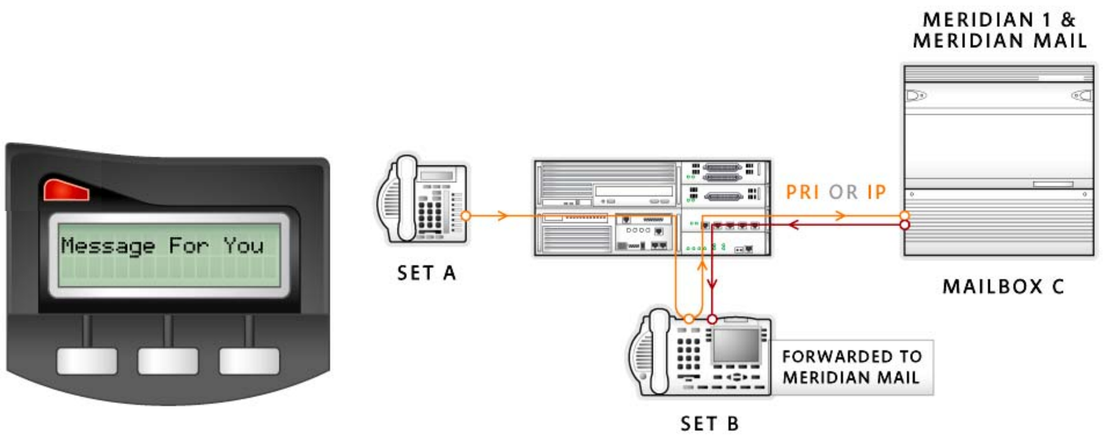
In previous releases of BCM, the Business Series Terminals and the Norstar digital telephones (the M7000 series) informed the user that there was a voicemail message waiting by displaying the “Message for You” prompt on the LCD display.

The BSTs, T7316, 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 7-6 shows the MWI "lights" remote site message waiting lamp.

***Figure 7-6***



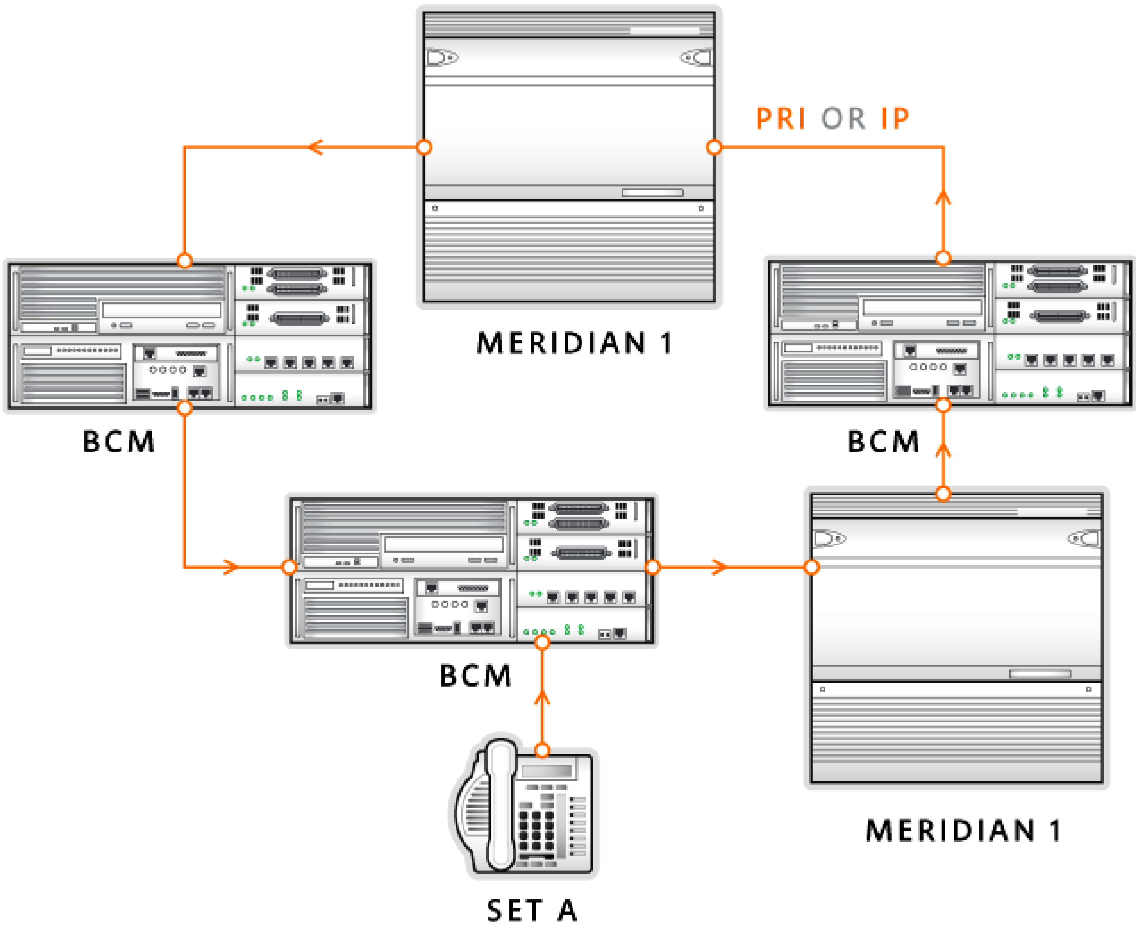

## ISDN Call Connection Limitation (ICCL)

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 7-7 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.

*Figure 7-7*



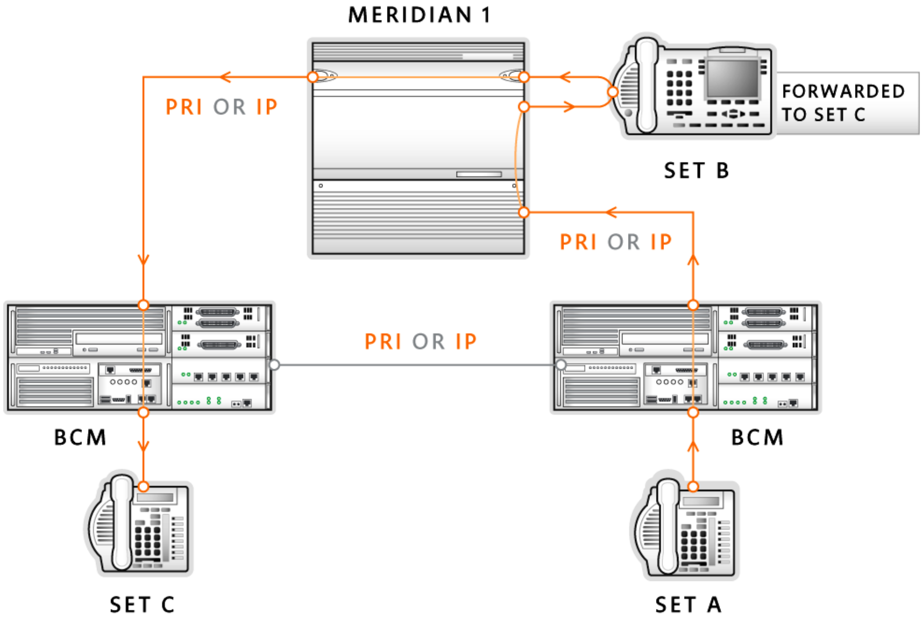

**Trunk Route Optimization (TRO)**

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.

Figures 7-8 and 7-9 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 Meridian 1 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.

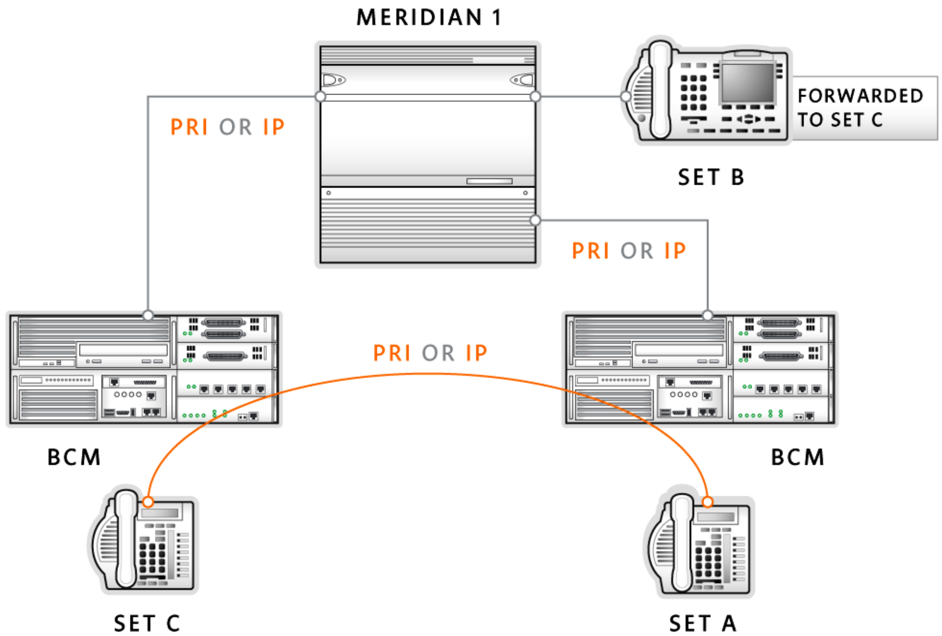
***Without TRO (2 PRI/IP Links)***

***Figure 7-8***




***With TRO (1PRI/IP Link)***

***Figure 7-9***



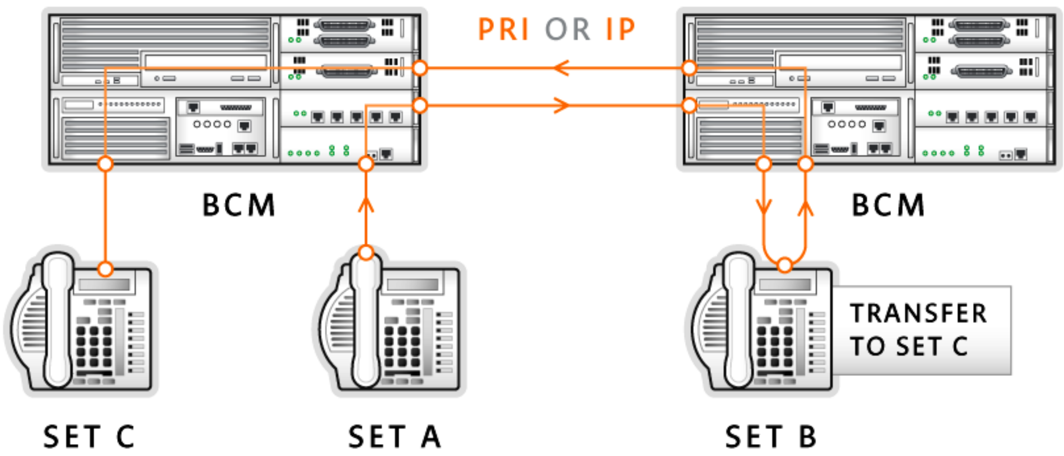
### Trunk Anti-Tromboning (TAT)

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.

Figures 7-10 and 7-11 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.

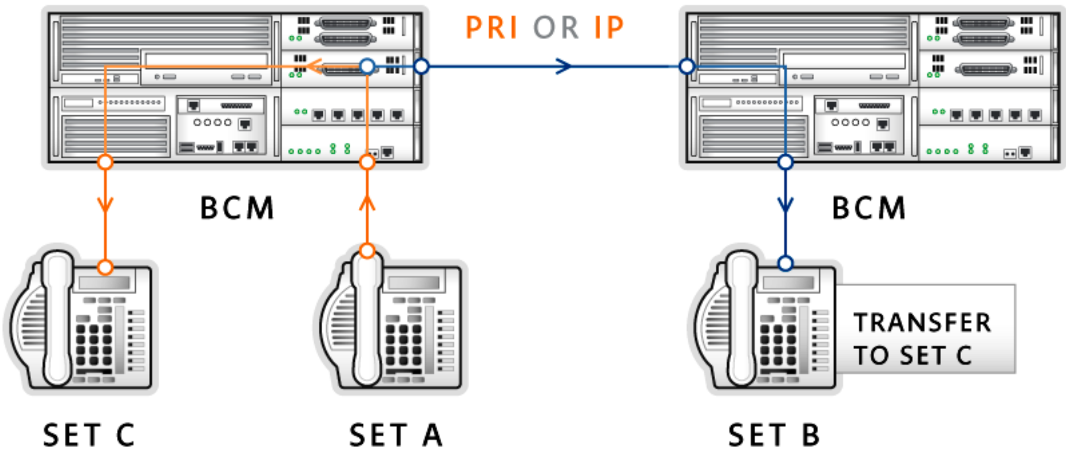
***Without TAT (2PRI/IP Channels)***

***Figure 7-10***



***With TAT (1 PRI/IP Channels)***

***Figure 7-11***

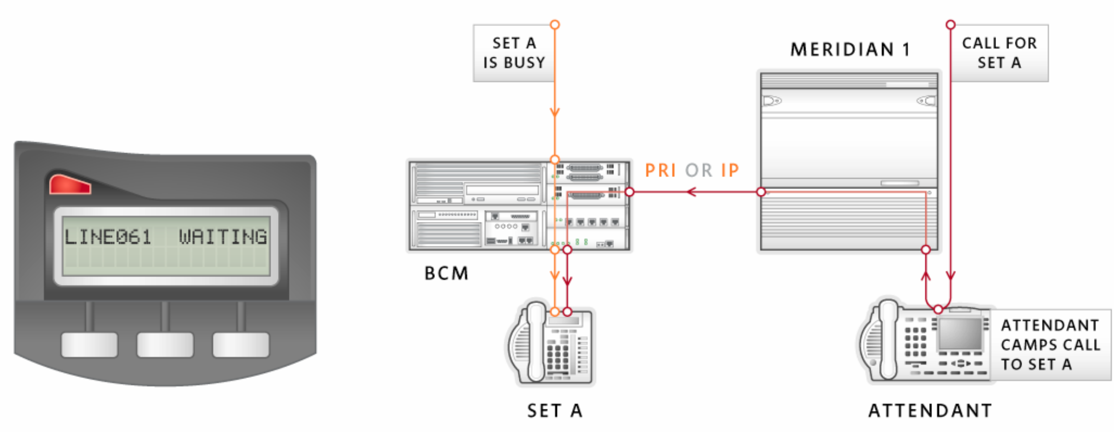



## Camp-On

Camp-On allows an attendant on the Meridian 1 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 7-12 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 Meridian 1 activates the Camp-On feature against Set A at the BCM.

***Figure 7-12***

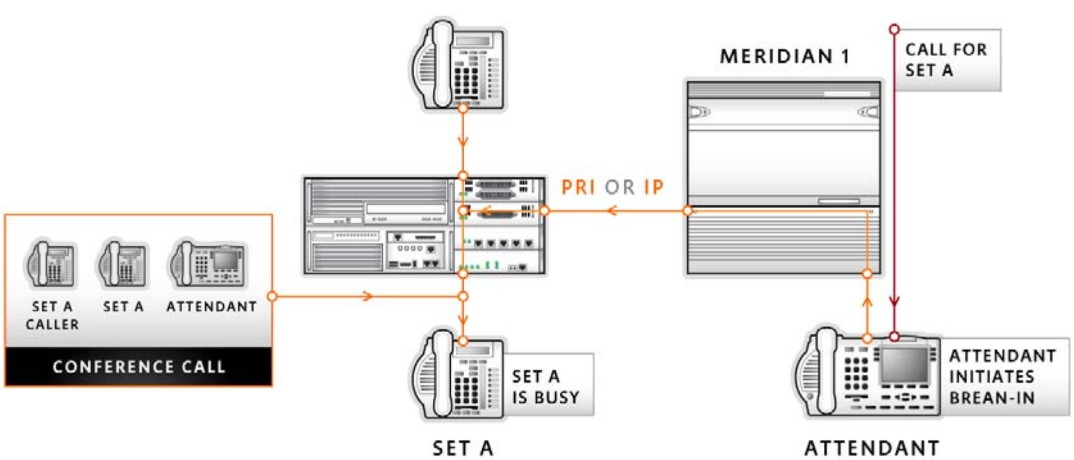



## Break-In

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 7-13 illustrates the call flow when the attendant at the Meridian 1 uses the Break-In functionality on an active call at the BCM.

***Figure 7-13***





# Call Center

# Call Center

## Chapter Highlights

- BCM 3.0 Call Center Applications– can be leveraged in a converged VoIP network to help improve agent productivity and increase customer loyalty, resulting in significant increases in revenue.
- Skills-based and Language-based Routing – directs calls to specialist agents, resulting in improved customer service.
- Agent Phone Set Display – provides agents with caller data such as CLID and DNIS right on their phone's LCD screen.
- Call Priority – helps contain costs by answering calls placed to the toll-free number ahead of calls entering the system via the local line.
- Delegated Call Center Management – enables the administrator to offload certain management tasks to the call center supervisor while still maintaining control over the system.

## Call Center Sales Scenario

### The Challenge

A trucking company with 70 full-time employees, James & Son serves a wide range of customers, both large and small. On a daily basis, James & Son receives a high volume of inbound calls, from both customers and truck drivers, in addition to numerous generic calls that are directed by attendants to accounts payable and receivable. The company's two attendants spend most of their day on the telephone and have little time for other duties.

James & Son found that as its business grew, the sheer volume of calls was becoming too much for attendants to handle. It became clear that the existing call center would not be able to handle the anticipated leap in call volumes. Realizing that increased call center efficiency was essential to meeting its business goals, James & Son decided that it had to upgrade or replace its existing system; it needed a way to route calls properly, allow customers to self-serve and manage the call center more effectively. James & Son identified the need to develop its existing call distribution system into a solution that could handle large volumes of calls, while delivering a high level of service to its customers. tight integration between voice and data would help shorten call times and improve customer service.

## The Nortel Networks Solution

BCM Call Center applications provide businesses with a highly cost-effective tool to manage how calls are distributed within their organizations. BCM assists companies to better manage incoming calls and enables more calls to be processed by the same number of agents – with immediate benefits including increased customer satisfaction, staff productivity and revenues.

More efficient call allocation translates into reduced long-distance call charges, increased customer satisfaction and reduced hold time.

BCM Call Center can be tightly integrated with Voice Messaging. It complements Voice Messaging capabilities by adding enhanced Custom Call Routing and call queuing functionality. BCM Call Center has also been designed for intelligent integration with the BCM and telephone sets.

## Implementation

To ensure easy integration, the Call Center applications are programmed with the same Graphical User Interface (GUI) administration tool used for Voice Messaging. If the distributor or technician is familiar with messaging, the call center applications are an easy, intuitive, next step. Call Center applications are compatible with all BCM hardware platforms. Agents can use the M7310, M7324, T7208, T7316, IP telephone i2004 and IP softphone i2050.

Call Center applications also use the telephone sets' LCD display to assist users in using features. The functions of the soft keys also change by activity, bringing up functions when appropriate, thus "teaching" the agent how to use the system.

## Business Impact

The new system has provided James & Son with a great deal of flexibility. Callers have the option to move their call out of the system to speak with an attendant. This capability encourages customers to use the automated system for administrative inquiries, freeing attendants to handle more service-focused calls. The automated call answering function automatically directs the majority of James & Son's incoming phone calls. This function reduces phone traffic to the attendants and improves customer satisfaction by delivering calls to the intended destination more quickly. James & Son has received very positive responses from customers who have commented on the improved customer service.

James & Son has also realized cost savings with the implementation of its new system. The Call Priority capability reduces costs by answering calls placed to the toll-free number ahead of calls entering the system via the local line.

# Overview

The BCM provides a call center strategy designed specifically to meet the needs and dynamic nature of small to medium-sized businesses. It's a single source, cost-effective, integrated, scalable solution.

With the introduction of BCM Release 2.5, Nortel Networks increased the size of Call Center Professional to 50 queues and 80 Active Agents (from 16 and 48 respectively), enhanced the Intelligent Routing and added multimedia capabilities. Building on the capabilities past releases, Release 3.0 of the BCM now offers additional features, including 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 Call Center Reporting Package and/or upgrade to the Professional Call Center. The Professional Call Center includes reporting capabilities. With the Call Center Reporting Package, both the Basic Call Center and Professional Call Center support an IP wallboard and softboard.

The Basic Call Center 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 ACD Agent Groups
    - 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.

BCM Professional Call Center also 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
- 150 Built-in user Recorded Announcements
- 50 Skillsets
- 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 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 Call Center Reporting 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.

# Emerging Trends

Today's call centers vary in terms of size, scope and complexity. The relative scale of a call center operation must be measured using more than one of these factors. Traditionally, call centers were large and formal; today, a small order department might serve a national market and require integration of sophisticated applications such as Voice Mail.

Formal call centers are generally departments whose sole function is to respond to incoming calls. Informal call centers, on the other hand, are groups or departments that perform a significant amount of telephone activities but do not exclusively answer or respond to call center calls.

# Benefits

When calls are distributed properly, businesses can see increased customer satisfaction, staff productivity and revenue, as well as an improved office environment. BCM Call Center applications assist companies in managing incoming calls and allow more calls to be processed by the same number of agents. BCM provides businesses, branch offices and departments with an effective tool to manage and organize call distribution. Calls can be distributed to the longest idle agent, to ensure equitable distribution of calls, or they can be distributed using the Top-down or Preferred manner.

Research has shown that up to one third of callers inquire about a product or service advertised while they are on hold. The Call Center applications provide announcements that encourage callers to continue holding by advertising products and services.

In addition, businesses can increase agent productivity by using reports and the real-time status display options provided by the BCM Call Center Reporting to manage individual and agent skillset performance. Industry studies show that an increase in agent productivity can range from 20to 40%, depending on current call answering practices.

A business' objectives are to:

- Increase revenues
- Reduce costs
- Improve customer service
- Optimize staffing resources.

BCM Call Center applications enable businesses to meet these objectives in the following ways:

## Increase Revenues

Call Center applications can help businesses create new revenue opportunities by expanding their market coverage, marketing to their account base, extending their sales hours and educating their customers. Businesses can use the routing features to automatically direct calls and offer extended hours, while providing superior customer service and increasing revenues. With a Call Center application, a business can cost effectively expand geographically, even internationally and can turn marginal accounts into major customers.

Intelligent routing allows a business to better match incoming caller requests to an experienced agent to maximize sales potential.

## Reduce Costs

Call Center applications can reduce a business' operating costs, sales costs, complaint costs and warranty and service costs. BCM Call Center applications can help reduce operating costs by increasing flexibility of employees logging in and out of skillsets to match calling traffic and minimizing or eliminating receptionist call handling. Additionally, it can increase agent satisfaction and reduce agent absenteeism and turnover.

More efficient call allocation can also translate into reduced long-distance charges through better management of phone lines. BCM Call Center can help reduce trunk costs by shortening customer hold time on 800/888 lines and reducing the number of returned calls to customers. Multimedia Call Center can eliminate long-distance hold time costs for either the customer or the business 1-800 line, significantly reducing costs.

BCM Call Center can reduce sales costs by:

- Improving customer service
- Building customer loyalty
- Decreasing customer complaints.

## Improve Customer Service

BCM Call Center can increase customer service by extending customer reach, through more effective call handling and by answering calls immediately. Moreover, businesses can easily make changes – fine-tuning and optimizing the Call Center. For example, a business might record the following message that provides callers with important information: “All tickets for the concert have been sold – if we can assist you in some other manner, please stay on the line.”

## Optimize Staffing Resources

BCM Call Center applications also help businesses to optimize their staffing resources. Call routing abilities, for example, help to balance the workload and offload work from busy employees. Multiskilled agents can be logged into more than one skillset at a time, maximizing their efficiency. Call Center applications enhance overall work environment and staff morale by reducing noise levels and ensuring an even distribution of calls.

# Vertical Markets & Potential Applications

The following table identifies potential applications of Call Center in various markets.

***Table 8-1***

| | Air Freight | Gov’t State /Local | Cable TV Co. | Credit Approval Co. | Distribution Co. | Banks, Credit |
|---|---|---|---|---|---|---|
| Appointment Center | | | √ | | √ | |
| Catalog Sales | | | | | √ | |
| Claims | | | | | √ | |
| Classified Ads | | | | | | |
| Circulation | | | | | | |
| Credit Authorization | | | | √ | | √ |
| Customer Service | √ | √ | √ | | √ | √ |
| Dispatch | √ | | √ | | √ | |
| Help Desk | | √ | √ | | | |
| Information | √ | √ | √ | √ | | √ |
| Order Entry | √ | | √ | | √ | |

| | Air Freight | Gov't State /Local | Cable TV Co. | Credit Approval Co. | Distribution Co. | Banks, Credit |
|---|---|---|---|---|---|---|
| Registration | | √ | | | | |
| Reservations | | | | | | |
| Shareholder Services | | | | | | √ |
| Technical Services | | √ | √ | | √ | |

## 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 Networks 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 Voice 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 8-2***

| Features | Basic Call Center | Professional Call Center |
|---|---|---|
| Skillsets | 2 | 50 |
| Configured Agents | 20 | 250 |
| Agent IDs | 20 | 250 |
| Active Agents | 10 | 20, expandable to 80 |
| Number of Active Calls in all “Call Center” Skillsets | 15 | 48 |
| Agent Priorities | Yes; 20 | Yes; 20 |
| Dynamic Agent Priorities | No | Yes |
| Maximum Number of Active Calls per ACD Skillset | 15 | 48 |
| Number of Lines which can be configured (answered) for the Call Center | 15 | 48 |
| Number of Voice Ports (shared with Voice Mailer dedicated) | 12 | 12 |
| Number of Routing Tables per Queue | 2 | 2 |
| Number of Recorded Announcements | 10 | 150 |

| Features | Basic Call Center | Professional Call Center |
|---|---|---|
| Number of Steps per Day Routing Table | 20 | 20 |
| Number of Steps per Night Routing Table | 20 | 20 |
| Number of Overflow Rules per Queue | 10 | 10 |
| Number of Queue Mailboxes | 2 | 50 |
| Supervisor Silent Monitor | Yes | Yes |
| Intelligent Routing **Basic** is the system's ability to route a call to the Operator, Auto Attendant, or Queue Mailbox based on programmable single digit caller input. | Yes | Yes |
| Intelligent Routing **Advanced** provides tremendous flexibility. Callers can be routed to other extensions, skillsets, or external numbers. | No | Yes |
| Intelligent Routing – CLID/ANI | No | Yes |
| Intelligent Routing – DID/DNIS | No | Yes |
| CLID/ANI or DID/DNIS Priority | No | Yes |
| Intelligent Routing – customer input | Partial – Single Digit | Yes |
| Intelligent Overflow allows calls to overflow or move based on preprogrammed conditions. | Available | Available |
| Call Distribution – Linear | Yes | Yes |
| Line Priority | Yes | Yes |
| Line Priority – Dynamic | Yes | Yes |
| Call Distribution – Longest Idle | Yes | Yes |
| Forced Announcement | Yes | Yes |
| Forced and Manual Call Presentation | Yes | Yes |
| Delay Answer Routing Step | Yes | Yes |
| No Answer Routing Step | Yes | Yes |
| Disconnect Routing Step | Yes | Yes |
| Agent Not Ready/Make Busy | Yes | Yes |
| Auto Agent Not Ready/Make Busy | Yes | Yes |

| Features | Basic Call Center | Professional Call Center |
|---|---|---|
| Auto Agent Log Out | Yes | Yes |
| Multiple Language Support | Yes | Yes |
| Queue Status | Yes | Yes |
| Queue Status Threshold Alerts | Yes | Yes |
| Day Mode of Operation | Yes | Yes |
| Night Mode of Operation | Yes | Yes |
| Record a Call | Yes (Voice Mail) | Yes (Voice Mail) |
| Automatic or Manual Mode of Operation change (Day/Night) | Yes | Yes |
| 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. | Yes | Yes |
| Wallboard Support | Yes (With Call Center Reporting) | Yes (With Call Center Reporting) |

## 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 8-3***

| Feature | Description | Benefits |
|---|---|---|
| Recorded Announcements | Play for callers as a step in the Routing Table<br>Are user-recorded and can be easily changed to reflect the business' daily requirements<br>Users can record up to 150 announcements and do not require additional equipment<br>**Note**: Many competitive systems require that a business purchase a separate device to record announcements. | Improves customer service<br>Saves money, as these announcements do not require additional equipment |
| Forced Announcements | Play from start to finish regardless of whether an agent is available<br>Can provide information about products or services, or advise callers to have their customer identification available | Provides more efficient call handling by giving the caller more information prior to being connected to the agent |
| 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<br>Is particularly important when agents are logged in to more than one skillset or when calls overflow | Prevents confusion<br>Improves customer service |
| 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 | Improves efficiency, as the calls gets routed to the appropriate agent |
| Intelligent CLID Routing | Routes calls according to where the originated | |

| Feature | Description | Benefits |
|---|---|---|
| Intelligent DID/DNIS Routing | Routes calls according to the destination or the number dialed by the caller | Improves efficiency |
| CLID/ANI or DID/DNIS Priority | Allows the administrator to determine the order in which routing decisions are made | If both CLID/ANI and DID/DNIS are available the system administrator can determine which should take priority |

### 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 8-4*

| Feature | Description | Benefits |
|---|---|---|
| Call Distribution | There are two methods for distributing calls to agents:<br>Longest Idle – the system selects the agent who has been available the longest since last handling a call<br>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 | Results in a more balanced call-handling load |
| Delay Answer Routing Step | Instructs the Call Center to delay answering incoming calls for a specified time period<br>Provides the caller with ringing tones until the delay time passes or an agent becomes available | In normal operation if all agents are busy the call center will answer the call immediately and play an announcement to the caller. If a delay answer routing step is used, the call will ring for the length of the delay answer routing step. In the case of 1-800 numbers this can save the business considerable toll charges. |
| No Answer Routing Step | When this command is the first step in the routing table, the call will continue to ring until the caller hangs up<br>Typically used after hours | Saves money, as businesses do not incur 800/888 line charges |
| Disconnect Routing Step | Disconnects callers when they encounter this step | Typically used after hours by businesses that do not want to accept messages<br>Usually used in conjunction with an announcement that might advise the caller that the business is now closed |
| Skillset Mailboxes | Belong to each skillset<br>Allow callers to leave messages for the appropriate skillset | Improves customer service |

### 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 8-5***

| Feature | Description | Benefits |
|---|---|---|
| Agent IDs | Numbers assigned to each individual<br>Can have different characteristics<br>A single employee could have multiple Agent IDs and use the appropriate ID based on the call center's requirements at the time<br>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<br>At the time of log in, the agent set will display the skillsets that the agent is set up to log into<br>The agent can then log into all of these or select which skillsets they will log into | Provides tremendous flexibility, meeting the changes in call center traffic |
| Break or Wrap (post-call completion timer) | The length of time an agent is provided between calls to process paperwork<br>Its duration is user-definable (0 – 60 seconds) | Improves employee efficiency and customer satisfaction by ensuring that agents have sufficient time to complete paperwork before accepting the next call |
| Not Ready or Agent Make Busy | Notifies the system that an agent is logged in but does not want to receive calls<br>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 | Can be used if an employee requires more time then provided by the "wrap" feature after completion of a call<br>Improves employee efficiency and customer service |

| Feature | Description | Benefits |
| --- | --- | --- |
| 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 | Ensures calls don't go unanswered<br>Improves efficiency |
| 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 voice 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 voice mailbox | Allows managers to record conversations |
| Supervisor Silent Monitor | Allows supervisors to silently monitor the conversation of both agents and callers. Supervisors can monitor both incoming and outgoing calls | Lets managers evaluate the level of customer service agents are delivering to customers |
| Skillset (Queue) Status | Displays the following information to an agent through the LCD on their set:<br>Skillset number and status: Enabled, Disabled or Uninitialized<br>Number of agents currently logged on to that skillset<br>The number of calls currently in the skillset<br>Amount of time the oldest call has been waiting in the skillset. The wait time appears in minutes and seconds | Provides information to the agents so that they can more effectively do their job<br>Increases employee efficiency, allowing them to provide a higher level of service |

| Feature | Description | Benefits |
| --- | --- | --- |
| 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<br>The LCD associated with the Skillset Status feature key displays the following:<br>If the LCD is off, all of the calls that are queued to the skillset are within the acceptable wait time<br>If the LCD is flashing slowly, at least one waiting call has exceeded the marginal wait time<br>If the LCD is flashing quickly, at least one waiting call has exceeded the acceptable wait time | Gives the agents an indication as to how busy the their skillsets are. Agents can then agent call handling times according<br>Improves employee efficiency |

BCM Release 3.0 introduces Silent Monitor as part of Basic and Professional Call Center.

- In addition, Silent Monitor for Hunt Groups is part of the core telephony of BCM 3.0.

### Silent Monitor for Hunt Groups

This feature is part of the BCM core telephony 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, T7316, M7310 and M7324 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 voicemail 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 8-6***

| Feature | Description | Benefit |
| --- | --- | --- |
| Delegated Call Center Administration | Lets the BCM administrator delegate and give passwords for call center administration to a call center delegated supervisor | Increases flexibility without sacrificing security |
| Multiple Language Support | *Set Display*<br>The Call Center telephone set display is available is in:<br>• North American English<br>• Latin American Spanish<br>• Danish<br>• Dutch<br>• French<br>• German<br>• Norwegian<br>• Spanish<br>• Swedish<br>• UK English<br>• Italian<br>*Reporting*<br>The Call Center reporting is available in:<br>• North American English<br>• Latin American Spanish<br>• French<br>• Spanish<br>• German.<br>• Italian | |

| Feature | Description | Benefit |
|---|---|---|
| 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 Call Center Reporting 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.”
- *ip*View wallboards available with the BCM with Call Center Reporting Package.

The Call Center Reporting 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:

- M7310
- M7324
- T7208
- T7316
- Internet Telephone i2004
- Internet Telephone i2002IP SoftPhone i2050

# 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.

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.

## Benefits of Multimedia Call Center

The Multimedia Call Center application is a great way for businesses to increase the value and investment in the various components of their communications infrastructure and enhance the services offered by the BCM. By enabling real-time voice and chat communication, the Multimedia Call Center allows customers to place a request for a customer service agent just by clicking a voice button.

Multimedia Call Center offers substantial benefits, enabling businesses to:

- **Increase revenues** – Businesses can use the power of the Internet to reach untapped international markets. In addition, businesses can upsell and cross sell the customer. With Multimedia Call Center refresher announcements, businesses can promote other products and services, resulting in increased sales. Also, Web Push and Follow-me browsing allows the customer to send pages to the agent and vice versa; thus, it is easier to eliminate the competition by countering other products that the customer might be considering.
- **Increase employee efficiency and morale** – Functions like “Wrap” and “Not Ready” allow agents to complete any necessary paperwork before the next call, increasing productivity. Intelligent routing distributes calls to agents in an equitable manner while ensuring that a given call will be routed to the agent with the appropriate skillset.
- **Improve customer satisfaction** – Accessibility is one of the key components of customer service. With Multimedia Call Center, businesses can be easily reached, and this accessibility will result in improved customer loyalty.

- **Reduce costs** – As the hold time is on the IP network, Multimedia Call Center reduces network costs.

### Financial Model Due to Lower PSTN Costs as a Result of IP Queuing

- Average hold time of a call is 4 minutes
- Talk time is 2 minutes
- Cost of toll service is $.10/min
- Total time per call is 6 minutes, averaging 10 calls per hour.

The cost associated with the hold time would be eliminated because while the caller is in queue the request uses the IP Network. Therefore:

- 4 min/call X $.10/minute = $.40/call
- 10 calls/hour X 8 calls/day X $.40/call = $32/day
- $32. X 5 days/week X 52 weeks = $8,320.00/year.

These figures are for one skillset or queue at 100% capacity. A typical customer will not be at 100% capacity but they will usually be very close. Having a call center implies that there are more trunks than agents.

100% of a business' traffic will not originate from the Web. Therefore, the model depends on the customer and what makes sense for them; for example, growing from 10% to 20% to 30% and so on. The BCM can have up to 50 skillsets or queues. Any number of the BCM's skillsets can be multimedia enabled. The savings per skillset should then be multiplied by the number of skillsets that are multimedia enabled.

Multimedia Call Center has the following features that help reduce the amount of time employees require with each caller:

- Refresher Announcements can ask callers for information available to the agent
- The agent is presented with the Web page that the caller was visiting when he or she called
- Web Push and Follow-me browsing allow the agent to send exactly the right information to the caller
- Page lists are lists of Web pages that the agent can quickly reference, and. if needed. send to the caller.

# 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

BCM Call Center Reporting is an optional package that runs on a customer-supplied PC alone or alongside existing applications. Purchasing a software keycode (separate from the Call Center Keycode) activates the Call Center Reporting application.

Call Center Reporting provides a cost-effective, reliable solution that meets the needs of small to medium-sized call centers that require enhanced management reporting capabilities. It provides real-time access to a wealth of information that can be used by Call Center managers to make the most efficient use of lines and agents.

Call Center Reporting helps manage the peaks and troughs in call traffic, providing information such as call waiting times, queue length and agent status. It offers 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 managers run Call Centers more efficiently. Having the right information allows a manager to respond to problems in a timely manner.

The Call Center Reporting software:

- Runs alongside existing software applications on any Windows 95/98/2000/NT/XP capable PC
- Supports up to 16 supervisor workstations on a single system without affecting the status display screens on each other's PC
- 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 *ip*View wallboards
- Provides password protection.

The Call Center application runs on the BCM and must be enabled and configured before the reporting software is enabled. Once customers purchase a keycode and enable the Call Center Reporting software via the Web-based interface tool, they can then download the Client software onto the Client PC. The Client PC must be running whenever statistical data needs to be collected. It is recommended that this PC be running at all times.

Any additional PCs that require access to information stored on the BCM may also download the Client software using the same process as the Client PC. These additional PCs are called Multiple Clients. The Client software can be installed alongside existing applications on any Windows 95/98/2000/NT/XP capable PC.

## Real-Time Status Display

The real-time status displays provide up-to-the-minute views of the activity in the call center. The screens and information provided are continually changing, reflecting the status of agents, skillsets and calls.

Once the user selects real-time status display at the system level on the Client PC, four windows open on the screen (only two windows can be seen at one time). The first two windows display in graphical format; colors and numbers illustrate activity. If a user closes or minimizes the two graphical windows, the view changes to windows that show the real-time status numerically.

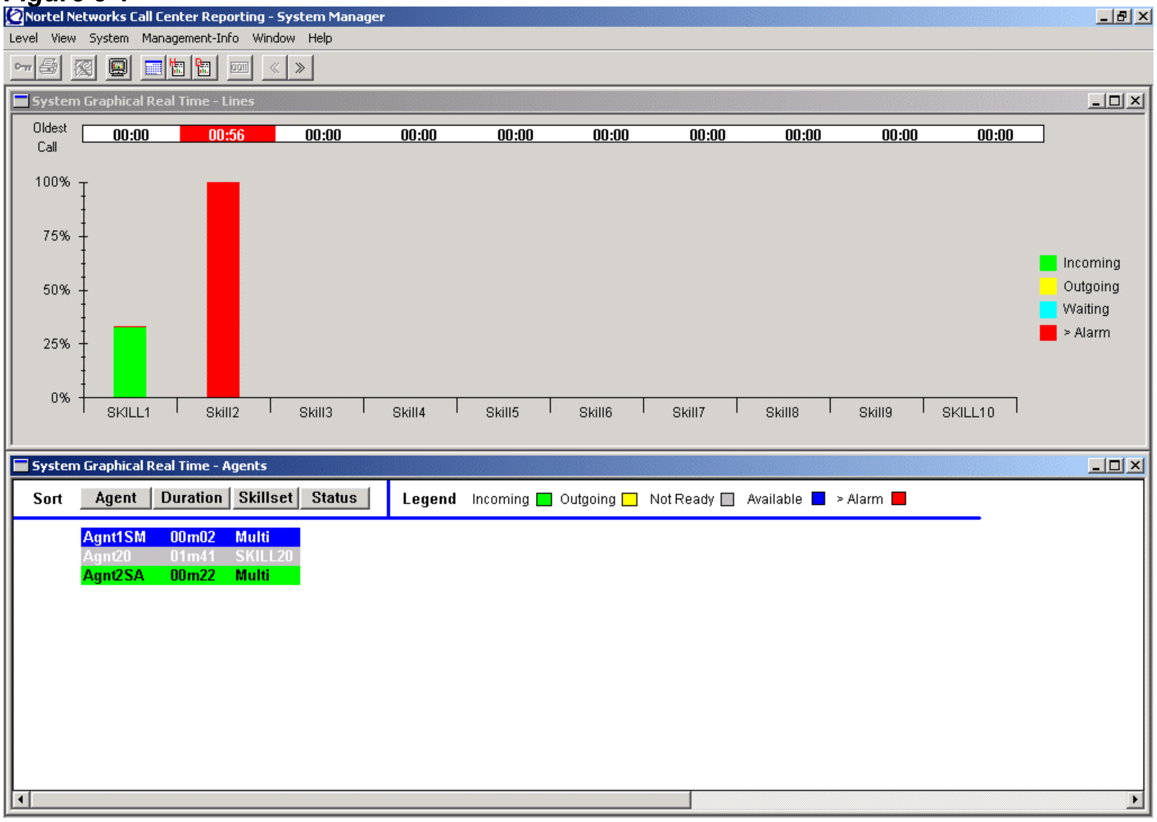
### Graphical Example: Real-Time System Status Display

The screen shot in Figure 8-1 shows a graphical display of the system lines (in the upper half of the window) and a graphical display of the system agents (in the lower half of the window).

The System Status window shows the following line activity for each skillset:

- Queue time of longest waiting call
- Number of incoming and outgoing calls
- Number of calls waiting and greater than threshold
- Number of lines available for calls.

*Figure 8-1*




Information displayed in the System Status – Agents window can be sorted by agent, status duration, skillset name or agent status. The System Status – Agents window shows the following activity:

- Names and status of all agents logged in
- Status duration of all agents logged in
- Skillset name to which each agent belongs.

## Numerical Example: Real-Time System Status Display

Users can access the numerical view of the *Real-time System Status* display by closing or minimizing the two graphical view windows shown in Figure 8-1. Like the graphical view, the numerical view has a window for system lines and a window for system agents. The windows in Figure 8-2 show, in numerical values, the same information as the graphical windows.

***Figure 8-2***

Nortel Networks Call Center Reporting - System Manager - [System Numeric Real Time - Lines]

Level View System Management-Info Window Help

| Skillset | Total | Waiting | > Alarm | Outgoing | Incoming | Oldest Call | Mode |
|---|---|---|---|---|---|---|---|
| SKILL1 | 4 | 1 | 1 | 0 | 2 | 02:25 | In |
| SKILL2 | 1 | 1 | 1 | 0 | 1 | 00:50 | In |
| Skill3 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill4 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill5 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill6 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill7 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill8 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill9 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| SKILL10 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill11 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill12 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill13 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill14 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill15 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill16 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill17 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill18 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill19 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| SKILL20 | 0 | 0 | 0 | 0 | 0 | 00:00 | In |
| Skill21 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill22 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill23 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill24 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill25 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill26 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill27 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill28 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill29 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| SKILL30 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill31 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill32 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill33 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill34 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill35 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill36 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |
| Skill37 | 0 | 0 | 0 | 0 | 0 | 00:00 | Out |

For Help, press F1

### Graphical Example: Real-time Skillset Status Display

This display shows Call Center activity for each individual line and agent in the skillset.

The upper left-hand section of the screen in Figure 8-3 is a graphical window that uses pie charts to show line and agent activity. As lines and agents become busy, or if they are dealing with incoming or outgoing calls, the charts will reflect the percentage of activity.

The upper right-hand section of the screen in Figure 8-3 is a graphical window that shows the status of each agent logged on to the skillset. Activity is shown as follows:

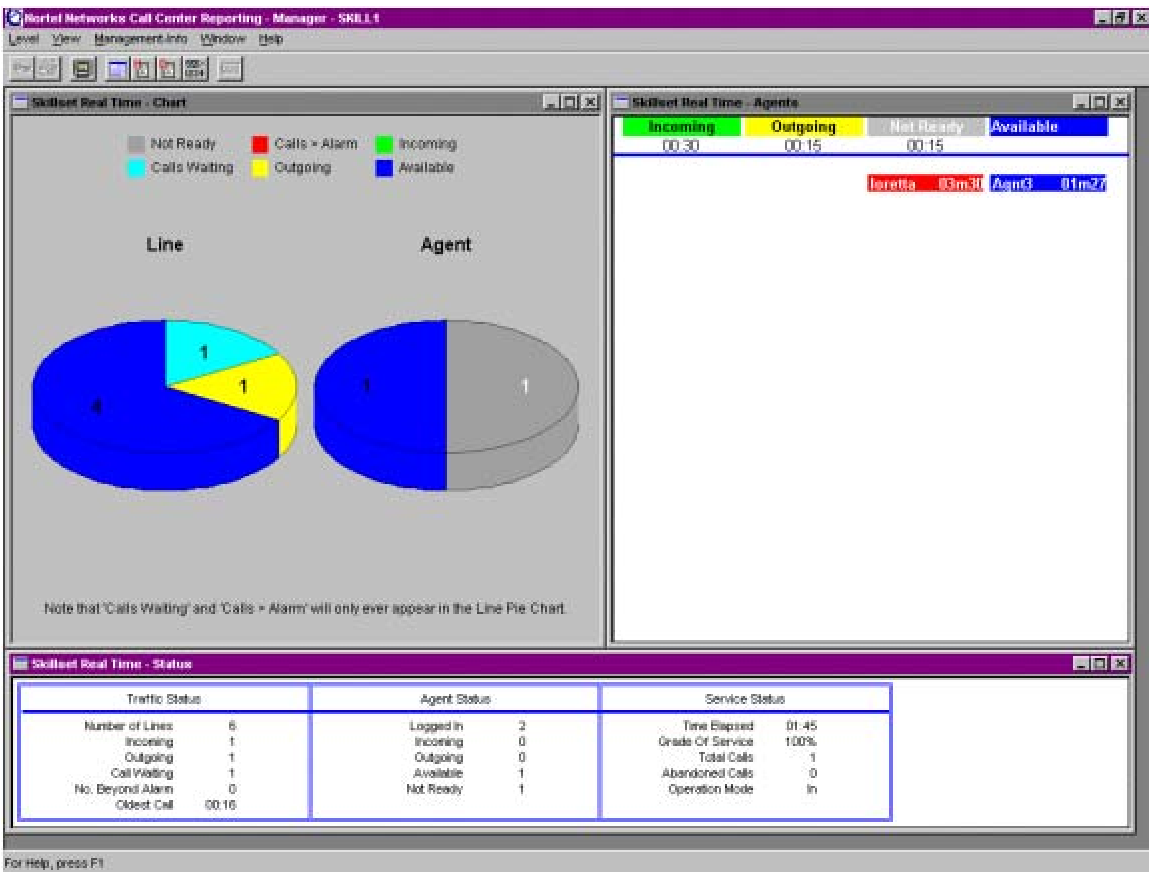
- Agents on outgoing calls are highlighted in yellow
- Agents on incoming calls are highlighted in green

- Agents who are busy are highlighted in gray
- Agents who are free and ready to take incoming calls are highlighted in blue.

The time thresholds along the top of the columns in the agent status window are programmed in the real-time setup windows. When an agent exceeds any of these thresholds, his or her name is highlighted in red.

The bottom section of the screen is a numerical window that presents the same information in a three-column, numeric format.

***Figure 8-3***




## Current and Historical Reports

BCM Call Center Reporting 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. Call Center Reporting 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 Call Center Reporting

Call Center Reporting 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 operating system
- Network Interface Card (NIC)

- TCP/IP protocol
- SVGA display.

## Wallboards and Softboards

With the addition of the optional Call Center reporting package, BCM Call Center can support up to 16 customized wallboards or unlimited *ip*View 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

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 Call Center Reporting application and can be located anywhere on the customer's LAN. The wallboard configuration is a simple, straightforward extension of the Call Center Reporting 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 for 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

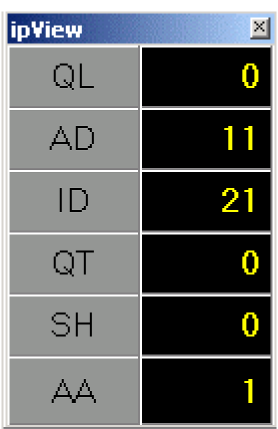
These alarms can be set up to send a message to the wallboard when any of the parameters reaches a critical threshold. The system can be programmed so that if any of the parameters is equal to, equal to or greater than, or equal to or less than a customized message can be sent to the wallboard.

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.

## *ip*View Softboard

*ip*View 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 Call Center Reporting software. A site license is provided; therefore, there is no limit to the number of agent desktops that can have *ip***View** Softboards installed.

***Figure 8-4***




Call Center statistics and messages from the BCM's Call Center Reporting application are displayed in real-time by the *ip***View** 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 *ip***View** 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 *ip*View is an intuitive logical extension of configuring the IP hard wallboards. OA & M is accessed through the appropriate tab from BCM's Call Center Reporting.

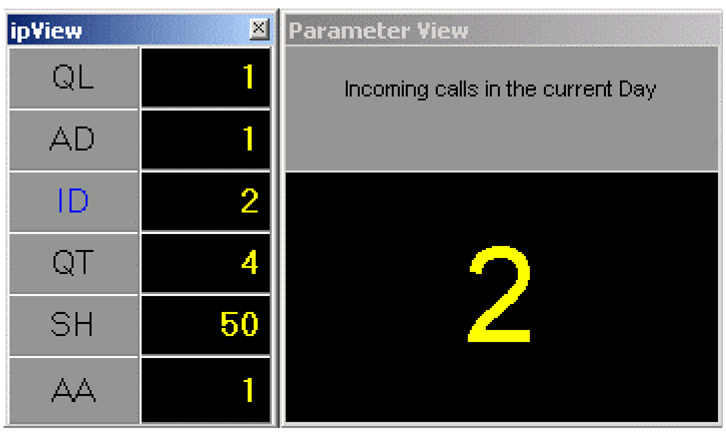
Like the wallboard, the *ip*View softboard can be installed anywhere over a customer's LAN/WAN and no additional server is required. All the information is provided directly from the Call Center reporting.

*ip*View softboard supports English, French and Spanish.

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

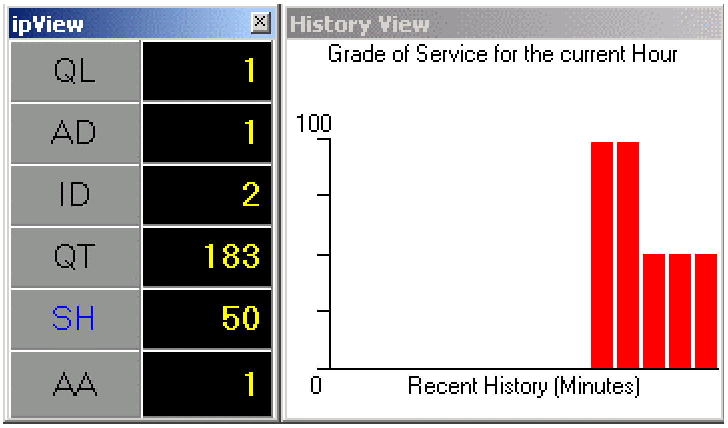
### Parameter View

***Figure 8-5***



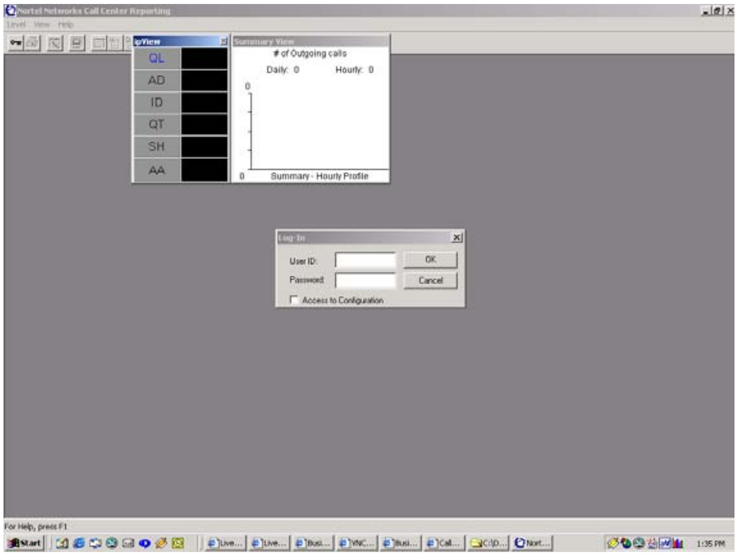
### History View

***Figure 8-6***



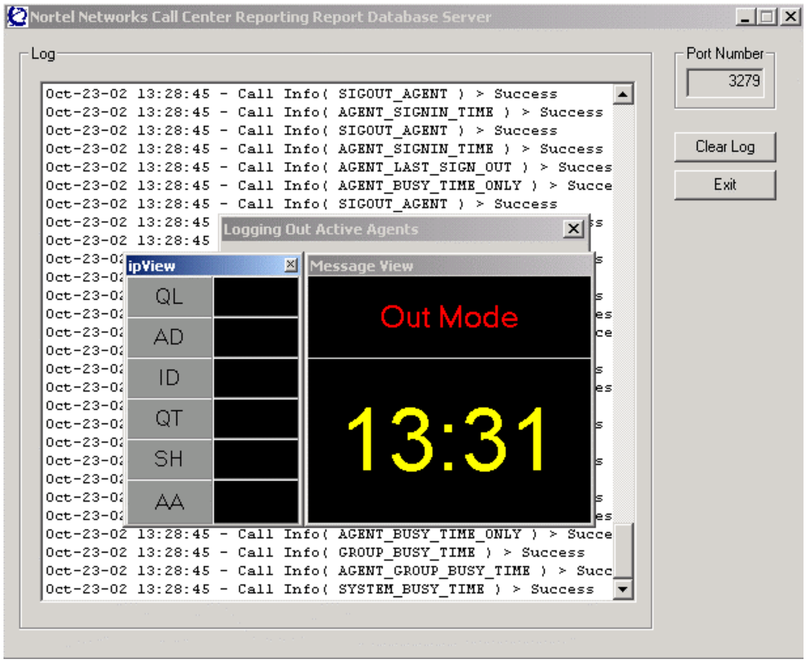

### Summary View

***Figure 8-7***



### Message View

***Figure 8-8***




## PC Requirements

***Table 8-7***

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

# 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.

## Call Center Reporting Keycode

This keycode option enables the Call Center Reporting capabilities for Basic Call Center. Professional Call includes Call Center Reporting; 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.

The Call Center Reporting also contains the *ip*View Softboard application software. For more details, please see the section on *ip*View Softboard.

Please note: Call Center Reporting 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 Call Center Reporting 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 Call Center Reporting 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 Call Center Reporting Package.

## Multimedia Call Center Keycode

This keycode option enables the Multimedia Call Center. With BCM 3.0, Multimedia Call Center 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.

## 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.



# Mobility





# Interactive Voice Response (IVR)

# Interactive Voice Response (IVR)

## Chapter Highlights

- BCM 3.0 IVR Functionality – adds the next logical level of customer service to the business enterprise with multiple sites.
- Extended Customer Access to 7x24 – offloads routine inquiries and service requests with a highly cost-effective solution by leveraging industry leading Nortel Networks self-service Voice Portal Solutions portfolio.
- Mission Critical Reliability – lets businesses deliver high performance voice services and scales to the needs of a growing enterprise.
- Award Winning Development and Management Tools – in addition to an experienced Professional Services organization, are available to support full IVR functionality.
- Integrating IVR Capability – enhances the BCM Office-in-a-Box Strategy.
- Open Modular Design – delivers a smooth transition path to new technologies.
- Most IVR Needs – are supported by an exhaustive list of standard and optional features.
- Embedded IVR Functionality – offers a huge upside sales opportunity to financial services companies, retail pharmacy networks, warehouse clubs, transportation firms, department stores and other multiple-site enterprises.

## Overview

Nortel Networks Portal Solutions portfolio is a complete suite of advanced voice processing and speech technology products and services. BCM 3.0 now 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 the BCM 3.0.

# Emerging Trends

An old bumper sticker that once graced the tail section of many vehicles has become the strategic vision for numerous large enterprises. The sticker read: "Think globally, act locally." This approach has become the mantle for drugstores, coffee shops, banks and hundreds of other chain or branch store operations. The parent company has superior buying power and passes the cost efficiency and breadth of product or service offer down to the local level. It can function as a major mover in its chosen line of business while at the same time deliver its products and services with local relevance and distinction. Having numerous outlets poses a unique set of challenges that emerging technology is addressing.

One of the main challenges faced by businesses with branch locations is the ability to deliver world-class communications service to every site, large and small. This requirement often has the added burden of wide variations in local revenue, limiting the financial base to support a large cost structure. This type of operation typically has diverse needs of physical support to keep the communications infrastructure in tune with corporate requirements and the changes so often needed to maintain a competitive position.

Gartner Dataquest and InfoTech have described Nortel Networks as the leader in contact center solutions for enterprises (#1 in market share, July 2002). With such experience and insight into the market, Nortel Networks concurs with industry analysts in seeing firms relying more and more on methods of communication delivery as critical focal points for managing and building customer relationships. The communications method deployed, be it PSTN, VOIP, Web or other media, has become not just a touch point but also an integrated element of the business strategy and resulting objectives. But, while multiple channels for customer access have become very common in today's workplace, the telephone continues to take the major share of traffic because of its universality and limitlessness of options when connected to the end business.

Regardless of the trends in other directions, the telephone continues to be the predominate form of communications with, according to Giga Information Group, 75% of patrons using the telephone as the principal access medium (Herrell, Elizabeth. *Key Steps for Optimizing a Contact Center Upgrade*. Giga Information Group, May 20, 2002). Communications managers continue to look for new ways to enhance the interaction, automate more functions and improve the experience. This is leading many to the converged network, often based on Internet Protocols (IP), to enable multichannel communications. The objective of several leading multisite financial services firms has been to provide collaborative customer care at the branch office.

While the first Internet revolution allowed firms to offer online access to products and services, the second revolution, which is currently taking place, in a much quieter fashion, is using the Internet to build integrated applications. This integration started in contact centers and is now moving to the branch.

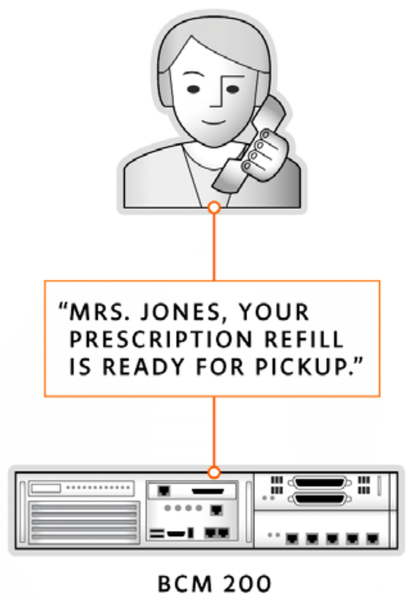
Customers, too, have become accustomed to technologies and are generally comfortable with alternative approaches that deliver the information or service exactly when they need it. As a result, Communications Directors continue to press cost control measures, especially in times of economic weakness, and for ways of delivering greater revenue. The automation of new features can often allow both reduced dependence on agents, as well as opportunities for fee-based services. The hunt for solutions that deliver a quicker return on investment is usually the one that extends existing systems to meet the new approach.

Businesses, both large and small, are focusing on quality of the interaction and customer relationship building more than mere call completion. This trend, along with an evolutionary meld of communications capabilities into the overall corporate strategy, begs the use of advanced features to intelligently route customers to knowledge workers, handle multichannel interactions and link separate data marts to enhance and support the interaction. The BCM 3.0 with IVR functionality adds the next logical level of customer service to the business enterprise with multiple sites.

## BCM 3.0 with IVR Value Proposition

Nortel Networks empowers enterprises with single site and multisite needs, from the SMB to the loosely coupled and highly replicated, through a cost-effective and fully converged voice and data solution. With an industry-leading and proven IVR runtime engine built in, the BCM 3.0 delivers the power of an enterprisewide contact center to the branch office in a flexible, feature-rich and scalable platform.

***Figure 9-1 – IVR embedded in the BCM 3.0***




## BCM 3.0 IVR Positioning

The BCM 3.0 with IVR functionality has been carefully positioned with other Nortel Networks products to support the full range of multisite enterprises. It starts with the small to medium-sized businesses (SMB) in a standalone or with small to medium-sized, multisite requirements. It can also handle enterprise-sized organizations with highly replicated sites or enterprises with loosely coupled and highly integrated sites. Where BCM ends, other Nortel Networks solutions can fill a customer's needs.

# Benefits

The addition of Interactive Voice Response (IVR) functionality embedded in BCM 3.0 provides a cost-effective all-in-one solution for end users to offload routine inquiries, handle transaction requests and service customer needs 24 hours a day, 365 days a year. Callers can access a broad range of information simply by responding to a series of prompts via their touchtone phone.

Nortel Networks Portal Solutions IVR portfolio is consistently ranked by many analysts, including Gartner in their Magic Quadrant (Gartner, Inc. *IVR Magic Quadrant for 1H02 – Challenges for Incumbents*. Research Note, January 9, 2002), as a leader in either the #1 or #2 position for completeness of vision and ability to execute. BCM 3.0 is leveraging Portal Solutions IVR proven and industry recognized capabilities to provide IVR functionality.

BCM IVR takes advantage of Portal Solutions established and proven full suite of development and management tools to empower local or remote application control. Nortel Professional Services or third-party providers can also be utilized for application creation.

The integrated IVR solution can effectively reduce maintenance costs by handling system and feature expansion via remote software keycode activation. The unique design allows for such enhancements without field hardware expansions.

By leveraging the BCM platform, IVR functionality can be fully integrated with existing BCM features, eliminating the need for extra-dedicated hardware resources. This completeness of offer delivers superior price performance, scalability and the lowest cost of ownership.

## Potential Market Applications

Nortel Networks has identified several potential markets that could benefit from implementing an IVR solution. These potential markets have been categorized based on the following industries:

- Financial services
- Healthcare
- Education
- Other.

### Financial Services

- Retail banks
- Brokerage firms
- Insurance agencies and servicing

### Healthcare

- Pharmacy chains
- Medical facility networks

**Education**

- Local school districts
- City schools
- Colleges

Other potential markets for IVR include auto dealers, warehouse clubs, airlines and department stores.

## Selling IVR with BCM

The following list highlights some of the main points sales professionals should direct at potential customers. BCM with IVR on BCM:

- Automates routine inquiries from staff, reducing cost of delivery and maximizing efficiency
- Leverages your investment in the BCM platform
- Leverages the Portal Solutions IVR best-in-class know-how and expertise
- Delivers proven mission-critical reliability to enhance your business reputation
- Provides high performance voice services in an open environment
- Scales with your requirements in both port capacity and feature function
- Offers lowest total cost of ownership
- Speeds service time –to market with simplified application development and streamlined network configuration/management
- Enhances the customer experience through dynamic nonblocking resource sharing
- Connects to most databases and a wide variety of network protocols.

# IVR Features

The IVR on BCM is the Media Processing Server 100 (MPS 100) from Nortel Networks 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 3.0 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, the 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 Networks 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 Networks Customer Contact and Voice Portal Solutions

IVR on BCM is part of the broad range of Customer Contact and Voice Portal Solutions that Nortel Networks 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 Networks 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 Voice Mail 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 3.0 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

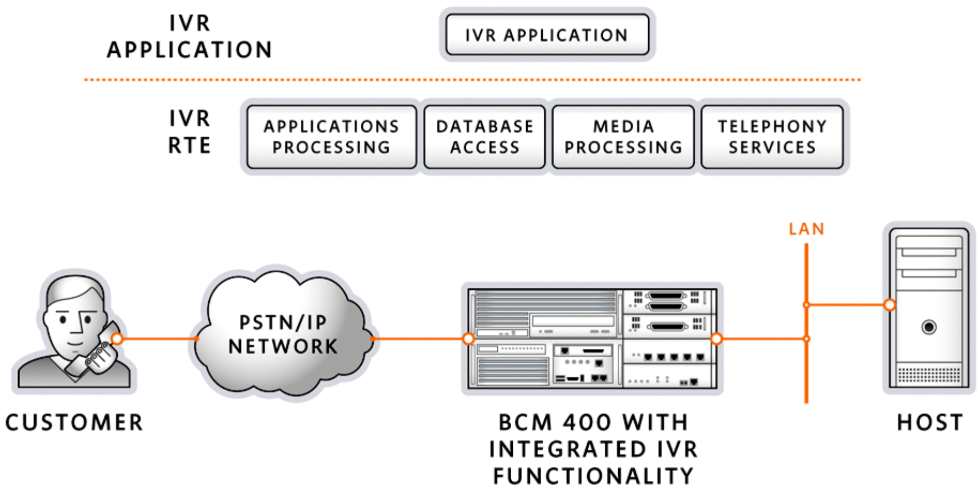
The Application is the second building block of an IVR solution. It turns the runtime engine functionality into the features and functionality that the customer uses. The application is developed for specific customer needs and, in many cases, is integrated with databases to enable real-time queries 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 Networks 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 9-2***




## 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 Voice Messaging. Or, they can start with Voice 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 Networks 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.

## Future Direction

The BCM product team is considering the expansion of IVR functionality that can be supported within the BCM platform. Additional features and functions currently supported in other Nortel Networks Voice Portal IVR solutions may be added in the future. Some of these additions include:

- Voice forms
- Advanced speech recognition
- Voice XML development environment.

No product announcement for these features is currently established.

# Mobility

## Chapter Highlights

- Nortel Networks Mobility Solutions for BCM enables employees with flexible voice communications to deliver superior customer service without being tied to their desk.
- 802.11 Wireless VoIP support with H.323+ is a multicell 802.11-based, feature-rich wireless VoIP integrated solution that utilizes H.323+ terminals from third-party vendors like Symbol Technologies.
- Business Series Terminal T7406 Cordless is a single cell, full-featured, multiline telephone for businesses that would benefit from a workspace mobility solution.

Note that Nortel Networks previous Companion UPCS multicell mobility system (PCI-United States or CT2Plus-Canada) is in the process of becoming Manufacturer Discontinued (MD) in North America. Support for Nortel Networks Companion UPCS installed base on BCM systems will continue to be offered as outlined in the products retirement notification.

## Overview

Nortel Networks offers mobility solutions that extend a site's ability to offer flexible access to the features and capabilities of BCM. The solution makes it possible for employees to deliver better customer service, and take care of more business without being tied to their desk. Depending on the site's requirements, Nortel Networks offers a mobility solution for either smaller single-cell or larger multicell applications. The following paragraphs provide an overview of the mobility solution offerings.

### 802.11 Wireless VoIP

Nortel Networks 802.11 Wireless VoIP BCM integration is an ideal solution for any enterprise looking to take advantage of the benefits that a converged wireless voice and data network has to offer, while leveraging their existing communications investments.

The 802.11 Wireless VoIP integration represents a key element in a BCM solution, providing enterprise businesses with a logical and realistic way of deploying standards-based IEEE 802.11

wireless services to save money and improve efficiency without sacrificing quality or convenience.

Supporting up to 80 IP users (resources dependent), the 802.11 Wireless IP integration on BCM delivers an advanced set of enterprise features to the wireless IP handset that are commonly found with Nortel Networks popular M7100 digital telephones.

The solution is well-suited for environments with a mobile workforce that can benefit from the increased mobility inherent with wireless networks, cost savings and simplification of reducing moves, adds and changes. In addition, this solution offers a mobile workforce the flexibility to enable innovative new productivity applications and Web-based wireless services over a single converged 802.11 Wireless LAN. An example of this capability is the ability to leverage the 802.11 wireless LAN network to address the integration of data applications, specialized mobile devices (i.e. PDAs) and other valued vertical market applications.

### Business Series Terminal T7406 Cordless

Nortel Networks offers a smaller, single-cell cordless office mobility solution that integrates seamlessly with the BCM system: the Business Series Terminal T7406 Cordless. The Business Series Terminal is a full-featured, multiline telephone for small to medium-sized business sites that would benefit from a desk-centric mobility solution. It covers an area of up to 282,000 square feet (approx 300 ft range from T7406 WallBase) and supports one to six users, enabling employees to be more productive while moving about the office. It is ideal for small enterprise, branch office, retail, medical office, warehouse and manufacturing environments.

## Emerging Trends

Over the last decade, there has been a significant growth in the use of mobile voice communications, and current telephony trends are continuing to emphasize mobility. The proliferation and success of mobile voice communications devices, both in the home (residential cordless) and in business (cellular and paging), have generated an increasing interest in private in-building mobility solutions for business. More and more customers are realizing the value of investing in a mobility solution for their business needs.

Beyond mobile voice communications, many enterprises are also looking to deploy real-world voice solutions over their existing packet networks for converged voice and data applications. These emerging customer requirements are intended to capitalize on the convergence of voice, data and wireless while providing investment protection by building on the feature richness and reliability of their communications platforms.

# Benefits

Nortel Networks Enterprise portfolio of wireless mobile communications solutions offers a number of significant benefits to both small and medium-sized business users. Each feature-rich solution is fully integrated into the BCM system but address two different customer requirements and applications: multicell converged wireless applications with 802.11 Wireless VoIP or single-cell wireless voice applications with the Business Series Terminal T7406 Cordless.

## 802.11 Wireless VoIP

The 802.11 Wireless IP solution builds upon the freedom of wireless by empowering employees with integrated voice and data applications and devices for true workplace mobility. This multicell solution offers the following benefits:

- **Increases employee personal productivity** – by enabling site workers to have instant access to information, reduce downtime, improve coordination and streamline operations with a wireless IP handset that functions like a digital or IP extension of the BCM system.
- **Extends and leverages the Wired IP network** – by using an industry standard IEEE 802.11 wireless LAN and Ethernet LAN to enable the advanced wireless VoIP applications over a converged network, thereby reducing the administration and maintenance costs of supporting two separate wireless communication infrastructures.
- **Offers an integrated, advanced feature set –** providing users with access to an impressive range of enterprise-class desktop features and functionality. 802.11 Wireless IP solution users also gain access to the rich integrated applications suite supported by the BCM system, including unified messaging with CallPilot and other integrated applications.

The wireless VoIP solution is geared towards environments with a mobile workforce that want to benefit from the increased mobility inherent with wireless networks, cost savings and simplification of reducing moves, adds and changes. In addition, this solution is suitable for environments that require flexibility to enable innovative new productivity applications and Web-based wireless services over a single converged 802.11 WLAN.

In addition to offering our customers a converged voice and data solution that leverages a common wireless infrastructure, the 802.11-based offering is also better able to address the integration of specialized mobile devices (i.e. PDAs) and valued vertical market applications.

The 802.11 wireless VoIP offering will apply to numerous vertical markets and branch office environments with requirements for both voice and data mobility and instantaneous accessibility. Some examples of good target markets for the solution include manufacturing, retail and education.

### Manufacturing

Manufacturing has a number of applications for the 802.11 Wireless offering, including supervisors who can work throughout the production floor and monitor status and technicians who can be contacted immediately to repair equipment, as well as the freedom for employees to be mobile but still be in touch.

### Retail

The retail market is very competitive – lost calls can mean lost business. The 802.11 wireless offering ensures that customers can get their calls through to the intended party, thereby improving customer service. The reduction in overhead paging also improves the environment for retail customers.

### Education

Education (K-12) has a number of applications for the 802.11 Wireless offering, including teachers who now could have a wireless IP phone in the classroom and also could be reached throughout their campus coverage area. Teachers, maintenance, security and the administration staff could also be notified quicker and better react to emergency situations.

## Business Series Terminal T7406 Cordless

The Business Series Terminal T7406 Cordless enables enterprises to deliver better customer service while making employees more productive with workspace mobility. This single-cell solution offers the following benefits:

- **Increases employee personal productivity –** by satisfying employees' needs to be on the move to get the job done while remaining accessible to customers and staff.

- **Offers cost-effective desk-centric mobility** – because it is designed for use in desk-centric environments ranging from small to medium-sized enterprise sites, to branch office or franchise site. The cost-effective mobility solution delivers advanced business features with plug-and-play simplicity.
- **Offers an integrated, advanced feature set** – by leveraging the features of the popular Business Series Terminals portfolio, giving users access to an impressive range of enterprise-class desktop features and functionality. T7406 Cordless Telephone users also gain access to the rich integrated applications suite supported by the BCM system, including unified messaging with CallPilot and other integrated applications.

The T7406 further demonstrates Nortel Networks the commitment to provide users with flexible and productive tools to empower a business' success with outstanding features, durability and reliability.

# 802.11 Wireless VoIP Support with H.323+

## Solution Overview

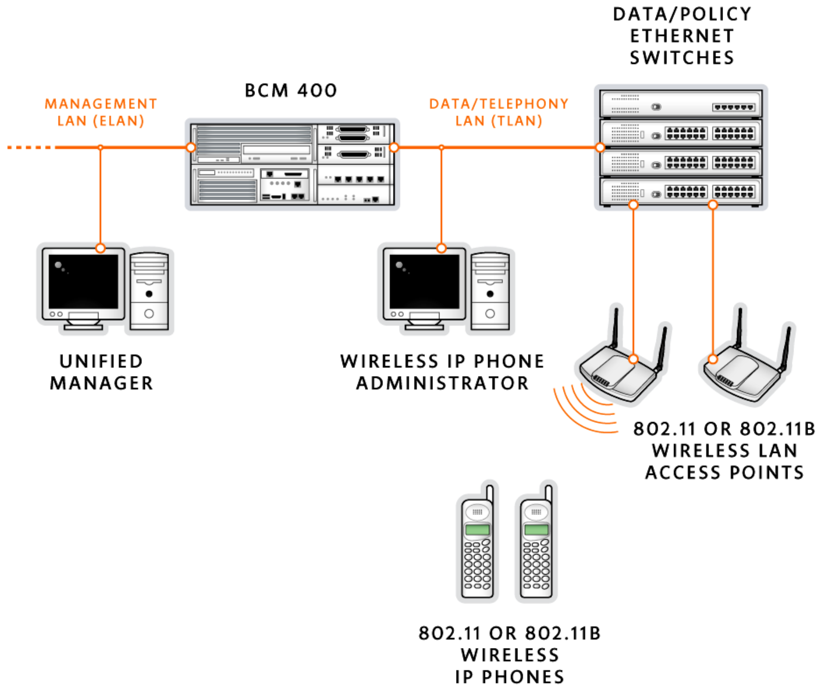
The BCM 802.11 Wireless VoIP solution enables third-party H.323+ Wireless VoIP phones to deliver the feature-rich suite of BCM services over an IEEE 802.11-based Wireless LAN (WLAN) connection.

The BCM Wireless VoIP solution typically consists of the following main components (also depicted in Figure 10-1):

- BCM with H.323+ 802.11 Wireless IP Support – is a software application that runs on the BCM Windows NT platform, providing an integrated H.323+ gateway (and limited Gatekeeper) functionality for the H.323+ Wireless IP Clients. This integration allows the H.323+ Wireless VoIP phones to have access to features and functionality available on traditional wired BCM telephone sets.
- Data/Telephony Local Area Network (TLAN) – includes data/policy/security Ethernet Switches.
- 802.11 or 802.11b Wireless LAN (WLAN) Access Points – are points of access to the TLAN/WLAN and defines the coverage area of the wireless VoIP solution.
- 802.11 or 802.11b Wireless IP Phones – are the H.323+ clients (end-user devices).

- System Management –is through Unified Manager, a web-based OA&M system hosted by the BCM. This provides the interface to administer the sets that will be supported by the BCM call-processing software
- Wireless IP Phone Administrator – is a standalone management tool specifically used to manage and configure the H.323+ NetVision handsets.

***Figure 10-1***




A key component of the BCM 802.11 Wireless IP offering is Symbol Technologies industry leading Spectrum 24 802.11 or 802.11b wireless LAN and NetVision H.323+ VoIP phones. Additional details on these products from Symbol Technologies can be found later in this chapter, at http://www.symbol.com or online at http://www.nortelnetworks.com/select.

# BCM Wireless VoIP Integration Functional Requirements

## Capabilities and Limitations of H.323+

H.323+ provides three main capabilities:

- Display control
- Audio control
- Keypad control.

### Display Control

This control provides the capability to write to the display.

#### *Default Symbol Phone Display Control*

The H.323+ Wireless IP Phone, by default, controls its own display to indicate:

- When a user is logged onto the phone
- When the phone is associated with an access point
- When there has been a successful registration transaction with a gatekeeper
- Incoming call identification from the Q.931 call setup message
- Standard H.323 call progress messages.

#### *H.323+ Enhanced Display Control*

H.323+ allows the BCM to update the display with call identification for outgoing calls, feature prompts, background displays, such as "Do Not Disturb," call forwarding information and time and date display.

In the event that the H.323+ Wireless IP Phone loses radio contact with the system, it will resume local control over its display to inform the user of its status.

The H.323+ Wireless IP Phone may go out of range of an access point, or be powered down and back up (a rare occurrence for wireline sets). If this occurs, users will need to maintain a copy of the display buffer in the BCM gateway application in order to refresh the current display when the H.323+ Wireless IP Phone re-registers with the system.

#### Display Size and Mapping onto M7100 Display Characteristics

Current H.323+ Wireless IP Phone display sizes are nominally two line by 10 characters, while that of the M7100 is one line by 16 characters. The current design is to parse the 16 character display message from the call server and present it intelligently on the 2 x 10 character display of the H.323+ Wireless IP Phones by wrapping on word boundaries. Words or strings longer than 10 characters will simply be split midway through.

#### Message Waiting Indication (MWI)

MWI can be provided in two forms. Since the M7100 does not have a separate MWI lamp, the regular character display indicates the presence of a voice message. With the H.323+ wireless IP station-side support, there is also information available from CTE to indicate the message waiting status. This information can be used to turn on an icon on the H.323+ Wireless IP Phone, leaving the display free for other background messages as described above.

### Audio Control

This control gives the user audio feedback when the H.323+ Wireless IP Phone is not idle.

#### Default Symbol Audio Control

The H.323+ Wireless IP Phone, by default, does not provide any tone generation outside the context of call progress.

- Audible alerting on incoming calls
- Call busy tone
- Call setup fail indication
- Out-of-range indication in an area with no radio coverage.

#### H.323+ Enhanced Audio Capabilities

The ability to provide tones while the phone is in the context of a call provides a mechanism to indicate that a second call is being presented. The only additional tones supported by the current H.323+ Wireless IP Phones, however, are call-waiting tone and message-waiting tone.

#### *Alerting Tone Attenuation*

The H.323+ Wireless IP Phone firmware controls attenuation of tones to avoid the possibility of presenting full alerting when the terminal is placed up against a user's ear.

### Keypad Control

This function allows the user to send keypad presses from the terminal to the proxy either in or out of the context of a call.

#### *Default Symbol Keypad Operation*

The H.323+ Wireless IP Phone, by default, sends key presses within the context of a call. They are sent as the dialed digits in a Q.931 call setup message, or during a call, as user indication messages.

#### *H.323+ Enhanced Keypad Operation*

This function allows users to put the H.323+ Wireless IP Phone into a mode in which key presses are sent to the terminal proxy even when there is no call active. This action allows users access to features, such as Call Forward or Do Not Disturb, that are normally activated from an idle terminal without first having an active call.

There is no mechanism for directly initiating a call from a feature such as System Speed Dial, or Voice Mail access via Feature 981. Enhancements have been made to the Function menu on the H.323+ Wireless IP Phone for each feature whether it is to be initiated on-hook, during a call, or off-hook with no call.

#### *Hold Key Configuration*

The M7100 set model on which the H.323+ Wireless IP Phone is based uses the Hold key to toggle between calls. If a user presses the Hold key during an active call, the key will place the call on hold and present dial tone. If a second call is being presented, the Hold key will allow the user to answer that call.

The H.323+ Wireless IP Phones have a configurable Hold key which can either toggle between two H.323 call sessions, or send a digit sequence to the far end with which it is in session. The latter model is used for BCM, allowing it to determine the second call context, since there is only one H.323 session allowed between the BCM and an H.323+ Wireless IP Phone.

#### *END Key Operation Change*

To allow the call-processing software to manage multiple calls over a single H.323 session, the H.323+ Wireless IP Phone does not terminate a call when a user presses the END key. An END key event, instead, is sent to the controller, at which point the call processing system decides whether the call should be dropped, connected to a second call on hold or ignored.

## BCM Feature Delivery

This support is provided as part of the BCM H.323 gateway, which includes direct media path negotiation.

BCM feature delivery is a Windows NT Application software deliverable. The wireless IP phone needs to support the H.323+ messaging protocol to ensure that the BCM application software can provide full BCM feature implementation.

### Feature Support with M7100 Set Model

The following list highlights some of the more common BCM features available in the H.323+ wireless IP offering. These features are part of the suite supported on the M7100.

- Calling Line ID
- Called/Calling Party Name Display
- Hold
- Conference Call
- Call Transfer
- Overhead Paging
- Call Forward All Calls
- Do Not Disturb
- Voice Mail box for set
- Voice Mail Access
- Ring Again
- Answer Key (twinning)

- Last Number Redial
- Speed Dial
- Dial Access to Group Call
- Call Park and Retrieve
- Message Waiting Indication
- Two-line call model/Call Waiting
- Automatic refresh of background display upon successful registration attempt by phone.

### FTE Permanence

In order to provide all of the features in the list, it is necessary to maintain persistent terminal proxy instances for each set operating within the BCM call-processing environment. Without this persistent terminal proxy, if a portable phone were turned off, powered down, or taken out-of-range of the access points that provide service to its home system, features such as Voice Mail and Call Forward would not work. There would also be an unacceptable delay after H.323 registration before the set could be used, as it takes roughly one minute to start a call-processing proxy entity to handle a set's signaling.

The starting of a terminal proxy occurs when it is administered into the BCM using Unified Manager. Although the proxy will not be usable until a successful H.323 registration has been performed, it will be available to the call-processing side to provide call redirection treatment, as though it were a wireline set.

### Idle Display Control

Whenever the H.323+ Wireless IP Phone has registered with the BCM and is within radio range, the BCM assumes control over the phone's display. The display on the H.323+ Wireless IP Phone will emulate that of the equivalent M7100 wireline set, with the exception that the physical layout is different.

#### *Battery Life Consideration for Time and Date Display*

The BCM core software maintains a current time display on all sets and is updated every minute.

### Basic Call Operation

Examples of an incoming and an outgoing call are included here. As a convention, incoming means that the call originated on the BCM and terminated on the H.323+ Wireless IP Phone. Outgoing means originated by the H.323+ Wireless IP Phone towards the BCM.

***Incoming Call***

***Idle screen display***

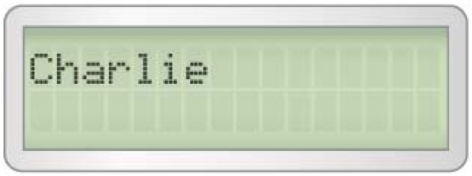
***Figure 10-2***


***Call presented***

***Figure 10-3***


***Call answered – press SND key***

***Figure 10-4***



***Call released – press END key or far end releases***

***Figure 10-5***

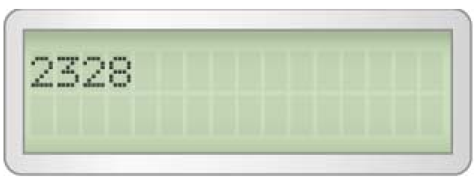

***Outgoing Call***

***Idle screen display***

***Figure 10-6***


***Number dialed – characters echoed to screen locally by phone***

***Figure10-7***

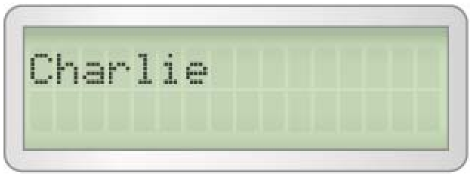



***Press SND – Call presented to far end***

***Figure 10-8***

***Call answered***

***Figure 10-9***



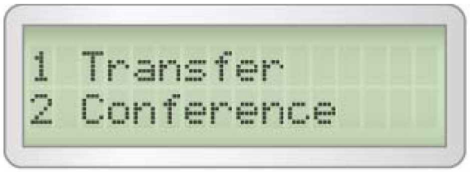

***Call released***

***Figure 10-10***


### Feature Activation

BCM features are available from the H.323+ Wireless IP Phone via the FCT key or Function Menu item. The H.323+ Wireless IP Phone function menu can be programmed to send A,B,C or D as well as 0-9, * and #. The digit A is mapped to the BCM set “FEATURE” key, and B represents the Hold key. The top nine popular features have shortcuts associated with them. For example, when the user presses the FCT key on the set, the following list is displayed:

*Figure 10-11*




**NETVISION PHONE (NVP)**

THE DISPLAY SHOWS ONLY TWO LINES. THE USER CAN SCROLL THROUGH THE FEATURES


**NETVISION DATA PHONE (NVDP)**

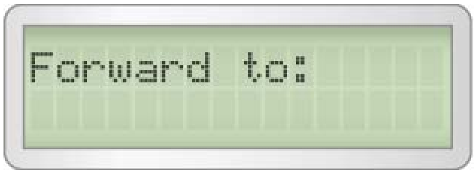
THE NVDP DISPLAY IS LARGER AND CAN SHOW MORE LINES OF INFORMATION. THE USER SIMPLY SCROLLS TO THE FEATURE THEY WISH TO ACTIVATE.

Current H.323+ Wireless IP Phone displays with only two lines allow the user to scroll through the features. Some features require the user to be in a call (e.g. Call Transfer and Conference), and others are allowed outside of a call (e.g. Call Forward.)

***Call Forward Example***

1. The user selects FCT->Call Forward->SND, or simply FCT->3
2. The H.323+ Wireless IP Phone will be programmed to transmit 'A' (Feature key) + 4 (standard BCM Call Forward feature code)
3. The user is presented with the following display message:

*Figure 10-12*

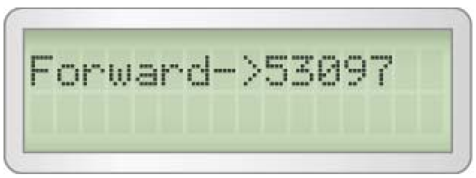


4. As a user enters each digit, the BCM call-processing software displays it on the screen:

***Figure 10-13***

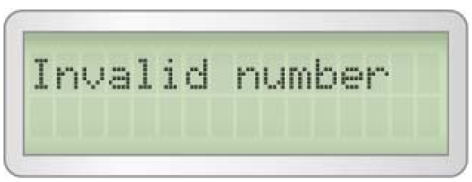

5. The user then enters digits that are passed back to the core for parsing. As soon as a valid DN is recognized by the core parser, the display will indicate:

***Figure 10-14***




6. If the digits cannot be parsed, the following error message displays for three seconds:

***Figure 10-15***




7. Because all of the keypresses and display updates are fully interactive with the call-processing software, there is no requirement to ask the handset to collect digits, then pass them on after the user presses the SND key.

***Call Transfer Example***

1. The user has an active call with “dest 1”
2. The user selects FCT->Transfer->SND, or simply FCT->1->SND
3. The Symbol set will be programmed to transmit A (Feature key) + 70 (standard BCM Call Transfer feature code)
4. The “dest 1” call is put on hold by the BCM core software. The user is presented with the following display message:

***Figure 10-16***


5. The user then enters digits that are passed back to the core for parsing and echoed on the set display.

***Figure 10-17***


6. As soon as a valid DN is recognized by the core parser, the display will indicate the destination identity:

***Figure 10-18***


7. Upon the “dest 2” set answering, the display will indicate the “dest 1” and “dest 2” DNs as follows:

***Figure 10-19***


8. If the digits cannot be parsed, the following error message is displayed for three seconds, before returning to the “Transfer To:” display.

***Figure 10-20***


9. Because all of the keypresses and display updates are fully interactive with the core, there is no requirement to ask the handset to collect digits, then passes them on after the user presses the SND key.

# BCM Software Compatibility

The minimum release of BCM Software required to support the Wireless Voice over IP Gateway is BCM load with H.323+ Wireless IP Phone support in Feature Pack 1.

## BCM Gateway Application Software

### Integration into Existing BCM VoIP Gateway

The existing VoIP trunk gateway application on BCM runs as a Windows NT service. This application will be expanded in capabilities to handle the H.323+ Wireless IP Phone, using the same RadVision H.323 stack as the H.323 Trunk side gateway and will provide the interface to the BCM core for making and receiving voice calls, activating and deactivating BCM call-processing features.

### Set Emulation for Call-Processing

The BCM call-processing software treats the sets as M7100 proprietary digital sets, with some minor changes to the MSC software to recognize that they are IP sets. Since they do not have TDM based media channels, the sets require that media control information be sent to them.

### Existing Gatekeeper Functionality

A gatekeeper agent is responsible for locating and registering with other gatekeepers on the network, and sending location requests to them in order to coordinate building a larger compound system to route calls within an H.323 network.

### 802.11 Wireless VoIP Gatekeeper – BCM GA

A basic gatekeeper functional unit can provide RAS services for H.323+ Wireless IP Phones. In order to populate the BCM routing tables with the transport addresses of the H.323+ Wireless IP Phones, the H.323+ Wireless IP Phones accept registration requests. Admissions requests are also supported, since the H.323+ Wireless IP Phones require an Admissions Confirm at the beginning of each call to proceed.

### 802.11 Wireless VoIP Gatekeeper – Feature Pack 1

This gatekeeper supports the proprietary messaging interface to allow keypad and display control of the H.323+ Wireless IP Phones. This gatekeeper handles this support because the message used to transport the H.323+ is the H.225 LRQ message. This allows a unified way to transmit messages both in and out of the context of a call.

# BCM Wireless VoIP H.323+ Clients

Over the past several years Nortel Networks has worked closely with Symbol Technologies to tightly integrate their NetVision(r) Wireless VoIP phone offering with several Nortel Networks platforms, including BCM. Below are further details on which products from Symbol have been tested with the BCM Wireless VoIP solution.

## Compatible Third-Party Wireless VoIP Phones for BCM

The following vendor has, or is in the process of, verifying its H.323+ Wireless IP Phones for compatibility with the BCM Wireless VoIP solution.

***Table 10-1***

| Wireless IP Handset | Code | Vendor | |
|---|---|---|---|
| NetVision (Phone Kit – 2 Mbps Frequency Hopping Spread Spectrum – 802.11 (HAC Compliant) | NV-9040-103-US | Symbol Technologies Inc. www.symbol.com | NORTEL NETWORKS™ Select Product |
| NetVision (Phone Kit – 11 Mbps Direct Sequence Spread Spectrum – 802.11b (HAC Compliant) | KT-NV-4046-300-WW | Symbol Technologies Inc. www.symbol.com | NORTEL NETWORKS™ Select Product |

For current status and ordering information, please refer to the vendor. For information on the Nortel Networks Select Product Program and Compatibility Testing, please refer to www.nortelnetworks.com/select.

## NetVision® Phone

The NetVision Phone from Symbol Technologies adds VoIP communications to Symbol Spectrum24® 802.11 frequency hopping or High Rate 802.11b direct sequence wireless LAN installations, allowing simultaneous voice and data support on the same wireless backbone. The NetVision Phone attributes are listed in Figures 10-21 and 10-22.

***Figure 10-21***

Symbol NetVision Phone for 802.11 FH Wireless LAN systems: NV-9040-103-US

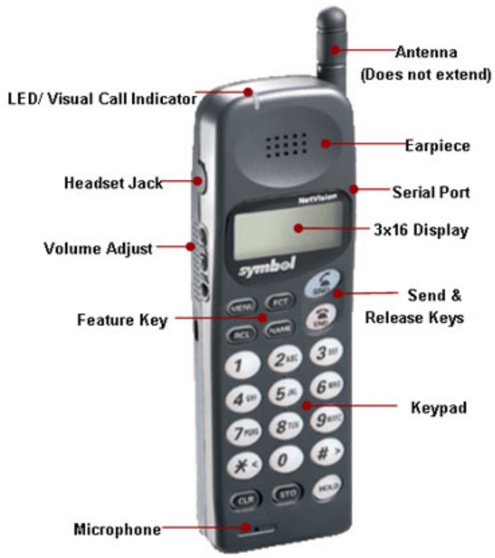


***Figure 10-22***

Symbol NetVision Phone for 802.11b "Wi-Fi" DS Wireless LAN systems: KT-NV-4046-300-WW (Note that this product replaces the previous Symbol NetVision Phone KT-NP-4046-100-US. Both products are compatible with the BCM wireless VoIP solution.


For further information and specifications on Symbol Technologies NetVision(r) product offerings, please refer to their website, http://www.symbol.com.

## NetVision DataPhone

Symbol Technologies offers a NetVision DataPhone for 802.11 FHSS customers (the DSSS version is not yet a released product) that is supported on BCM systems that are sold and used outside of North America. Due to FCC and Industry Canada regulations in North America, the NetVision DataPhone (FHSS) is not supported on BCM systems as it does not comply with FCC part 68, section 317 and CS-03, Part V, for hearing aid compatibility and volume control/ROLR regulations.

For further information and specifications on Symbol Technologies NetVision DataPhone product, please refer to their website http://www.symbol.com.

# System Performance Requirements

The BCM treats the H.323+ Wireless IP Phones the same way it treats the i2004 IP clients. Any limitations on resources or licenses will be shared with the i2004 clients. Certain resources, such as gateway ports, are also shared with IP trunks and may vary on a call-by-call basis depending on whether the call is terminated on an on-core TDM device.

# Voice Quality

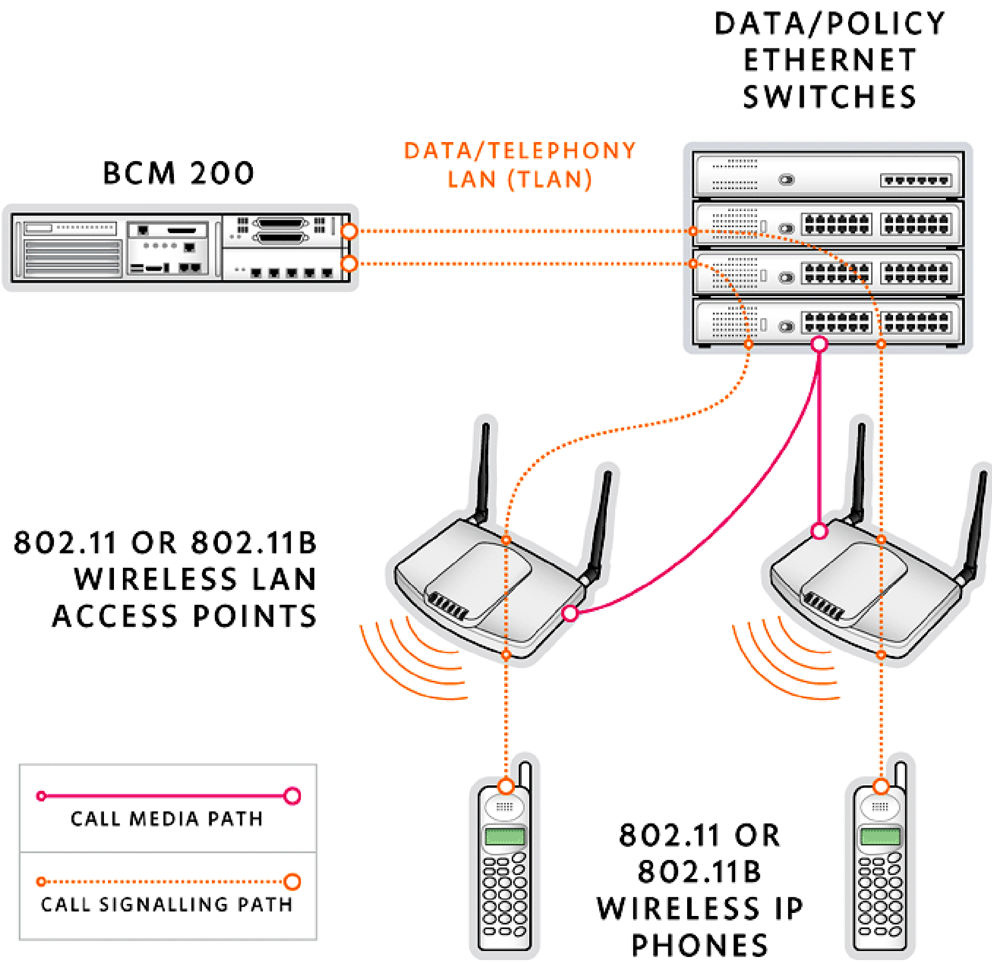
## Quality of Service (QoS) Mechanism

The current QoS mechanism is based on priority of UDP packets within the certain 802.11 Wireless LAN Access Point (AP). Symbol Technologies Spectrum24 Wireless LAN offering supports this capability within each of their APs, allowing voice to be prioritized over data.

## Media Path

BCM introduces direct IP to IP set media path. This means that the BCM does not need to perform double-jitter buffer and double-codec translation. The net effect is less CPU overhead for calls that do not terminate on the core, less delay in the speechpath and less network traffic for IP to IP calls.

*Figure 10-23*




# Wireless LAN Access Points

One advantage of the BCM Wireless VoIP solution is that it can leverage a standard's base IEEE 802.11 (FHSS) or 802.11b (DSSS) "Wi-Fi" Wireless LAN infrastructure. While this capability allows for customers to have a broad choice in selecting a wireless LAN vender, it is important for customers to be aware that the current IEEE 802.11 or 802.11b standard does not address several critical features for voice, enhanced security or mobility applications.

IEEE is in the process of introducing several enhancements to the 802.11 standard to overcome several of these shortcomings (i.e, 802.11e and 802.11i), but the following list of capabilities is recommended for to ensure the proper level of voice quality and mobility features when customers evaluate a Wireless LAN vendor for a wireless VoIP application.

- Provide QoS at the WLAN Access Point level – alleviates the possibility for serious contention issues when data devices/applications are present
- Ability to load balance voice and/or data users across multiple Access Points – maximizes bandwidth resources that can help QoS issues
- Offer power management options and features – increases battery life/talk and standby time numbers for the wireless VoIP phone and/or data NICs
- Support multiple security options – ability to support both standard security services (WEP) and more advanced services (i.e. Kerberos) that can provide suitable encryption capabilities.

An example of a vendor that offers the above features and capabilities in their Wireless LAN product is Symbol Technologies' Spectrum24(r) Wireless LAN Portfolio.

***Figure 10-24***

Symbol Technologies Spectrum24 802.11 FHSS and Spectrum24 High Rate 802.11b DSSS Access Points

## Compatible Third-Party Wireless VoIP Access Point for BCM

The following vendor has, or is in the process of, verifying its 802.11 WLAN Access Points for compatibility with the BCM Wireless VoIP solution.

***Table 10-2***

| Wireless LAN Access Point | Code | Vendor | |
|---|---|---|---|
| Symbol Spectrum24 802.11 Frequency Hopping Spread Spectrum Wireless LAN Access Point – Plenum Rated (2 Mbps – 100mw) | AP-3021-100-US | Symbol Technologies Inc.<br>www.symbol.com | NORTEL NETWORKS Select Product |
| Symbol Spectrum24 802.11 Frequency Hopping Spread Spectrum Wireless LAN Access Point – Plenum Rated (2 Mbps – 500mw) | AP-3021-500-US | Symbol Technologies Inc.<br>www.symbol.com | NORTEL NETWORKS Select Product |
| Symbol Spectrum24 High Rate 802.11b Direct Sequence Spread Spectrum Wireless LAN Access Point (11Mbps – Trilogy 2) | AP-4121-1150-WW | Symbol Technologies Inc.<br>www.symbol.com | NORTEL NETWORKS Select Product |
| Symbol Spectrum24 High Rate 802.11b Direct Sequence Spread Spectrum Wireless LAN Access Point- Plenum Rated (11 Mbps – Trilogy 3) | AP-4131-1050-WW | Symbol Technologies Inc.<br>www.symbol.com | NORTEL NETWORKS Select Product |

For current status and ordering information, please refer to the vendor. For information on the Nortel Networks Select Product Program and Compatibility Testing, please refer to www.nortelnetworks.com/select.

# Additional Resources for Symbol Product Offering within Nortel Networks Select Product Program

## North America: Contact Address

In North America, contact the following address for inquiries or information relating to Sales Support Services, Order Placement and Authorized Contact and Customer and Technical Support Services:

200 Athens Way, C601
Nashville, TN 37228
615-432-4995
fax: 615-432-5116
http://www.selectproduct@nortelnetworks.com/

## Wireless LAN Surveys – Do You Need a Wireless LAN Site Survey?

Wireless LAN Site Surveys differ in their complexity and level of effort based on the technology and space being surveyed. Small facilities or small applications potentially do not require one at all; customers can rely on the component specifications and a good evaluation of the facility blueprint.

However, for larger or complex installations, especially ones with mission critical applications, the site survey offers a level of security in this arena in which – unlike wired networks – there are many variables to consider. Some of the items needed to help ensure a successful Wireless LAN installation include:

- Identification area of wireless/mobile activity
- Measurement of radio characteristics for site-specific environments
- Survey and identification of power options
- Verification of host connectivity required
- Survey of existing network connections and existing equipment
- Analysis of results
- Design of network for coverage requirements
- Design of network as “legal” Ethernet segment

- Plan of integration into existing network
- Documentation of equipment placement, power considerations and wiring.

Site surveys provide these detailed specifications and serve as a guide for the network design and for installing and verifying the wireless communications infrastructure. A thoughtful, accurate and effective assessment of coverage and system throughput requirements will ease the wireless LAN installation process, while laying the groundwork for ongoing data and advanced voice applications networking requirements. Site Survey services are strongly recommended for facilities over 100,000 sq. ft.

## 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.

### 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 10-25***



# Computer Telephony Integration (CTI)

# Computer Telephony Integration (CTI)

## Chapter Highlights

- BCM Computer Telephony Integration (CTI) – provides businesses with the ability to turn a desktop computer into a powerful communications tool, connecting the intelligence of the PC with the power and flexibility of the BCM.
- Telephony Application Program Interface (TAPI) – is a standard program interface that enables computer users to use their PC to communicate over telephones or videophones to people or phone-connected resources elsewhere in the world. It converts industry standard commands from Microsoft Windows into a format that BCM TSP can understand and vice versa.
- Personal Call Manager (PCM) – is a screen-based telephony and contact manager application that brings the call control features of BCM telephones to the computer screen, offering features that allow users to increase productivity and improve customer service.
- TAPI Service Provider (TSP) – is based on the Telephony Application Programming Interface (TAPI) standard developed by Microsoft and is a simple and flexible tool for implementing Computer Telephony Integration through either a direct-connect or client/server configuration.
- Attendant Console – provides the capability to attach one or more Windows 95/98/Me/2000/NT 4/XP PCs to a BCM system for use by telephone system attendants.

# CTI Sales Scenario

## The Challenge

Miller & Reid is a medium-sized law firm specializing in all types of law. With 40 full-time lawyers, Miller & Reid is an extremely busy firm whose client base is constantly growing. Miller & Reid realized that in order to accommodate its growing client base, improve efficiency and provide better client service, it needed to take advantage of the advanced communications solutions available on the market.

Miller & Reid has a front-desk attendant who answers incoming calls and transfers calling parties to the appropriate destination. Before the implementation of an effective communications solution, Miller & Reid's attendant would notify called parties of incoming calls by paging them. This method often posed a problem, as lawyers were frequently busy or already on a call with a client. The paging system began to cause too many interruptions.

Given that Miller & Reid has many clients and the firm receives a large volume of calls on a daily basis, the attendant found that she was overwhelmed and was simply too busy to handle all of the incoming calls. Clients were becoming frustrated, as they would be placed on hold for long periods of time and would often get disconnected. Lawyers were also becoming annoyed, as they were missing important calls.

While the firm realized that it needed to make significant changes, it did not want to eliminate the need for the front desk attendant. It needed a solution that would relieve the attendant of some of her call answering duties and allow the lawyers to be more productive with respect to call handling.

## The Nortel Networks Solution

The solution for Miller & Reid was Personal Call Manager combined with Attendant Console. Miller & Reid decided to leverage its investment in the BCM and take advantage of its CTI applications by purchasing LAN (Local Area Network) CTE (Computer Telephony Engine) seat licenses to activate Personal Call Manager. In addition, Miller & Reid purchased a software keycode to activate the pre-installed Attendant Console application.

Personal Call Manager (PCM) is an award-winning TAPI-based telephony application designed for use with Windows 95/98/2000 NT operating systems. With Personal Call Manager, a user can access an internal Address Book and double-click on a name to dial, conference or transfer calls. PCM also allows users to answer calls and see call activity – all on their PC screen.

The Attendant Console functionality is an appropriate solution for businesses that wish to utilize a live attendant to answer incoming calls and increase attendant productivity, as well as provide better service to its callers. Computer Telephony Integration (CTI) simplifies call processing because the graphical user interface enables an attendant to answer and manage call flow with point-and-click convenience, rather than using a desktop phone.

The onscreen display shows a directory of all employees, so attendants can easily look up personnel by name or department. The display shows whether the person is already on the phone, if they do not want to be disturbed, or even if they are out of the office. The system offers more efficient call handling and improves customer service. Moreover, the system supports multitasking, enabling the attendant to use his or her PC for other applications and then bring up the Attendant Console application during a call. An online database also stores complete records of incoming and outgoing calls.

## Implementation

Attendant Console provides the capability to attach one or more Windows 95/98 NT PCs to a BCM system for use by telephone system attendants. To run the Attendant Console, the PC must comply with the following system requirements: Pentium processor, 32 MB RAM, 10 MB available disk space and a SVGA monitor with a minimum resolution of 800x600.

CTI applications like Personal Call Manager and Attendant Console are particularly useful for law firms because these businesses can integrate the applications with their billing systems. Lawyers can use PCM, for example, to monitor the time they spend on the phone dealing with clients. Implementation and integration with existing systems is usually simple, but is dependent on the type of billing system a business currently has in place.

## Business Impact

With Personal Call Manager (PCM), lawyers at Miller & Reid are now able to save time and increase productivity. They can activate the Do Not Disturb (DND) feature if they are too busy to take calls. With this feature the telephone will not ring, allowing lawyers to work undisturbed. They can use the Address Book feature to store telephone numbers and addresses of the clients or businesses they deal with most often; this capability saves time. Moreover, the Address Book allows them to sort or prioritize their contacts to meet their individual needs.

Attendant Console has significantly improved the front-desk attendant’s productivity. As Attendant Console works in a multitasking environment, the attendant can use her PC to perform other tasks such as word processing and then switch to the Attendant program when a screen pop notifies her of an incoming call.

Even before the attendant answers a call, the Attendant Console provides caller information to assist the attendant in handling the call. For example, the graphical user interface (GUI) informs the attendant of the caller’s telephone number, the type of trunk group the call came in on and the amount of time the caller has been waiting – enabling the attendant to answer the call more professionally and efficiently.

The GUI also provides information not normally available with hardware-based consoles. For any incoming call, the Attendant Console provides the attendant with the names of the lawyer or lawyers the caller most often calls and the status of each lawyer's telephone. The attendant is immediately able to see whether the target extension is busy and, if so, advise the caller of alternative employees for them to be connected to.

CTI integration has boosted productivity by putting users in control of all aspects of their voice communications. Additionally, with both Personal Call Manager and Attendant Console, the firm can now project a highly professional image. The firm has also realized significant cost savings, as lawyers are spending less time managing their phone activity.

# Overview

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, 32, 64 and Maximum seats. A try-and-buy keycode is also available for customers who wish to try LAN CTE before purchasing.

## Emerging Trends

Companies today are rethinking their business processes and exploring innovative ideas to increase efficiency and productivity, as well as to deliver a competitive edge. One of the most exciting of these ideas involves the convergence of telecommunications and multimedia technologies to create a powerful new kind of communication. The convergence of these technologies has created a new industry and made the promise of integrated multimedia communications a reality. CTI provides companies with the ability to turn a desktop computer into a powerful communications tool that can combine sight, sound, text, animation, video, graphics and other sophisticated telecommunications functions.

CTI has existed since the early 1980s and since that time has experienced an ongoing evolution in sophistication of features as well as cost benefits to users of systems employing this technology. In response to wider market opportunities, CTI software vendors have developed an extensive range of tools and applications that have made this technology more practical. As a result, many businesses are taking advantage of CTI benefits and are employing it in their business operations products.

## Benefits

The implementation of CTI helps agents deliver faster, more personalized customer service. Relevant information via screen pops and automated retrieval of database records allows a new call to be routed to a specific agent who handled that customer previously and with VoIP these transactions can be held over a single network infrastructure.

BCM offers CTI solutions that can enable a business to improve customer service, increase employee efficiency and generally reduce expenses. CTI enables a number of efficient functions in a business: intelligent call routing, screen-based telephony, intelligent dialing, automated display of information – based on caller-provided information from an Interactive Voice Response (IVR) or other interface and the coordinated transfer of data information along with a telephone transfer.

CTI offers companies the following benefits:

- Better customer service
- Increased effectiveness
- Reduced costs
- Multiple service levels.

# 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 Networks, 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 Networks, 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.

Several downloadable software components reside on a Windows NT server that is built in the BCM platform, eliminating the need for additional hardware equipment such as a Norstar CTA device. These software components include a number of applications, including BCM Personal Call Manager. Each software component has its own unique purpose that enables users to access CTI functionality. For instance, BCM LAN CTE enables TAPI applications, like Personal Call Manager, to communicate between the BCM platform and a desktop PC. LAN CTE allows the telephony application to communicate with and control the user's telephone.

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.

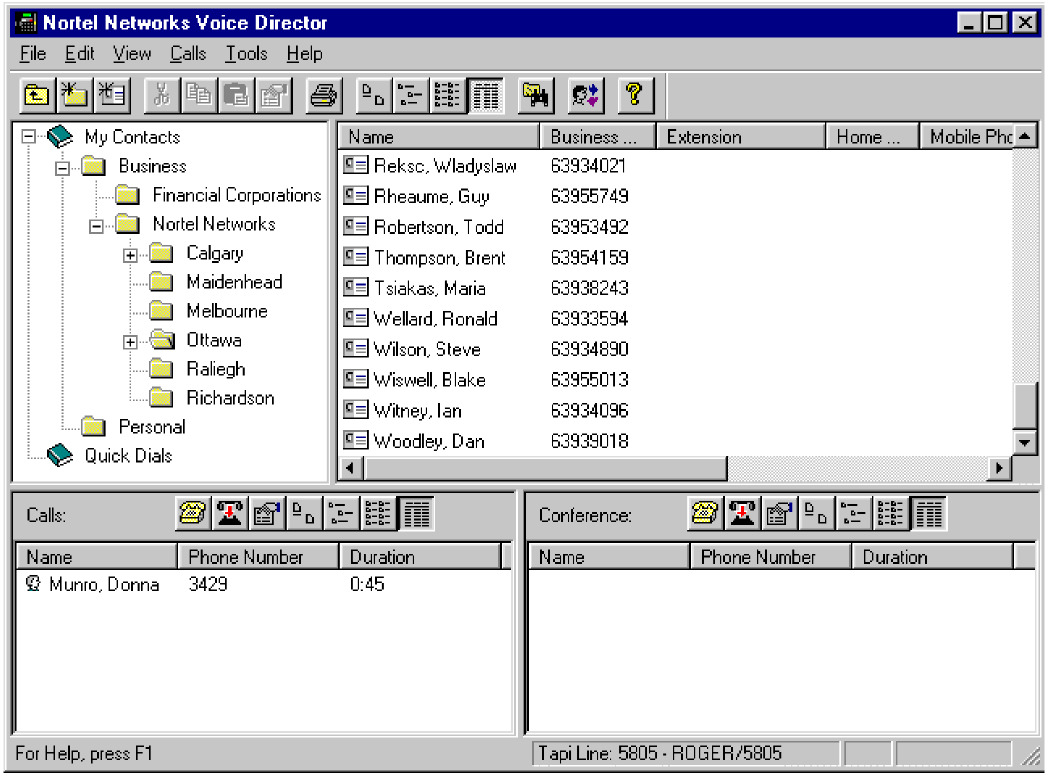
The BCM Personal Call Manager provides tools that make call processing easier. For this reason, a user's telephone must be equipped with the handsfree option or a headset for privacy. When using a handsfree telephone or headset, users can access all of the BCM call manager functions without picking up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

***Figure 11-1***




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).

## Target Market for Personal Call Manager

Where there is a person working at a desk with a telephone and a computer, there is an opportunity to implement PCM. Many businesses or individuals might not realize that they require CTI, as the myth that "CTI is only for call centers" is still prevalent in the marketplace.

Potential candidates for PCM are employees who:

- Often disconnect callers when they try to transfer calls
- Use a contact manager to look up a name and number, then dial the phone
- Shuffle through business cards, sticky notes or address books to find a phone number
- Dig through files and paper while talking to someone to see what they talked about last time
- Have both a phone and a computer on their desk
- Wish they could see who called but the caller did not leave a message
- Think they could use their time more efficiently
- Think they could justify an expenditure if the payback were less than one year.

An answer of "yes" to any one of the above questions indicates an opportunity to discuss CTI applications at the desktop and position PCM.

## Market Segments

The target market for PCM can be segmented into three distinct groups: knowledge worker, informal call center and formal call center. The knowledge worker and informal call center markets are the fastest growing segments of the CTI marketplace; these groups represent a significant opportunity for PCM.

### Knowledge Worker

Knowledge workers are typically employees who have both a telephone and a computer at their desk but are not part of a formal call center. This market includes all types of professionals working in offices, including accountants, lawyers and computer programmers.

In the past, the knowledge worker was not considered a candidate for CTI applications, as traditionally only businesses with call centers used this technology. Today, businesses are realizing that knowledge workers frequently talk to customers and, therefore, can benefit from the same kinds of CTI applications to which call center agents have access.

### Informal Call Center

This type of call center brings together the traditional formal call center, knowledge workers and SOHO professionals. While employees in informal call centers are not call center agents, they spend a good deal of their time answering the phone and responding to customer inquiries. Employees working in an informal call center often do not think of themselves as a call center, as they might not have an Automatic Call Distribution (ACD) system and they are not traditional agents dedicated to answering incoming calls. They may be part-time agents who perform other duties within the business.

Prospects for informal call centers are more difficult to identify, as the employees answering calls may not be identified as ‘agents’ or ‘telephone service representatives.’ An informal call center employee could be the local expert in a particular department who takes calls from specific customers, or during specific times of the day, in addition to performing other duties.

Typically, an informal call center involves using call center capabilities to provide an enhanced form of customer satisfaction or revenue achievement by making the most qualified persons available by phone.

### Formal Call Center

A formal call center exists when a company dedicates a group of employees to answering similar kinds of telephone calls. Individuals or agents within formal call centers specialize in answering incoming calls and concentrate on that activity. An ACD system manages incoming calls and distributes them to agents.

## Personal Call Manager Benefits

Implementing Personal Call Manager offers many benefits to both the knowledge worker and the formal or informal call center agent. PCM can increase productivity and enhance customer service as a result of the following factors:

- Incoming calls are displayed on the desktop PC through Calling Line ID, which shows the name or number of the calling party (user must subscribe to Calling Line ID service)
- Call control capabilities enable integration of telephony features such as dial, conference call and transfer integration into PC applications, reducing time and complexity
- Less time on the phone translates into reduced 1-800 charges and more time spent with potential new customers or selling additional products and services to existing customers
- Implementing CTI solutions in a client/server configuration reduces system administration and maintenance costs.

Managing telephony features through a simple drag and drop PC interface means call center agents can handle calls efficiently and professionally. Businesses in any industry can realize the benefits of PCM applications.

## 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.

## Attendant Console

BCM Attendant Console provides the capability to attach one or more Windows 95/98/Me/2000 and Windows NT 4/XP PCs to a BCM system for use by telephone system attendants. Attendant Console is an optional application that comes preinstalled on the BCM system. Users can activate this application by purchasing a software keycode.

The program provides a graphical user interface (GUI) for the easy handling of incoming call traffic and quick dispatch to the appropriate person or to voice messaging. Attendant Console runs with other Windows applications in a multitasking environment.

The primary attendant and backup or overflow attendants can be established wherever a Business Series Terminal is located and a local area network (LAN) connection is available. By purchasing software keycodes, businesses can add up to a maximum of four backup or overflow attendants to Attendant Console. With Attendant Console, the user can:

- Answer up to 144 incoming lines
- Recognize and label the calling party (note that target line names are not displayed on incoming calls)
- Access information about callers
- Add or change caller records
- Place numerous calls on hold
- See busy-line status of all digital/IP phones
- View a directory of all extensions, showing person and telephone status
- Park calls and access paging
- Transfer calls to voicemail (note that calls cannot be transferred to a guest mailbox – these calls can be transferred using the telephone)
- Direct calls to an extension and display the caller's name in the LCD window, even if the called party is on the phone
- Allow the called party to control the call using options available on the M7324, T7316, T7310, T7406 digital sets; and i2004/i2002/i2050 IP telephones and Softphone
- Provide management reports of how incoming callers are treated.

## Target Markets

The target market for Attendant Console includes any new customer purchasing a BCM, as well as any customer already owning a BCM.

The predominant market for the Attendant Console is higher line sized systems, or systems with high incoming call volumes. Another target market is businesses that want to use a dedicated front-desk primary answering position, either with or without a backup answering position.

Attendant Console is an appropriate solution for businesses that wish to:

- Have a live attendant handle incoming calls
- Provide better service to their callers
- Let employees know who is calling
- Avoid missing important calls
- Increase attendant productivity
- Let management know how well their agents are servicing callers.

## Attendant Console Benefits

Companies of all kinds are striving to offer world-class customer service, as well as understand the patterns of incoming calls in order to manage them appropriately. Additionally, in their attempt to offer a high level of customer service, many companies believe that they will differentiate themselves by having a live attendant answer all calls. Attendant Console is a good fit for all of these business requirements.

Increased performance and employee productivity are also important to the small site business and the GUI interface of the Attendant Console allows attendants to quickly and easily choose the necessary buttons and functions. The multitasking capabilities of Attendant Console also help maximize the productivity of the attendant, who also needs to perform other duties.

### Enhanced Customer Service

Even before attendants answer a ringing line, the Attendant Console can provide information that will assist them in handling the call. For example, the GUI can inform the attendant of the caller's telephone number, the type of trunk group that the call came in on and the amount of time that the caller has been waiting. This information allows the attendant to answer calls more professionally and efficiently.

The Attendant Console GUI can provide an attendant with further information that is not normally available with hardware-based consoles. For example, Attendant Console can provide the attendant with the names of the employees that a particular caller frequently calls, along each employee’s telephone status. The attendant can immediately see whether the target station is busy and, if so, he or she can quickly connect the caller to an alternative employee. When the system transfers calls back to the attendant (in situations where the target station user does not answer or requests the attendant’s service), Attendant Console can inform the attendant as to the status of the situation.

When calls are coming in too fast for the main attendant to handle, he or she can configure Attendant Console to automatically transfer calls to another attendant. If the Attendant is unavailable to take calls, Attendant Console can direct all calls to a backup attendant position. These are important and valuable features to the BCM customer because they ensure that calls are not lost and that callers are not frustrated by delays or poor service.

### Decreased Operating Costs

Attendant Console requires a Pentium-based PC running Windows 95/98/Me/2000/NT 4/XP. Businesses do not require a proprietary monitor and computer hardware to run Attendant Console, resulting in minimal additional equipment on the attendant’s desk. Since Attendant Console is software-based, businesses can realize a lower entry cost to obtain it than with competitive products that require them to purchase a proprietary computer and keyboard.

### Improved Employee Productivity

Because the Attendant Console only displays buttons that specifically relate to a current call, the GUI appears less cluttered and complicated than a typical attendant console-type application. Moreover, since there are fewer buttons, an attendant can quickly and easily select those features necessary to handle any particular call.

Attendant Console works in a multitasking environment; this capability means that attendants can use their PC to perform other tasks, such as word processing and then quickly switch to the Attendant program when a screen pop notifies them of an incoming call.

The Attendant Console not only increases the personal productivity of the attendant, but also that of employees who receive calls. Attendant Console can communicate with an employee's telephone set, even when an employee is busy on another call; for example, it can send caller information to the two-line LCD display window on the employee's set. The employee can then decide what to do with the call and control it accordingly, by using soft keys. These capabilities allow employees to increase productivity because they do not have to leave or disrupt their current call.

## Attendant Console Configuration

Attendant Console is flexible enough to allow a variety of different configurations and work styles. In the most basic setup, it can assign a single attendant position to handle all incoming calls. A second alternative is to assign a "backup" or "overflow" attendant, to assist the main attendant. In this situation, the main attendant can choose to redirect incoming calls to the backup attendant. This feature is extremely useful because it does not require the backup attendant to physically relocate to the main attendant's desk. The Attendant Console Server software resides on the BCM system. The Attendant Console software and Reports software are installed on each Pentium-based PC on the LAN that is being used by an attendant.

## Attendant Console Architecture

The Attendant Console consists of three interrelated components:

- Server software
- Attendant software
- Reports software.

### Server Software

This software comes installed on every BCM system. Once a business or user activates a software keycode, the Server software receives notification of all incoming calls, outgoing calls and change of status of all telephones. The Server software can also request that the BCM transfer calls, place calls, park calls and perform other telephone functions. The Server software can support multiple attendant and assistant positions.

### Attendant Software

This software provides a PC-based GUI to perform the same functions as a telephone when an attendant is answering and handling incoming calls. System administrators can install Attendant software on each PC on the LAN that is being used as an answering position by a primary, backup or overflow attendant, up to a maximum of five PCs.

### Reports Software

This software automatically collects information on incoming calls and tracks how callers are treated. Attendants or supervisors can use this important information to spot trends and provide answers to questions before they become problems. Attendants can request reports from any Attendant or Assistant Console to view on a PC screen, send to an attached printer or export to other applications for further data manipulation. Attendant Console reports include:

- **Calls to Employee Report** – shows the types of calls (personal, customer, vendor, etc.) employees are receiving over a defined time period.
- **Calls by Customer Report** – shows how employees have been handling incoming calls (call taken by employee, call taken by assistant, call routed to voicemail or call routed to operator.
- **Extension Directory Report** – lists all employees that are in the Attendant Console database along with their department information.

## Call Handling

The attendant and assistants have access to many features, all of which are executed through the Attendant Console main window. The primary features of the main window are listed below.

- Answer calls
- Enter caller information
- Find an extension
- Transfer calls
- Handle callbacks
- Visual Call Announce in the Business Series Terminal’s telephone LCD window
- Call Control by the called party
- A caller database

- Automatic reporting on incoming calls
- Do Not Disturb icons (indicate when an employee's extension is set to DND)
- Call Forward icons (indicate when an employee has call forwarded to another extension)
- Hunt Group icons (the color of the regular phone icon will appear as blue if that extension is part of the Hunt Group).

**Note**: Visual Call Announce and Call Control are only available on M7324, T7316, T7316, T7310, T7406 digital sets; and i2004/i2002/i2050 IP Telephones and Softphone.

In the main window, the Attendant screen displays information about the incoming call, including the caller's name, the company name and the name of the person the caller normally phones. The Attendant screen also shows a Company Directory with employee telephone status and personal status, including "away from office" or "away from desk."

The attendant can also perform a search for employees by name or department in the Company Directory. Once the system selects the employee's extension, the attendant can transfer the call to their telephone, voice mailbox or to another number (e.g. home, cellular, pager).

**Note**: To transfer a call to a Guest mailbox, the attendant must use the Feature key on their telephone.

Using a Windows 95/98/Me/2000/NT 4/XP-compatible PC, the attendant can type the caller's name and the system will transfer the call to the appropriate party. The system only requires the attendant to enter the caller's name the first time; in subsequent calls, Attendant Console will 'remember' the caller's name and link it to the CLID.

The attendant can also park the call automatically, page the called party and view parked calls to see details of all calls parked on the system. Or, the attendant can process calls according to 'memo' instructions related to the caller. The main window displays the current availability of employees. The Notes field, for example, identifies if an employee is in a meeting or out of the office. When they are not covering calls, attendants can be working on other tasks on their PCs.

### The Assistant

The assistant performs the same functions as the attendant when answering incoming calls. The assistant is used to cover incoming calls for an individual, a work group or a department. On a Windows 95/98/Me/2000/NT 4/XP-compatible PC, a call notification dialog box pops up on the main PC window to alert the assistant that a call is ringing the covered telephone set. This notification lets the assistant pick up the ringing call if the called party does not answer. The assistant can easily add or remove telephone set(s) to be covered by selecting the extension(s) from the assigned tab.

#### Visual Call Announce

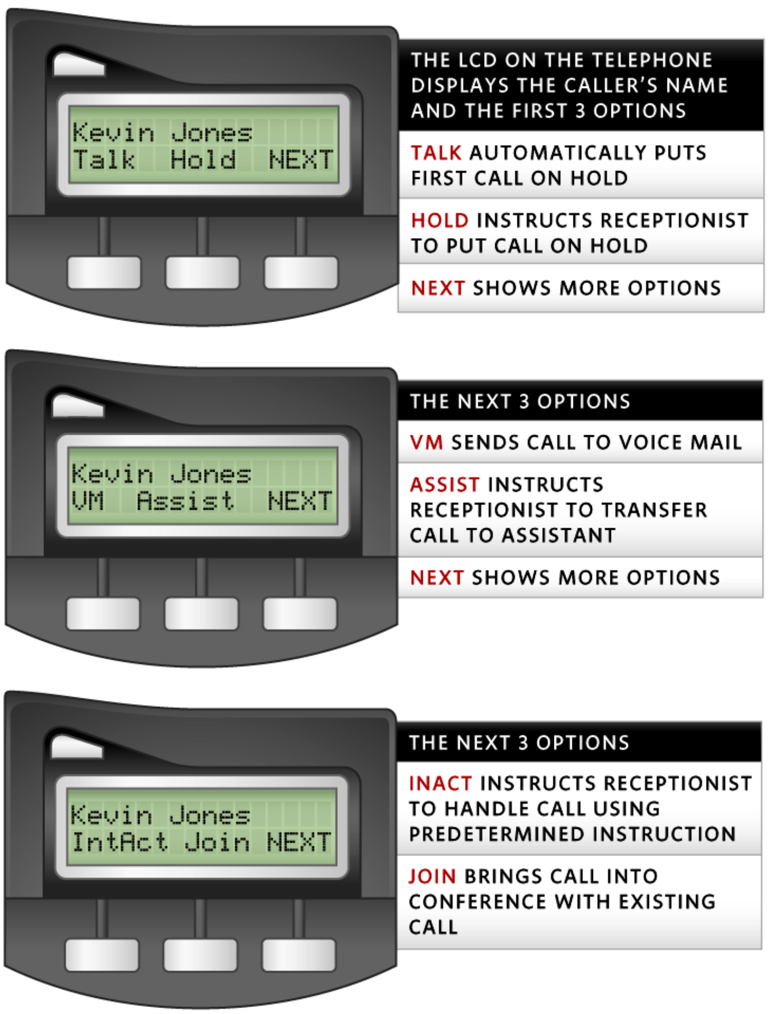
Once the attendant has routed the call to the extension, a low-level audible tone will alert the called party, even if they are on their phone. The caller's name displays in the LCD window M7324, T7316, T7316, T7310, T7406 digital sets and i2004/i2002/i2050 IP Telephones and Softphone.

In a busy office environment, Visual Call Announce provides more privacy and less noise than paging.

#### Call Control

Attendant Console lets called parties determine how they want to handle incoming calls, even when they are already occupied on a call. Rather than having all calls automatically forwarded to voicemail when they are busy, employees now have the option of managing their calls using the three interactive soft keys below the LCD window of their M7324 or T7316 telephone.

*Figure 11-2*




By pressing an interactive soft key beneath the LCD Window of the M7324, T7316, T7316, T7310, T7406 digital sets; and i2004/i2002/i2050 IP Telephones and Softphone, the following options are available:

- Talk to the caller by pressing the “Talk” soft key
- Ask the caller to hold by pressing the “Hold” soft key, which delivers a preset message to the attendant, who can advise the caller of the hold request
- Automatically transfer the caller to voicemail by pressing “VM”

- Send the caller to an assistant by pressing the “Asst” soft key, which delivers a pre-set message to the Attendant, who can speak to the caller and transfer them to the assistant
- Send the call back to the attendant by pressing the “Interact” soft key, which delivers a pre-set message to the attendant informing them to speak to the caller
- Join the caller with the current call by pressing the “Join” soft key.

If the called party is on the phone, the name of the new caller will overwrite any information in the LCD window. The telephone’s LCD window displays the new caller’s name for all calls transferred from the Attendant Console but will disappear in approximately 20 seconds.

### Caller Database

One of the benefits of the Attendant Console is that businesses can install it as a passive system, meaning that it can work in the background without affecting the way BCM operates. Thus, Attendant Console can build a database on all incoming caller information.

The Attendant Console will automatically create a caller database by collecting the following data elements on incoming calls:

- Date and time of day
- BCM line on which the call was received
- Caller’s name (name and company from Caller ID or entered by using the PC’s keyboard)
- Caller’s telephone number (from Caller ID)
- Extension of the person called
- Whether the call went to a voice mailbox or was referred to an assistant
- Duration of the call (minutes and seconds).

To identify future callers, Attendant Console automatically builds a database of caller records based on Caller ID and the attendant’s keyboard entry into the Caller ID fields. This capability makes the attendant’s job easier and more efficient, since a name is entered only once rather than every call. The Attendant Console database automatically downloads extension data (name and/or number) from the BCM.

Using the Edit function in the Caller Information group box, attendants can enter information into the caller database by filling out a simple, Windows-oriented data entry form. The caller database provides an important tool for management to see how callers to their organization are being treated.

The caller and employee database is a Microsoft Access 97 database that users can open and manipulate with any application compatible with Microsoft Access 97 databases.

# PC System Requirements

Whether users are installing both the Attendant and Reports programs or just the Attendant program on a PC, they require the following PC hardware and software:

## Minimum Hardware Requirements

- Pentium-based PC
- 32 MB of memory or greater
- 10 MB available disk space
- SVGA monitor with a minimum resolution of 800x600 and .28 dot pitch or smaller
- Windows-supported keyboard and mouse
- Windows-supported printer (optional, for report printing)
- PCI-bus Network Interface Adapter, 10/100 MB Ethernet.

**Note**: Excludes NE2000-Class cards.

## Minimum Software Requirements

- Windows 95/98/Me/2000 or Windows NT 4.0/XP Operating Systems
- Attendant Console Server program (included in Attendant Console product).

## Compatibility

Attendant Console is compatible with the following:

### Attendant Telephones

- M7324
- T7316
- T7310
- T7406
- i2004
- i2002
- i2050

### Visual Call Announce and Control

- M7324
- T7316

### Monitored Telephones

- Business Series Terminals:
    - T7100
    - T7208
    - T7316
    - M7324 (including CAP, if present)
    - M7310
    - M7208
    - M7100

# Nortel Networks Developer Program

Nortel Networks is creating a new era of Network Solutions and is working closely with innovative companies to provide a unified experience for customers. Standards and open interfaces on Nortel Networks platforms have created limitless opportunities for developers of enhanced solutions and the Nortel Networks Developer Program has been designed to maximize these efforts.

In order to take full advantage of today's opportunities, developers of complementary solutions need a way to communicate, not only to their end customers, but also with the manufacturer of the equipment with which their product must integrate. These developers also require simplified access to a broad range of open interfaces, information and technical support services. Finally, they need to develop effective ways to present these solutions to the marketplace, whether directly to the end customer or through a well-established distribution network.

## Overview

The Nortel Networks Developer Program is a global initiative. It offers technology businesses that develop value-enhancing applications for Nortel Networks product platforms the means to work together in providing total solutions to our mutual customers.

The top tier of this program, Developer Partner, provides companies with an extended and recognized relationship with Nortel Networks. These companies have tested their solutions for compatibility in a Nortel Networks lab environment and have taken further ownership of their relationship with Nortel Networks by meeting additional membership criteria and being approved by Nortel product owners. Our Developer Partners deliver solutions that increase our customers' ability to deliver enhanced services to their workforce and to their customers.

For more information on the Developer Program, please visit www.nortelnetworks.com/prd/dpp.





# Virtual Private Networks (VPN)

# Virtual Private Networks (VPN)

## Chapter Highlights

- The growth of the SOHO User – is growing to meet that changing demand. BCM now supports Client Side VPN with the ability to create 16 tunnels to SOHO users.
- Nortel Networks VPNs – are solutions that allow businesses to leverage the cost savings and ubiquity of public shared networks or the Internet for secure internal and external communications.
- Remote Access Outsourcing – allows mobile workers and telecommuters to access headquarters by way of a remote access server. Local access numbers and Internet access save money, as businesses can replace 800 numbers, modem banks and personnel costs.
- Extended Intranets – use dedicated local access lines to connect to Internet Service Providers (ISPs) and allow a business' branch offices and remote offices to access the headquarters' network via a public IP/FR/ATM network. Businesses save money because dedicated leased lines are charged by the mile.
- Extranets – are private communications links that are established between companies to enhance their ability to work together effectively through dynamic network connections.
- VPN Security – provides protection against interception of information through privacy, authenticity and integrity.
- Internet Protocol Security (IPSec) – provides privacy, integrity and authenticity for networking requirements for transmission of sensitive information over the Internet.

## VPN Sales Scenario

### The Challenge

National Trust is a large bank with headquarters in Toronto and eight branch offices in various locations across the country. Before implementing an effective communications solution, National Trust's phone costs for staff dialing into the network were high. In addition to domestic long-distance phone calls, the bank spent a significant amount of money on fax calls. Because remote access costs for ISDN and toll-free dial-in solutions were increasing rapidly as new workplace initiatives were established, the bank wanted to look for alternative solutions for providing connectivity into their network.

In order to meet its aggressive business goals, the bank knew it had to evaluate a wide range of options. National Trust was highly motivated to explore VPN technology options because of the rapidly developing requirements for supporting the growing types of Internet connectivity. The bank had also recognized the fact that more clients and executives were using small data networks at remote locations that would need access to the corporate network in the future. With a new VPN solution in place, the bank saw an opportunity to increase revenue while lowering its costs.

## The Nortel Networks Solution

Nortel Networks met National Trust's challenge with a VPN approach that geared initially towards establishing a remote access solution to support National Trust's mobile workers and telecommuters and secondly to provide Intranet and Extranet connectivity for clients and employees. The solution for National Trust was the Nortel Networks complete VPN solution, which involved a remote access application followed by the implementation of an extended intranet that was highly reliable and integrated easily with previously existing processes and procedures. In addition, National Trust installed Contivity switches to provide security as well as centralized management over the Internet.

Businesses generally implement remote access applications first, as they are the easiest to implement and understand. These applications allow mobile workers and telecommuters to access the headquarters by way of a remote access server. The cost savings of replacing 800 numbers, modem banks and associated personnel costs with Internet access can be significant. Using this type of VPN, remote workers can access company services from anywhere, as long as they have access to the Internet. No long-distance dialup connections or leased lines are needed.

Extended intranets typically use dedicated local access lines to connect to Internet Service Providers (ISPs). Businesses can realize significant cost savings as dedicated leased lines are charged by the mile. Two short local lines are less expensive than using one long leased line. VPN extended intranets allow a company's branch offices and remote offices to access the headquarters' network via a public IP/FR/ATM network.

Nortel Networks VPNs secure data through privacy, authenticity and integrity, guaranteeing companies protection against interception of information. Internet Protocol Security (IPSec), a standard for security at the network or packet-processing layer of network communication, offers security advantages, as it allows businesses to handle security arrangements without requiring changes to individual user computers.

## Implementation

VPN solutions from Nortel Networks can be implemented easily and seamlessly. Implementing Contivity devices at Meridian 1 sites allow for easier deployment of BCM. Since it is standards-based, Contivity interoperates with existing routing, authentication, directory and security services. This means Contivity can bridge the transition during the introduction of new IP services into the network. Contivity can be initially installed behind an existing IP access device (router, DSL modem, etc.) without disruption to the network. Or, a business deploying Contivity as a VPN gateway can later add firewall services and/or transition Contivity to the primary Internet access device for that site. By implementing a Contivity 1000 series switch, for example, businesses can realize cost savings while ensuring secure transmission of important information across the public Internet. With a Contivity 400 switch, an ideal device for connecting branch offices to multiple locations, businesses can make the transition seamlessly – requiring no changes to the network, workstations, applications, or operating systems.

## Business Impact

With the implementation of Nortel Networks VPN solutions, National Trust has successfully achieved a solid infrastructure that has and will continue to serve the business well. The bank has realized savings on equipment costs by using an Internet Service Provider (ISP), which has remote access equipment. In addition, National Trust no longer has to install and maintain expensive modem banks and it can avoid the cost of adding new equipment to the modem banks as the number of dial-in users increases. By using the Internet, National Trust has been able to avoid the cost of leased lines and has realized a significant reduction in its phone bills.

Security is an issue for all companies on the Internet, but particularly for National Trust because of the confidential information it handles. In the past, employees had to encrypt all information, including emails that were sent over the public infrastructure. Now, with the VPN, National Trust's authorized users can send email through its intranet, which is protected using PPTP protocol and firewall-to-firewall security. Through the use of digital signatures, National Trust can insure that no one is impersonating an authorized user and can use the signatures to detect data modification and ensure the integrity of the information.

## Overview

In simplest terms, a virtual private network (VPN) involves the use of 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, it is now possible to use public networks to cost-effectively broaden the reach of intranet applications.

The key feature of the shared network is that a given organization using VPN does not have exclusive use or control of it. Businesses can use VPNs to link corporate sites, replace dial-in lines and link to partners and suppliers. When a business uses a VPN to link the corporate network to those of partners and suppliers, it is referred to as an extranet.

***Figure 12-1***


VPNs are real and are being driven by end user requirements. With BCM, Nortel Networks is better positioned than ever to capture VPN market share. Offering a more affordable solution for smaller branch offices brings a whole new dimension to our VPN portfolio. By combining this low-cost branch solution with our existing industry-leading Contivity Extranet Switches, Nortel Networks is very well positioned against the competition in serving the enterprise with multiple locations.

## Emerging Trends

The consistent message from market analysts is that total VPN growth is exploding globally and the proliferation of IP has enabled the Internet to be the optimal alternative for VPNs. Organizations want simplified communications, with a single network securely carrying all traffic, with IP as the underlying protocol.

This growth means that the dominant trends for corporate access in the next few years will be VPNs and convergence using IP telephony

VPN market growth has been driven by the following applications:

- Connectivity with business partners
  - Just-in-time business
  - Shortened sales cycles
  - tighter customer relationships
- Connectivity for remote employees
  - More home-based and mobile workers
- Connectivity for different corporate sites
- Increased corporate mergers and acquisitions
- Unified interfaces for voice, data and video traffic.

Businesses are drawn to VPNs as a way to link corporate branch offices with each other and to support telecommuter access to corporate network resources. Another attraction VPNs hold for small and medium-sized businesses specifically is that they can support sales over the Internet that promote ecommerce and can expand their market reach. VPNs also provide the ability for the small and medium-sized business to link with partners and suppliers using extranets.

# Benefits

VPNs offer the following specific benefits compared to the traditional model of leased lines and 800 number dial-in lines:

- **Lower telephone bills** – Instead of spending money on leased lines, a company can implement a VPN and use the public Internet to avoid the cost of the leased lines. Some companies see as much as an 80% reduction in their phone bills. Total savings will depend upon the number of dial-in users and the length and type of the dedicated lines being replaced.
- **Lower equipment costs** – A business saves on equipment costs because ISPs have remote access equipment. Lower equipment costs are also a result of not having to install and maintain expensive modem banks, and businesses can avoid the cost of adding new equipment to the modem banks as the number of dial-in users increases.

- **Simplified network management** – Network management is simplified because the Internet service provider is now maintaining the wide area network and the remote access equipment that supports the VPN.
- **Better scalability** – Growing the number of sites connected is just a matter of adding more local access lines. Adding new remote users is as easy as adding software to the new PC and authenticating the new user to match the new name and password to the company database, and generally requires little or no change to the infrastructure.
- **Partnerships with partners, suppliers and customers** – Businesses can quickly and cost-effectively form partnerships with suppliers, contractors and business partners, promoting better business relationships.

# VPN Applications

There are three basic applications for VPNs:

- Remote access outsourcing
- Extended intranets
- Extranets.

## Remote Access Outsourcing

Remote access outsourcing is often the first application a company implements, as it is the easiest to understand. The illustration in Figure 12-2 shows Remote access outsourcing. Mobile workers and telecommuters access the headquarters by way of a remote access server. BCM supports remote access.

By replacing 800 numbers, modem banks and associated personnel costs with local access numbers and Internet access, businesses can realize cost savings of 50 to 70%. The more mobile and work-at-home employees a company has, the higher the cost savings.

Small ISPs can also use remote access outsourcing to outsource their remote access to a larger ISP. This is called Virtual Internet Service Provider (VISP).

**Remote Access Outsourcing Features and Benefits**

- Easiest to implement and understand
- Users work-at-home or are "road warriors"
- Small ISPs outsourcing RA infrastructure (VISPs).

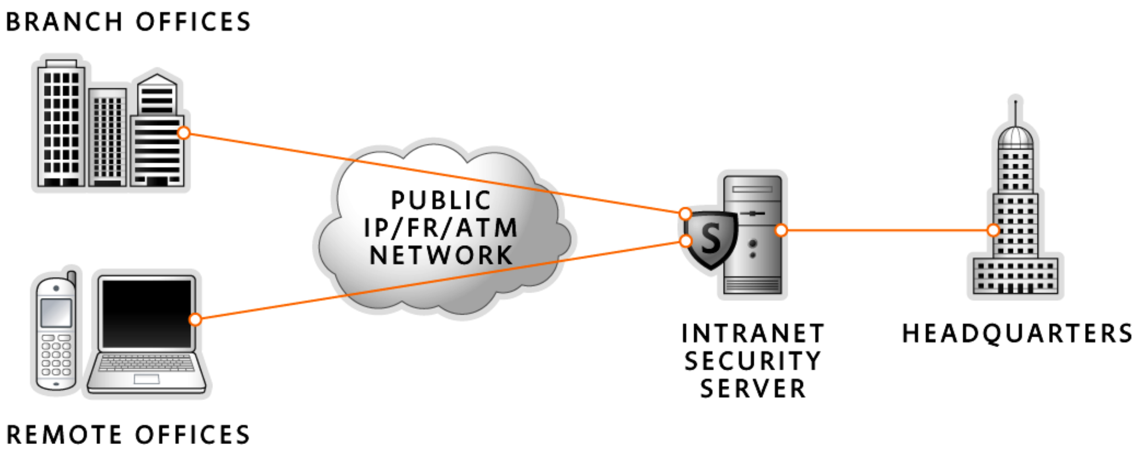
## Extended Intranets

Businesses can also use VPNs as an alternative to leased lines for linking corporate sites. These extended intranets generally use dedicated local access lines to connect to ISPs. The cost savings come from the fact that dedicated leased lines are charged by the mile, so two short local lines are less expensive than one long leased line.

Another advantage of using the Internet is that to fully mesh sites with dedicated lines requires considerably more lines than using the Internet to achieve the same level of connectivity.

Figure 12-2 illustrates a VPN extended intranet. These intranets allow a company's branch offices and remote offices to access the company headquarters' network by way of a public IP/FR/ATM network. An intranet security server restricts unauthorized access to the company headquarters' network. BCM with IPSec can support extended Internet access as shown below (Note: IPSec requires a keycode).

***Figure 11-2***



### Extended Intranet Features and Benefits

- Alternative WAN (site-to-site VPNs)
- More likely to use dedicated access
- Used to connect many sites over the Internet.

## Extranets

A third use of VPNs is to connect the corporate intranet to customers, suppliers, contractors and business partners by way of a public IP/FR/ATM network. This network connection is called an extranet. Extranets are private communications links established between companies to enhance their ability to work together effectively through dynamic network connections.

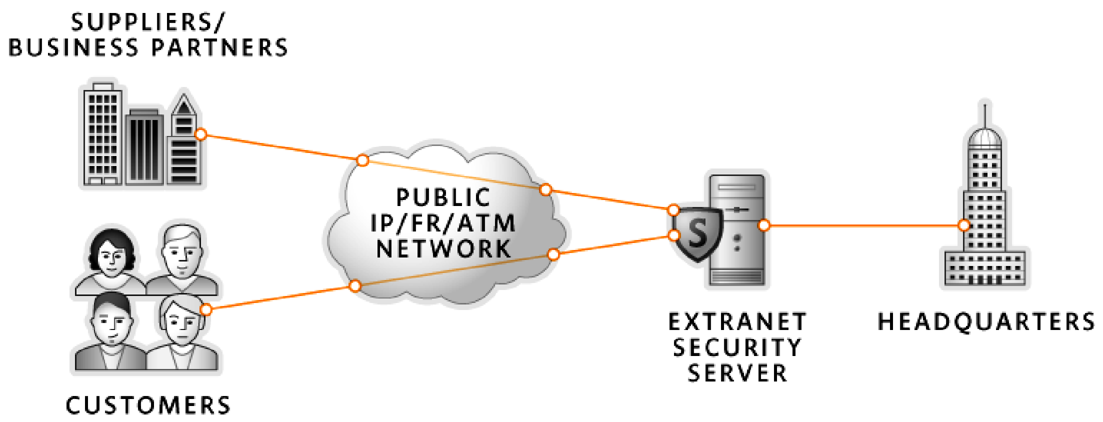
VPNs constitute the ideal infrastructure for creating extranets, making them practical and affordable. With the right VPN management technology, companies can define VPNs that cross business boundaries, providing a secure method of exchanging data and other resources with trusted partners, suppliers and other key business associates.

The advantage of extranets is that they leverage the Internet to replace the time- consuming and costly practice of installing leased lines to other companies' networks.

Extranets require careful attention in the areas of interoperability and security. It is essential that a business choose a solution for the extranet that will communicate with the solution chosen by its business partners. Standards are critical to the success of extranets.

Security is an important part of an extranet implementation and the information made available to these business partners needs to be carefully controlled. An extranet security server restricts access to portions of the headquarters' network not intended for their use. Like other VPN implementations, extranets can use either dialup or dedicated access lines. The illustration in Figure 12-3 shows an example of an extranet. BCM can supply the small to medium-sized business with extranet capability.

*Figure 12-3*




## Extranet Features and Benefits

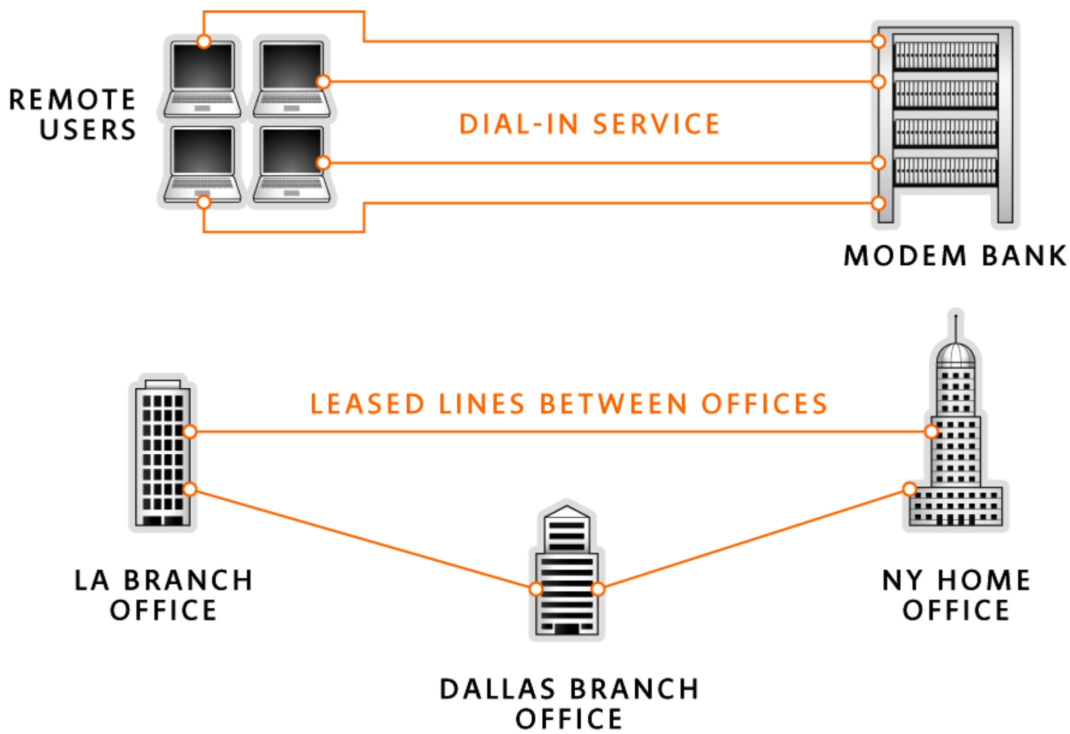
- Dynamic connection to business partners
- Security and interoperability need to be developed
- Can be dialup or dedicated.

## Traditional Networking Infrastructure

In the non VPN traditional networking model, shown in Figure 12-4, companies purchased dedicated lines to link corporate sites. For remote or mobile employees, companies purchased 800 numbers and large modem banks.

This solution often made it impossible to justify connecting small remote offices to the corporate intranet and connecting to suppliers, contractors and business partners was simply too time consuming and expensive to even consider. As the number of remote and mobile employees increased, it became obvious that this solution did not scale well.

***Figure 12-4***

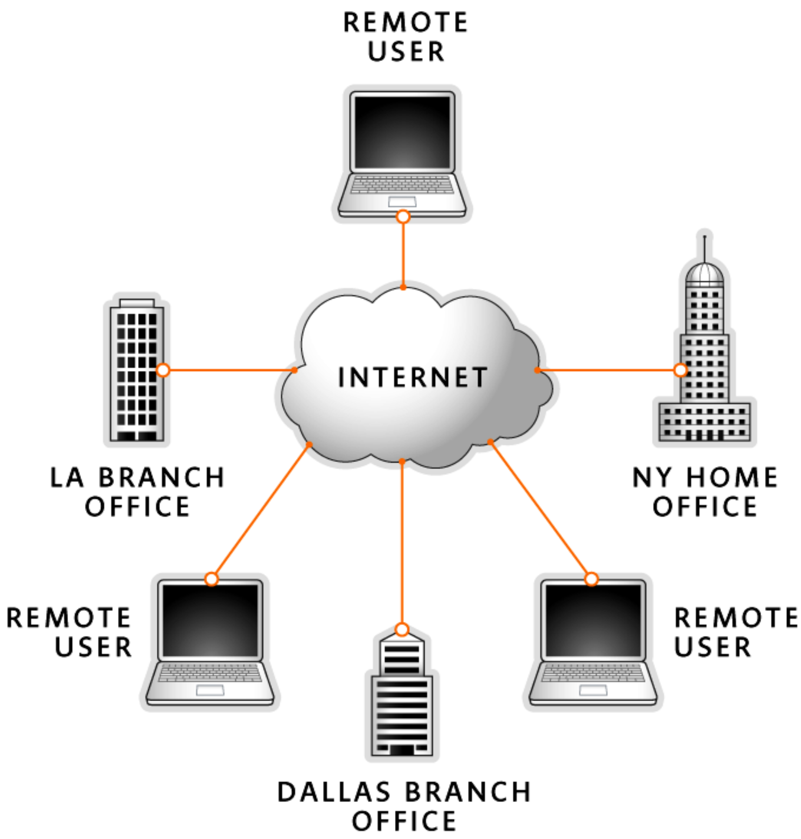


# VPN Infrastructure

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 12-5, 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 12-5***

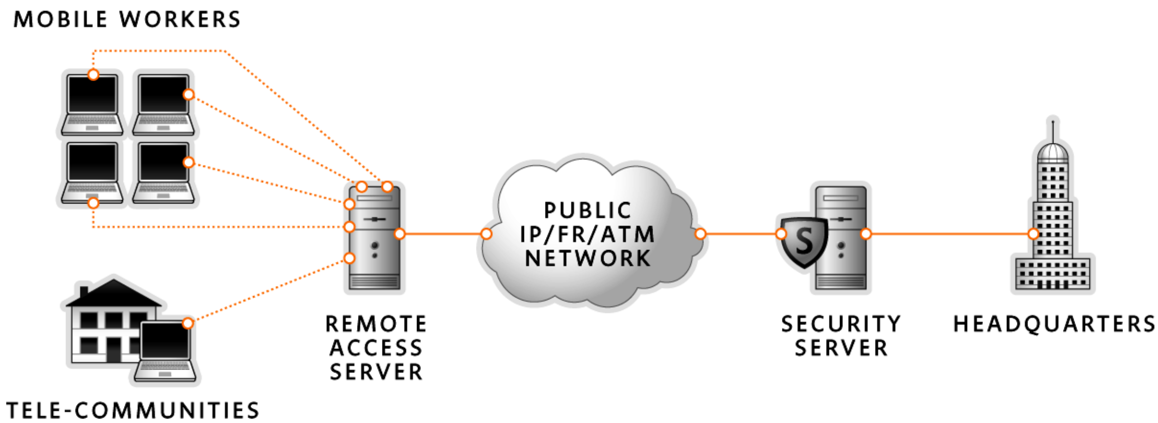


# VPN Service Models

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.

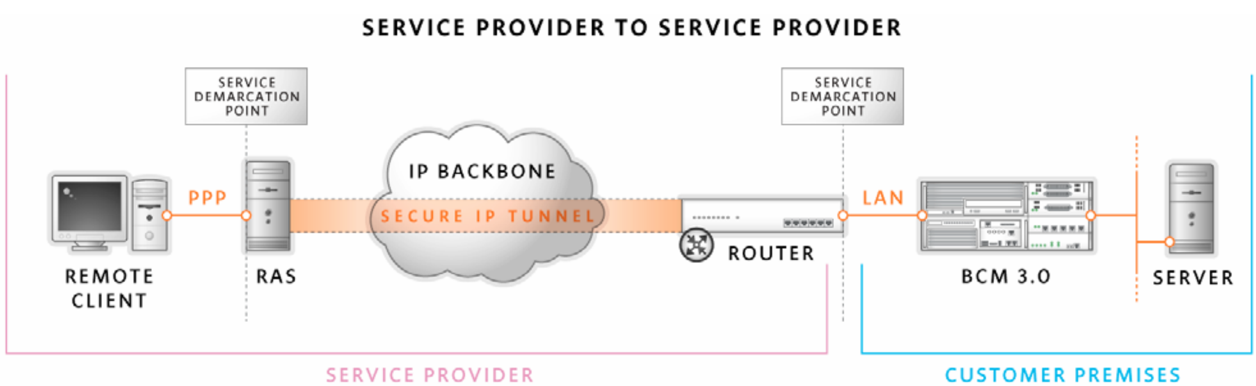
***Figure 12-6***




## Service Provider to Service Provider Model

Figure 12-7 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 12-7***

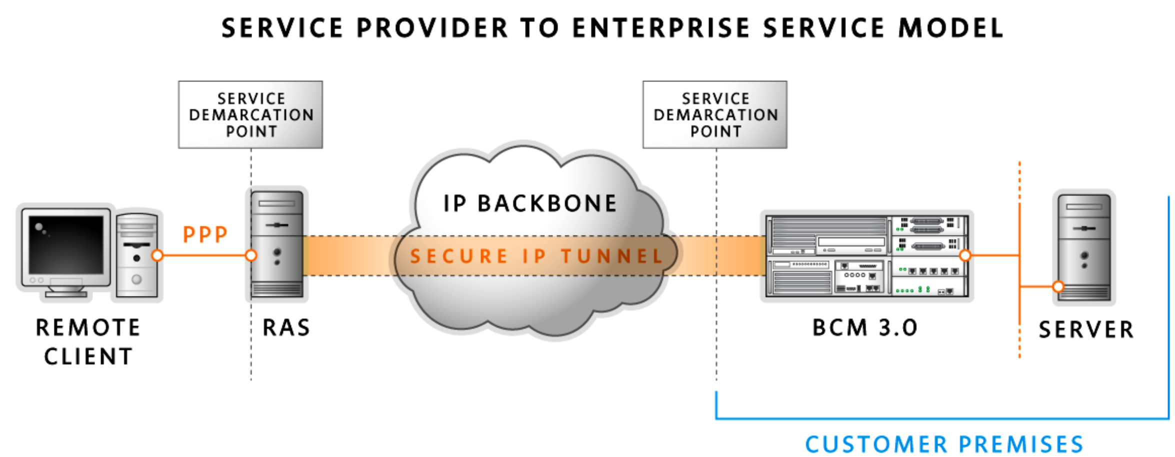


This model is often used to support the remote user functionality of VPNs. The remote user dials in to a service provider's Remote Access Server (RAS) and requests a tunnel to the corporate network. The RAS sets up a tunnel to a router owned by the service provider that connects to the corporate network. 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 12-8, 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 12-8***

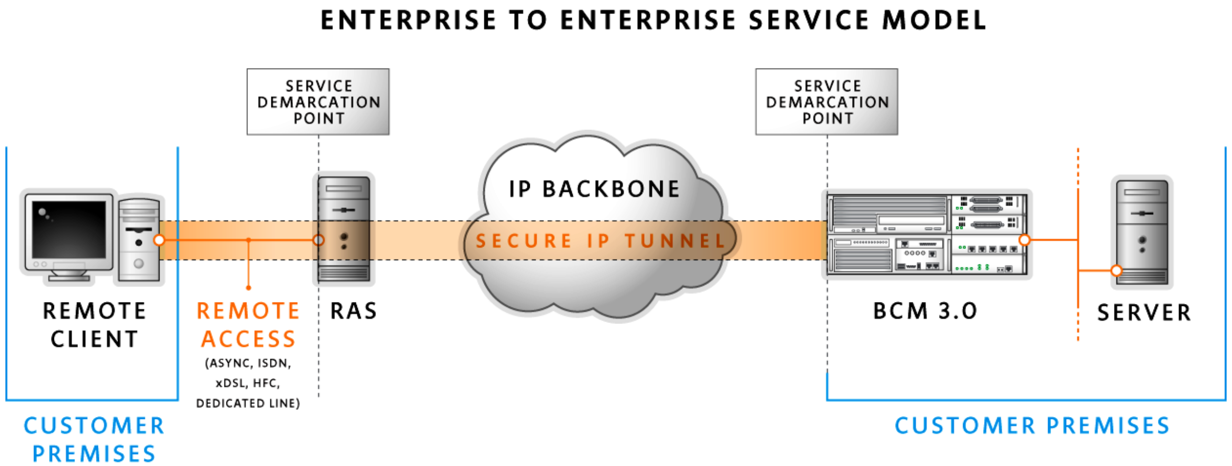



## Enterprise to Enterprise Service Model

In the enterprise to enterprise model shown in Figure 12-9, 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.

***Figure 12-9***




# 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.

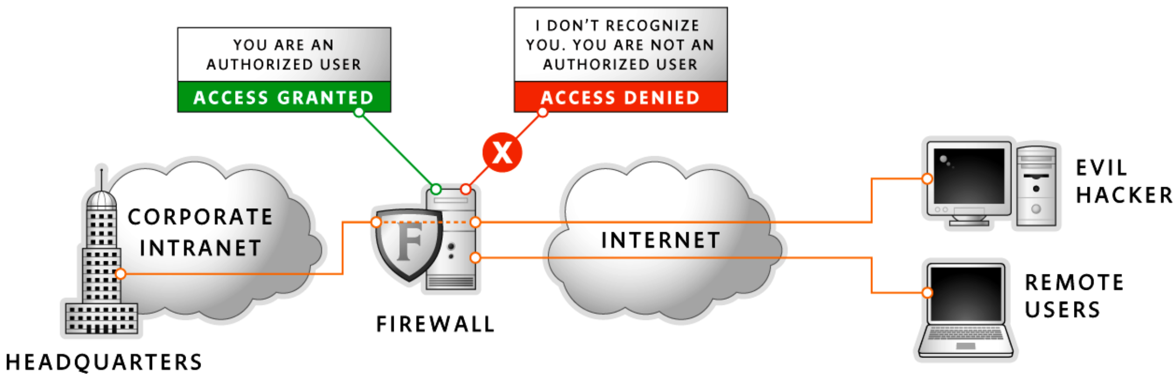
## Privacy

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 12-10***




# 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

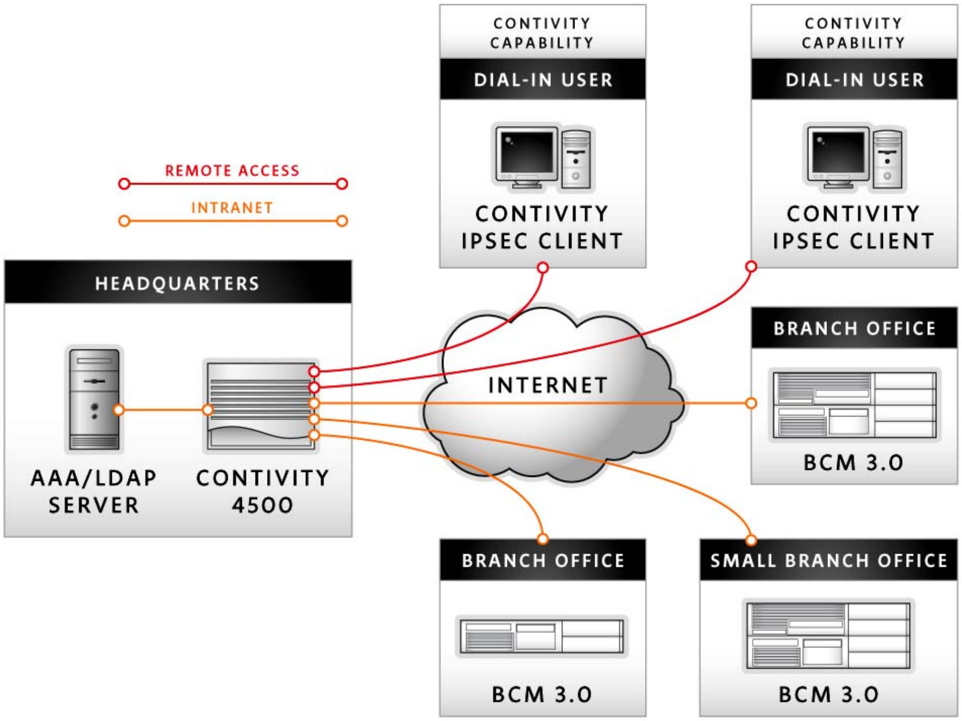
Small, medium-sized and large businesses use VPNs for typical applications. Each of these businesses, depending on its size and connection requirements, has a unique need.

## 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 12-11***

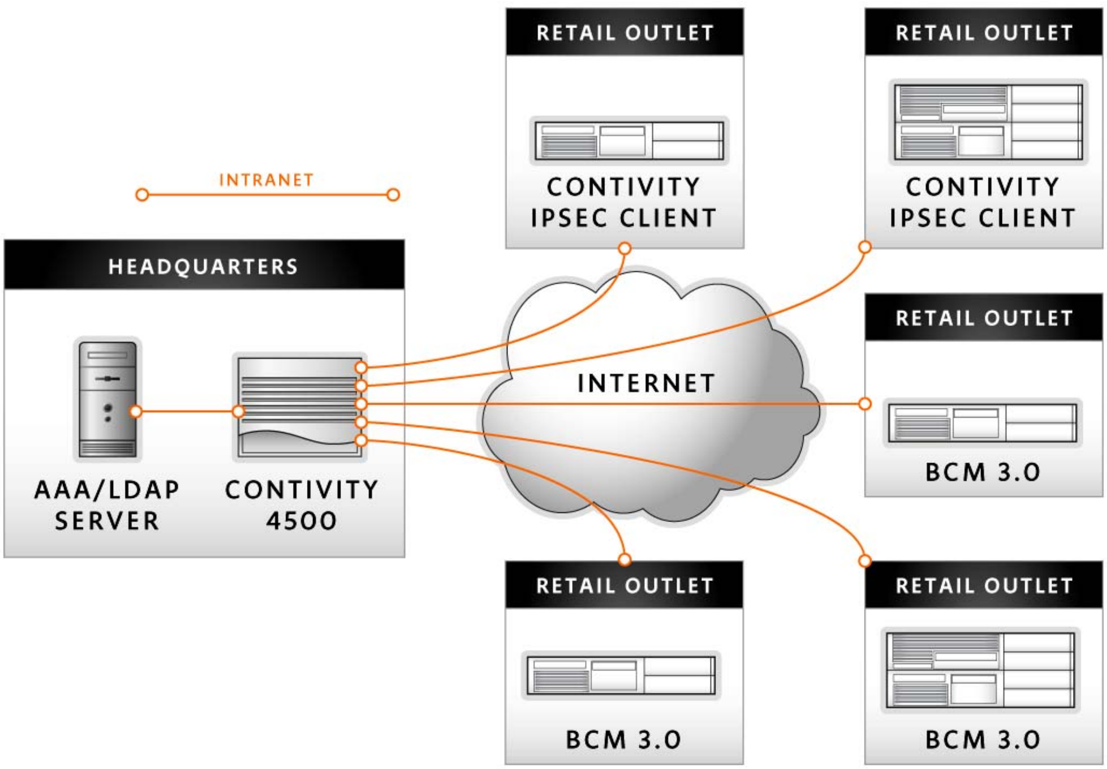


## Small to Medium-Sized Business

The typical small to medium-sized business needs an inexpensive way of connecting many branches.

Figure 12-12 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 the client support capabilities of BCM that are delivered as part of BCM 3.0.

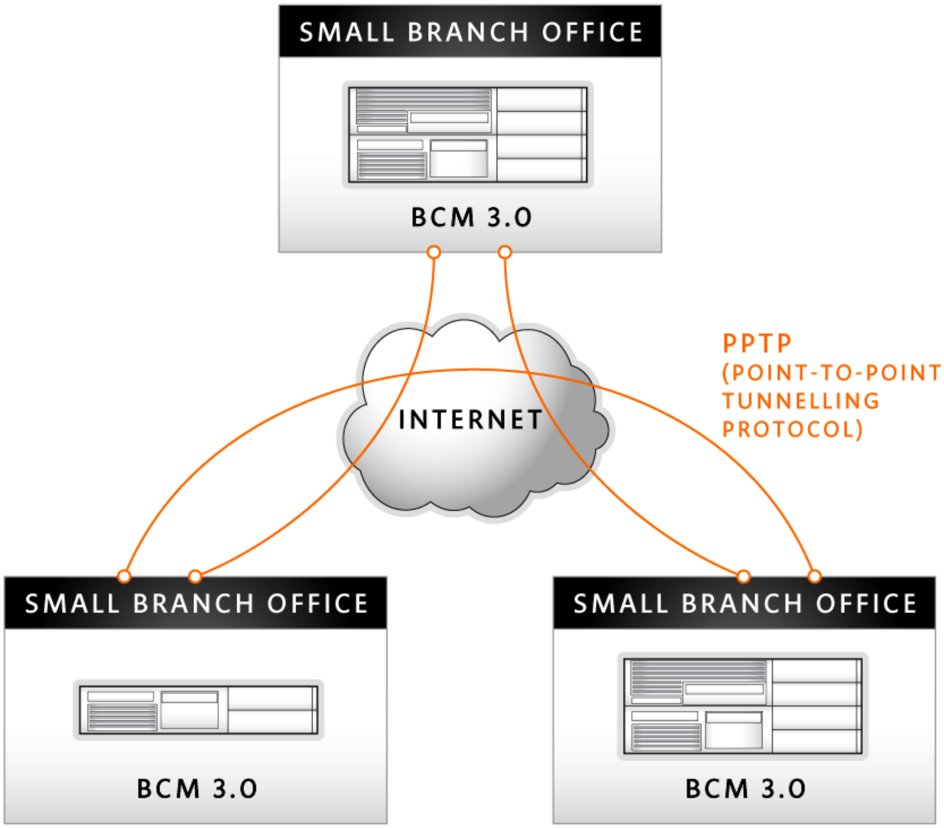
***Figure 12-12***



## Extranets

Figure 12-13 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.

***Figure 12-13***



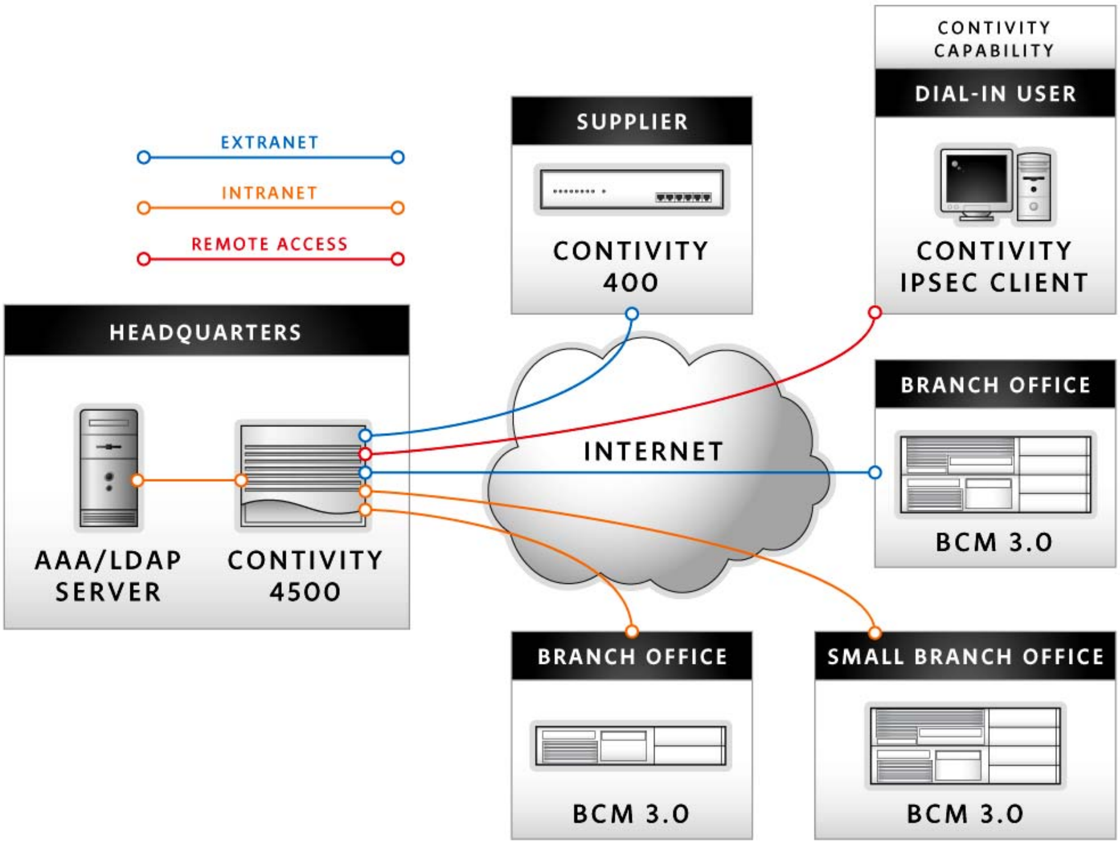

# Nortel Networks Complete VPN Solution

Figure 11-14 shows the entire range of possibilities with a complete VPN solution from Nortel Networks. 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 12-14***




# 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 12-1 identifies the supported encryption and authentication levels.

***Table 12-1***

| Method (strongest to weakest) | Encryption of IP Packet Payload | Authentication of IP Packet Payload | Authentication of Entire IP Packet |
|---|---|---|---|
| ESP Triple DES SHA1 | Yes | Yes | No |
| ESP Triple DES MD5 | Yes | Yes | No |
| ESP 56-bit DES SHA1 | Yes | Yes | No |
| ESP 56-bit DES MD5 | Yes | Yes | No |
| ESP 40-bit DES SHA1 | Yes | Yes | No |
| ESP 40-bit DES MD5 | Yes | Yes | No |
| AH HMAC SHA1 | No | No | Yes |
| AH HMAC MD5 | No | No | Yes |

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 BCM allows up to 16 secure tunnels to be established between BCM and Contivity and/or BCM to BCM.

## IPSec Client Support

BCM 3.0 supports IPSec Client Termination in addition to branch office mode (server-to-server) connections supported in BCM Release 2.5. 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.

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 3.0 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 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 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 IPX tunneling
- 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 BCM.

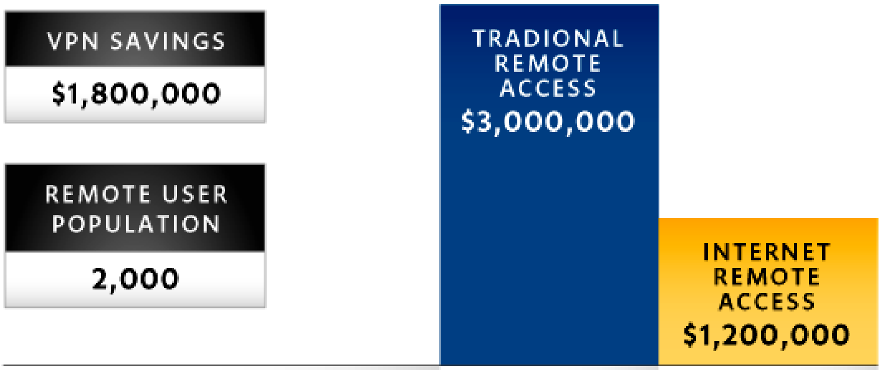
**Note**: While BCM supports both IPSec and PPTP, they may not be used at the same time.

## VPN Value Proposition

VPN solutions allow companies to leverage the cost savings and ubiquity of the Internet for secure internal and external communications.

This is just one of many reports (shown below) available today reflecting the savings that can be realized by using the Internet as an alternative to a dedicated network.

***Figure 12-15***




| | | |
|---|---|---|
| T1 LINES | $48,000 | $68,400 |
| HARDWARE | $208,000 | $44,800 |
| PHONE, ISP CHARGES | $2,100,000 | $1,080,000 |
| EQUIP. MGMT & USER SUPPORT | $600,000 | $0 |

SOURCE: FORRESTER RESEARCH

This report deals with a remote access network, since that is the most prevalent VPN application most companies are using today. It also implies that there could be cost savings in scenarios where fixed sites are linked together or networking from LAN to LAN, site to site.

Compare the costs of a traditional remote access application for two thousand remote users to a VPN or Internet-based remote access model and the savings are $1.8 million dollars a year, or $900 per year per user. Other numbers may be slightly higher or lower, but the key point is that there is significant per-user cost savings achieved by leveraging the Internet.

Businesses will look for VPN cost savings first, but the second consideration is the flexibility that VPN offers the company. Businesses can augment their existing network with applications that they can place on the Internet. An important example would be if an organization is opening up a new business or office in another country and needs to get it connected into the network. A dedicated line is just too expensive and takes too long to implement. VPN remote access provides the flexibility to accomplish this quickly and inexpensively.

Businesses can also take advantage of VPN to quickly link their suppliers, partners and even their customers. This is a perfect example of the value of an extranet. With the Internet and the VPN technology, companies can now create an extranet quickly and with less cost than in the past.





# System Management and Software Options

# System Management and Software Options

## Chapter Highlights

- Unified Manager – the single point for element management of all aspects of individual BCMs.
- CallPilot Manager – a Web-based application used to administer the voicemail and call center applications in BCM systems.
- Backup and Restore Utility (BRU) – supports the backup and restore of BCM system programming.
- Network Configuration Manager (NCM) – a multisite management feature that provides centralized configuration and system management capabilities for a number of BCMs in a network.
- Desktop Applications – a variety of desktop applications and tools such as Call Center clients, Voice Mail, Interactive voice response (IVR) that lead to increased employee productivity and customer satisfaction.

## System Management and Software Options Sales Scenario

### The Challenge

Headquartered in New York, Union Capital Bank has over 70 branches nationwide and is one of the largest financial institutions in the United States. After experiencing steady growth and two major acquisitions in the last six years, Union Capital's patchwork system was causing frequent problems and bringing down the company's bottom line.

The disparate systems at each of the bank's branches prevented centralized remote management and Union Capital was incurring high costs from contracting third party network administrators to manage and maintain the network at each branch. Ad hoc systems of outdated equipment meant that some branches could not offer even essential features, like voicemail. Union Capital needed a unified network to provide standardized service to all branches and to eliminate third party costs.

## The Nortel Networks Solution

Working with Nortel Networks, Union Capital developed a solution that balanced centralized control and management with autonomy at the branch level. Using a Nortel Networks Meridian 1 PBX at the head office and BCMs at each of the branch locations, Union Capital created a consistent platform for hardware and software solutions that enhanced the efficiency of management and corporate operations.

BCM Unified Manager provides a simplified interface for users to access all programmable parameters at both the installation and administration levels. Drop-down menus provide access to dialog boxes where users can enter, modify and delete data as well as access performance charts and tables. The BCM hardware hosts helpful information, providing product documentation and tips. The setup and management wizards are user-friendly, allowing installers to program the BCM quickly and easily.

The BCM user-managed functions allow users to program the features and autodial keys on their telephone and program and change their voicemailbox functions. With the Backup and Restore Utility (BRU), system administrators can readily back up and restore all the configuration and user data.

BCM Network Configuration Manager (NCM) 2.0 is software that allows network managers to configure and manage BCM systems. It enables system managers to maintain and manage any number of BCMs on the network. This software complements Unified Manager by providing a single view of the network with networkwide capability. The multi-site management capabilities with BCM Release 3.0 provide the ability to apply programming changes to all BCM systems within a network from a centralized location.

## Implementation

Network Configuration Manager offers standardized configuration and installation. It enables system administrators to archive multiple configuration templates, dramatically reducing the time and effort required to bring a new BCM unit onto the network. Configurations for an existing device can be downloaded to the Network Configuration Manager Server, edited and resaved and then downloaded onto a new device. This approach eliminates the need for senior management personnel to go on site during an installation, thereby maximizing the efficiency of a business' system administrators. The template-based approach ensures that device configuration across the network is tightly controlled, simplifying maintenance tasks and reducing overall management expenditures.

### Business Impact

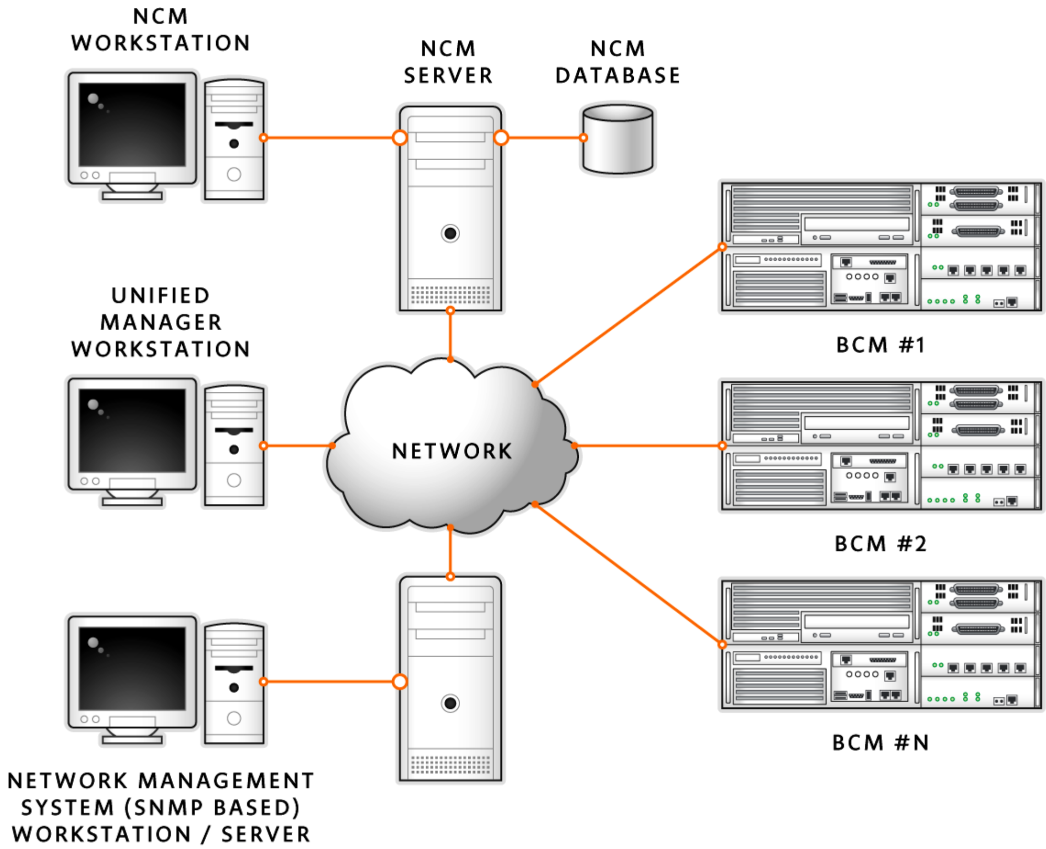
Streamlined installation and deployment has brought a new branch online every week since implementation. In addition, with a survivable and redundant system, even if connections to the head office are lost, each branch can still operate independently. The solution's centralized management capabilities and ease of programming have eliminated Union Capital's need for third party managers and have increased the company's operational efficiency. With the Unified Manager application, the unit can be easily managed from any Web-enabled workstation.

In addition to central administration and maintenance of its network, Union Capital has benefited from streamlining software deployment and download to all users across the network. Implementing a BCM solution has helped to future-proof the bank's network. Union Capital Bank wants to offer PC banking for their customers in the near future. Now, its network will facilitate all of the necessary applications so that the company does not have to rely on the data center to move forward.

## Overview

The BCM management framework comprises two fundamental components: Unified Manager and Network Configuration Manager. 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. Via Unified Manager, any BCM in the network can be configured and managed from any PC on the network. The Network Configuration Manager (NCM) is an optional multisite management tool providing centralized inventory database and configuration capabilities for networks comprising many BCMs. Figure 13-1 depicts a BCM network together with the Unified Manager and NCM.

*Figure13-1*




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 tend to simplify the learning curve for new system administrators and minimize training required for system configuration. A standard Web browser interface also provides 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, a business can use any PC on the network to administer the entire system, or multiple systems, from a single location. System administration of the BCM using a digital telephone set is not supported.

BCM 3.0 offers significant enhancements in the area of system management and software options, including:

- Network Configuration Manager (NCM) 2.0 and support for multisite management
- Unified Manager improvements and additions.

## Emerging Trends

Converged networks, like those that can be built with Nortel Networks BCM, offer enterprise customers great efficiencies in the area of technical support, leading directly to savings in personnel costs. Since a converged product like the BCM encompasses all of the functions traditionally provided by a separate PBX, a router, a voicemail server, multiplexer and remote access server, it is an ideal platform for businesses that require a complete communications solution with simplified management. The use of IP telephones greatly reduces the effort previously associated with user moves and changes, a further benefit provided by a converged network solution.

At the same time, businesses are increasingly taking advantage of systems that provide centralized configuration and management capabilities to enable multiple systems to be efficiently administered from one location. Having all communications functions provided in one footprint reduces the cost of the total solution and reduces the complexity of managing a network with separate devices for each function. Remote administration from a PC desktop is another marketplace demand, and businesses are choosing systems based on this capability.

When all communications capabilities are provided by a single platform, system availability becomes a critical factor. Requirements for proactive notification of potential service impacting issues to a centralized monitoring facility becomes critical in order to address any issues that may arise in a timely manner that will minimize system downtime.

## Benefits

With the advanced system management and software options offered by BCM Release 3.0, businesses can realize specific benefits of simplified administration, flexibility and cost savings.

With BCM Release 3.0, Nortel Networks has focused on enhancing the product reliability and ease of operability. Each BCM platform is delivered with preloaded, software-based applications that are enabled via keycode, either locally or remotely, providing installation and service flexibility.

Via Unified Manager, the single Web-based management tool, BCM can be configured and managed from any location via an IP connection. This integrated browser-based OA&M tool provides a simple, intuitive method of managing the system from any Web-enabled workstation.

The optional Network Configuration Manager capability reduces the total cost of ownership for enterprises with multiple BCMs within a network. Using the NCM client/server in a headquarters location and with NCM enabled on the BCMs in the network, businesses can save significant time and money in supporting mass programming changes.

Additionally, wizard templates are provided to reduce initial installation and maintenance times. These templates allow the technician to run multiple tasks associated with a particular function simultaneously instead of performing routine tasks one at a time.

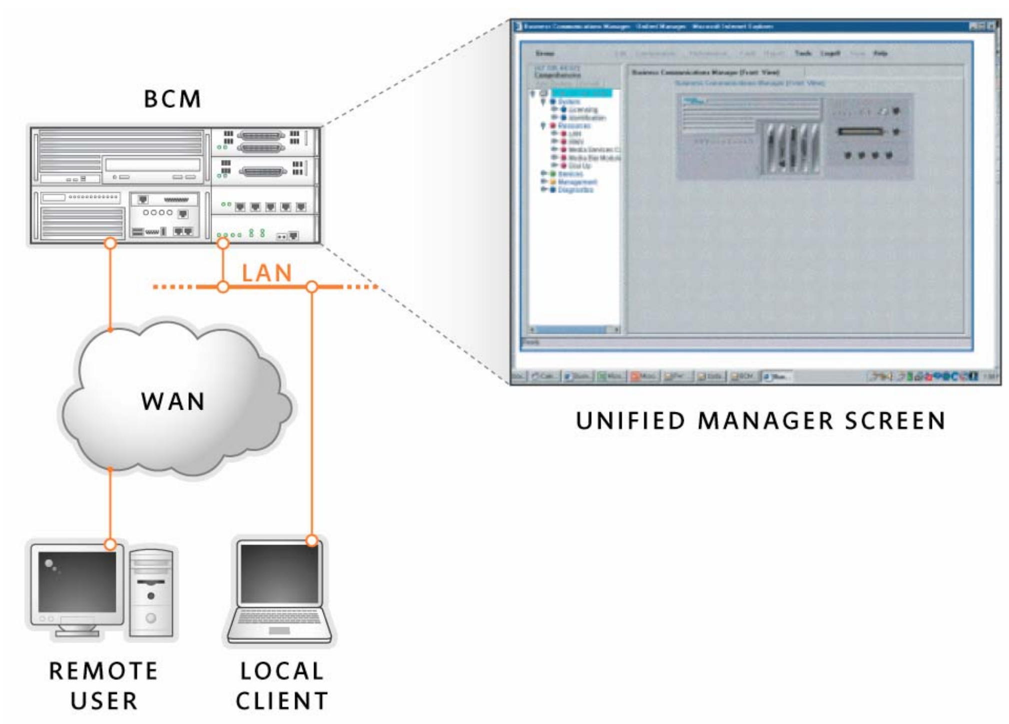
BCM 3.0 also provides a dedicated maintenance area that streamlines administration screens, reducing the number of screens a technician must view for system diagnostics and troubleshooting. This single maintenance area reduces errors and contributes to gaining overall installation and serviceability efficiency.

Alarms and events for specific BCMs can be viewed via the Unified Manager Maintenance area or they can be configured to generate SNMP traps via the SNMP system on the BCM. This ensures that personnel at a Network Operations Center can receive timely notice of issues via third party SNMP watcher tools or management frameworks.

# Unified Manager

The Unified Manager, shown in Figure 13-2, is the single point for managing all programming for individual BCM systems. With Release 3.0, Nortel Networks has made several enhancements to this browser-based management tool. Access to the Unified Manager is password protected, making it secure for both enterprise customers and small to medium-sized businesses.

***Figure 13-2***




Technicians can use Unified Manager to quickly set up users, mailboxes and directory numbers (DNs). With such speed and efficiency, a business can be up and running in no time. Changes can also be handled remotely through the Unified Manager.

## Unified Manager Capabilities

BCM Unified Manager has the following capabilities:

- Provides the ability to perform BCM remote administration and configuration via the IP network
- Provides the ability to view and change configuration settings for all applications on a BCM

- Provides the ability to monitor events and alarms for troubleshooting purposes
- Enables 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 administrator's existing NT login, which means he or she is not required to maintain a separate ID just for system administration
- Provides controlled, multilevel access to users, enabling system administrators to provide user customized access and control to Unified Manager.

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 (NMS). The Optivity NMS Discovery and Launch enables visibility of BCM elements and supports the launch of the BCM Unified Manager from within Optivity.

## Unified Manager Enhancements

BCM 3.0 provides a number of improvements and additions in Unified Manager, including:

- **Login page improvements –** In previous releases of BCM, each wizard required a separate login. With BCM Release 3.0, the Setup and Management Wizards login page is restructured and users are able to log in to a wizard session once and then simply click on wizard icons to launch new wizards.

- **Network Update wizard** – A Network Update wizard has been added to simplify the task of changing network settings after the initial installation of the BCM.

- **Network-loaded templates** – The Add Users wizard allows a system administrator to initialize a group of DNs on a BCM with settings from a predefined DN record template. These DN record templates are stored locally on each BCM system and must be edited (with the Edit DN Record Template wizard) on the BCM before the template can be used. The network-loaded templates feature removes the restriction of having to re-enter the DN record template configurations on each new BCM. It allows an enterprise to create and manage a single template repository and to use those templates from Add Users wizards running on remote BCMs. One of the BCM systems can be designated as “server” for this DN record template repository as the templates can only be edited on a BCM system. The template-editing wizards can be run remotely for editing templates in the repository over the network. (Any HTTP server on the network could be used as the repository, although editing of templates must be performed on a BCM system.)

- **Button programming in Add Users wizard** – The Add Users wizard allows a system administrator to apply DN record settings, including telephone set button programming, to a group of one or more DNs all at once. Previously, limitations in the handling of programming for buttons on DNs that do not have sets attached prevented the inclusion of button programming in the DN record information. Those limitations were removed in FP1 and button programming is now included in the DN record settings that the Add Users wizard applies.

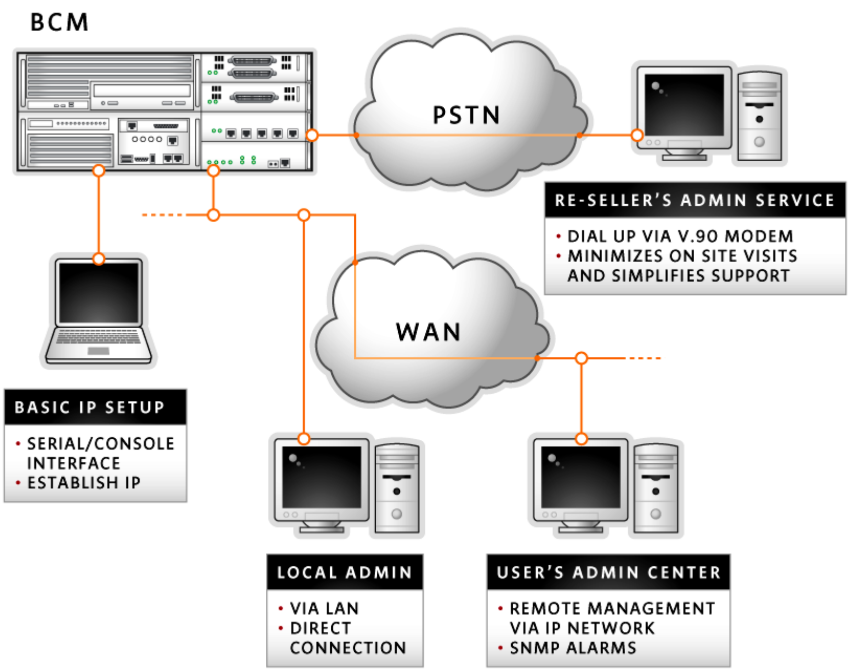
- **Multiple levels of user access and permissions –** Administrator access to the Unified Manager has been redesigned to provide much greater flexibility and security in defining administrator access privileges. Users can be assigned access as a member of a user group, with the access privileges specified for that group. Up to four user groups can be defined, with privileges ranging from read-only of restricted parts to full read and write capability. In addition, user group privileges can be set that enable or restrict access to specific areas of system administration and control.

# BCM Management Access Options

Connection to Unified Manager is 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 13-3***



## BCM Unified Manager Interface

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.

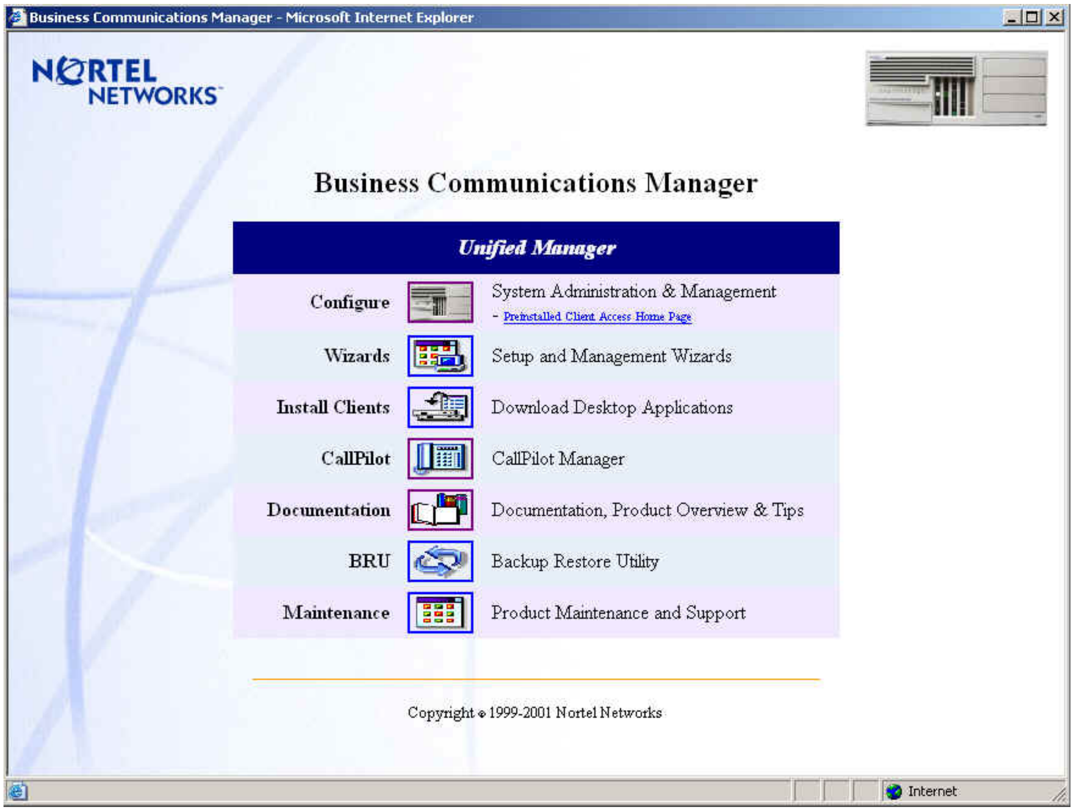
The administrator can also develop templates to use on multiple system configurations. This means the multisite administrator can create one configuration for multiple branches and simply download the configuration at other sites.

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.

Unified Manager can also be integrated into Optivity Network Management System (NMS) via the Optivity Integration Toolkit. This enables BCM discovery, launch and alarm integration on Optivity NMS.

The Unified Manager main page is shown in Figure 13-4. 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 voicemail and call center services. Documentation is also provided, as is access to product maintenance and support.

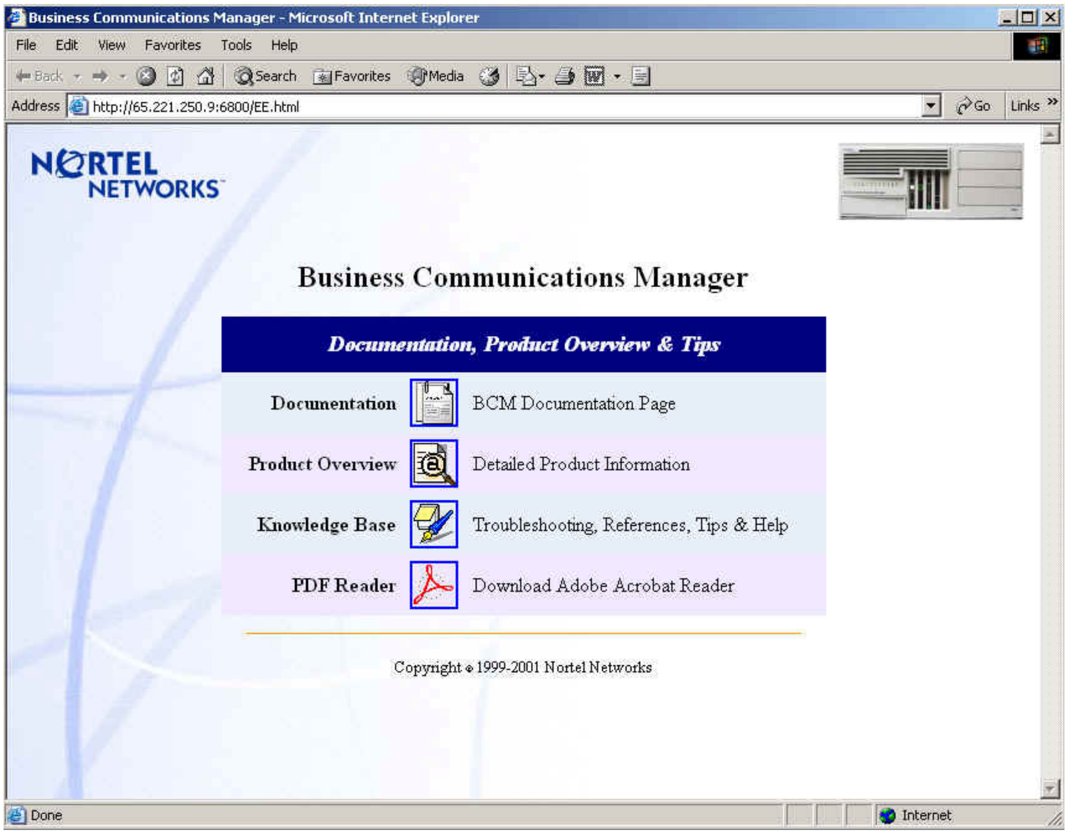
*Figure 13-4*




Significant improvements to Unified Manager in BCM Release 3.0 speed up the installation process and put as much information as possible at the system administrator's fingertips.

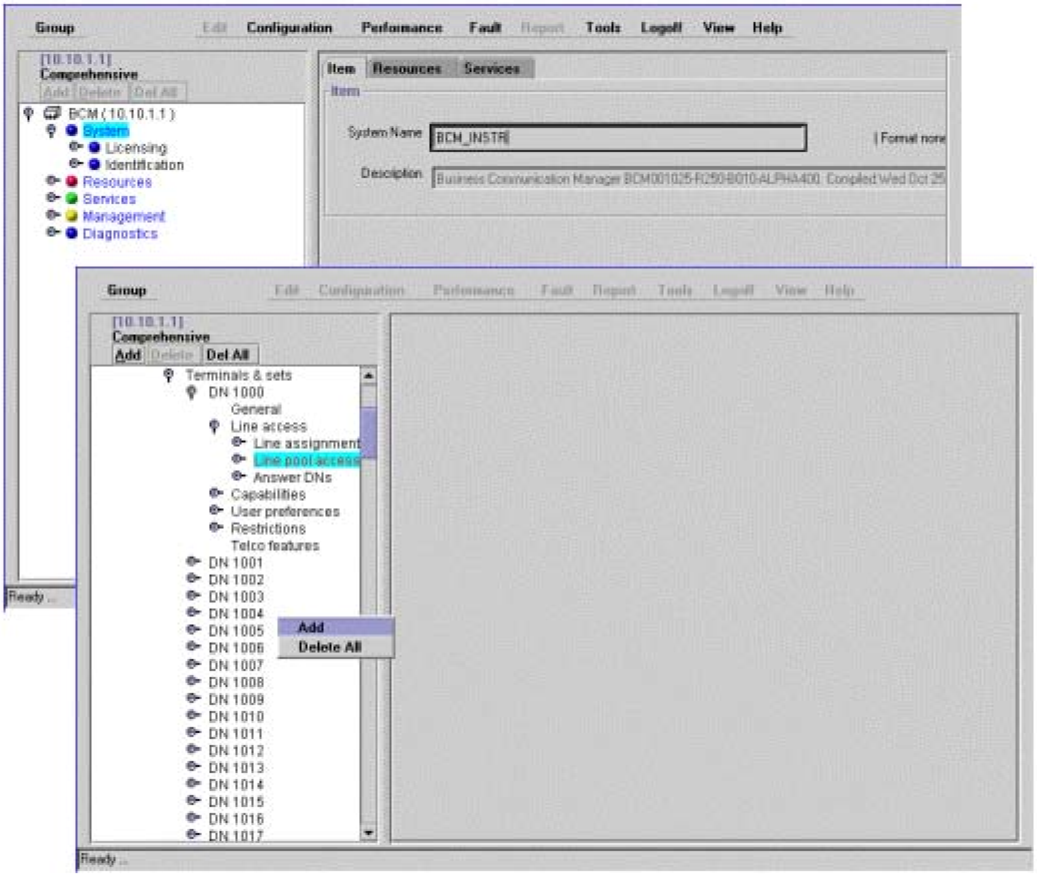
As shown in Figure 13-5, the product documentation suite and a product overview are provided directly on the hard disk of each BCM system.

*Figure 13-5*




The “Configure” selection of the Unified Manager main page (Figure 13-6) provides access to the BCM programming interface. This is organized into five groupings entitled System, Resources, Services, Management and Diagnostics. Figure 13-6 also shows examples of screen shots from the Configure portion of the Unified Manager.

*Figure 13-6*

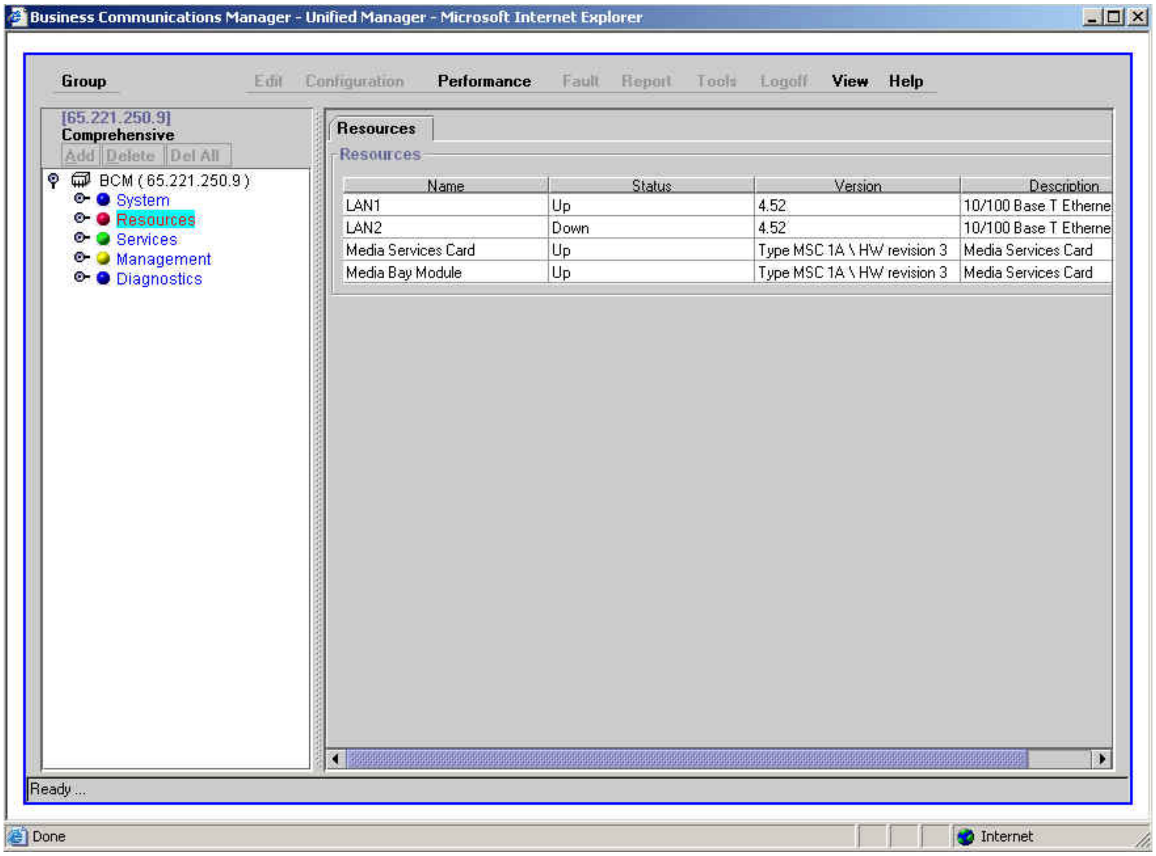



The Services and Resources selections enable the configuration of telephony and data services and resources, as described in the following sections.

## Resources Menu

The illustration in Figure 13-7 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.

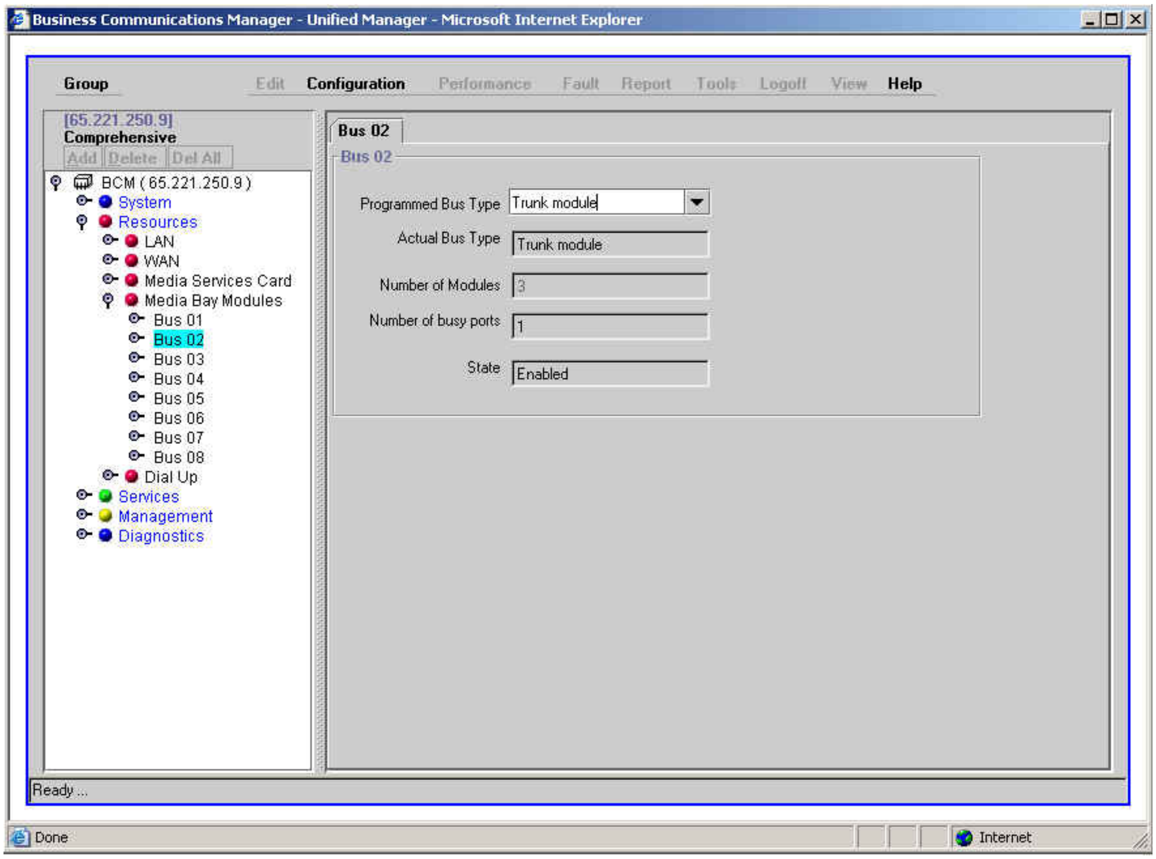
***Figure 13-7***




When the Resources command button is expanded, users can access all of the configuration menu options to enable and configure BCM resources.

The screen shot in Figure 13-8 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.

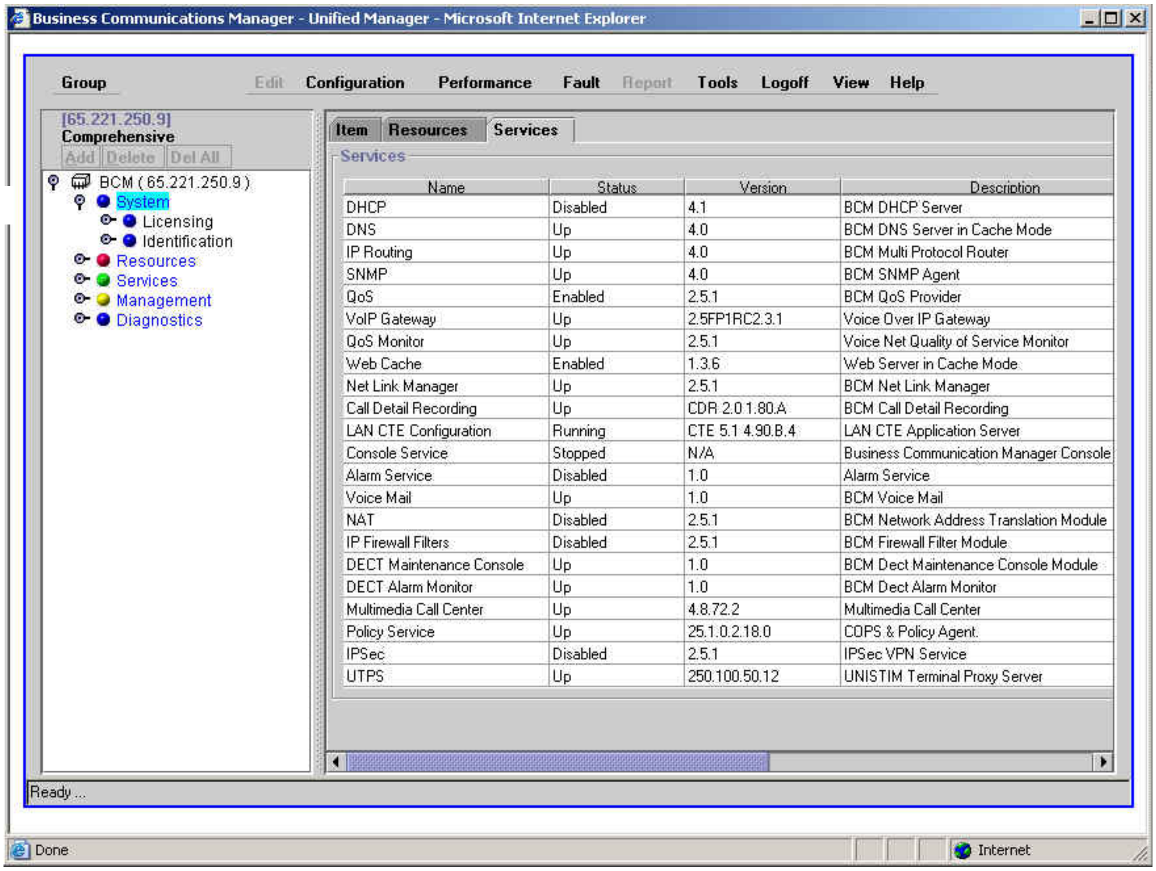
*Figure 13-8*


Business Communications Manager - Unified Manager - Microsoft Internet Explorer
Group
Edit
Configuration
Performance
Fault
Report
Tools
Logoff
View
Help
[65.221.250.9]
Comprehensive
Add
Delete
Del All
BCM ( 65.221.250.9 )
System
Resources
LAN
WAN
Media Services Card
Media Bay Modules
Bus 01
Bus 02
Bus 03
Bus 04
Bus 05
Bus 06
Bus 07
Bus 08
Dial Up
Services
Management
Diagnostics
Bus 02
Bus 02
Programmed Bus Type
Trunk module
Actual Bus Type
Trunk module
Number of Modules
3
Number of busy ports
1
State
Enabled
Ready ...
Done
Internet

## Services Menu

The screen shot in Figure 13-9 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 13-9***




## 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 13-10 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 13-10***

## CallPilot Manager OA&M Interface

CallPilot Manager is a Web-based management tool that can be accessed from the Unified Manager main page. It is used to administer the Voice Mail and Call Center applications in BCM systems. (CallPilot Manager replaces the separate application called Voice Mail Manager used to administer previous releases of BCM.)

The look and feel of CallPilot Manager is compatible with the CallPilot Web OA&M, the Web-based administration tool for the Meridian 1 messaging environment.

The CallPilot Manager application does not replace 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.

Since users can now access CallPilot Manager using a Web browser via Unified Manager, it is no longer necessary for administrators to load a CallPilot specific management application onto their PC. A business can use any PC that has access to the network for Voice Mail and Call Center administration.

Management of CallPilot 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.

## 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 successfully 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.

In BCM 3.0, the voice application backup section includes all the voicemail 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. In NCM 2.0, BCM 3.0 backups do not back up the voicemail messages that may be in the system when the backup is taken. Only the user recorded greetings are backed up or restored.

## System Monitoring and Performance Statistics

There are several methods for monitoring systems and viewing BCM performance data. The Unified Manager enables alarm information for a specific BCM to be viewed. In addition, some system performance information is visible via the maintenance section of the Unified Manager.

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).

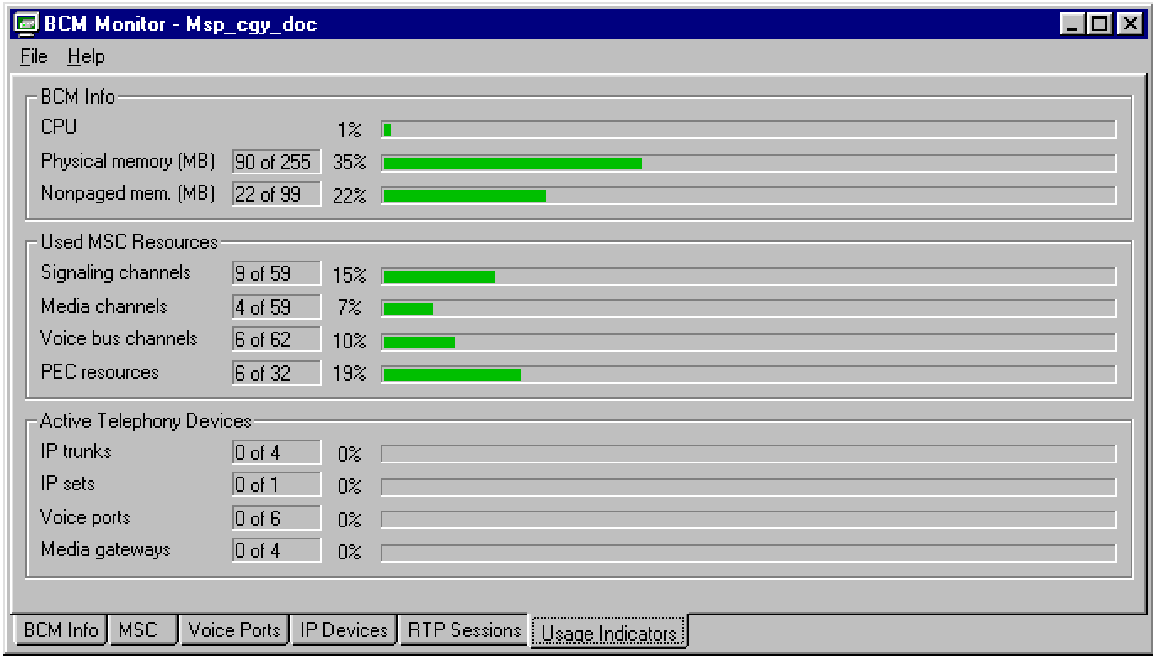
If a performance display is active, it is automatically updated with real-time performance information in user-selectable time increments.

The focus of the real-time monitoring capabilities is:

- Overall system status
- IP telephony functions of the BCM system
- Utilization of resources on the Media Services Card (e.g. signaling channel usage)
- Operation of telephony applications (e.g., Voice Mail, Call Center, etc.).

Figure 13-11 shows an example of the System Monitoring information that is available.

***Figure 13-11***




Multiple instances of the BCM Monitor application can be used on a single PC to monitor several remote BCM systems at the same time.

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.

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.

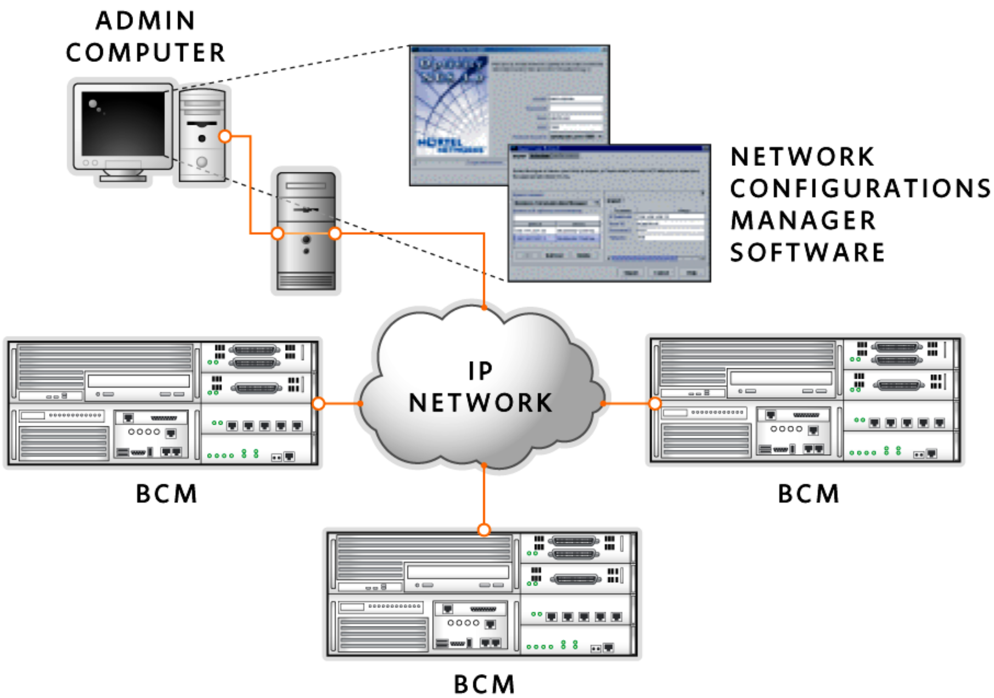
Finally, BCM also supports MIB II. This SNMP MIB enables SNMP capable management systems to poll the BCM for status and performance information, particularly in the areas of WAN and LAN interfaces and throughput/performance of the BCM data router.

## 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 13-12, 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 single 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.

*Figure 13-12*




## Emerging Trends and Benefits of NCM

- **Network Management** – Businesses are increasingly spending more money to support voice and data networks as business critical processes move online. Converged networks, like those created with BCM, offer solid opportunities for businesses to improve technical support efficiencies through simplified networks and centralized/remote management. Analysts estimate that a converged network can offer up to a 50% reduction in the staff a business requires to manage and administer the network. Assuming a 1% overhead of personnel required to manage a network, a 200-employee company could save approximately $100, 000 per year with a converged network.

- **Deployment –** Most customers understand the value of standards and consistency in branch applications. Yet, even when those standards are well documented and understood by an engineer, there are always differences in interpretation. Two engineers can set up and configure a system correctly, but do so in very different ways.

BCM Network Configuration Manager allows businesses to create one or several standard configuration templates for the BCM. Users can save these templates and download them onto new BCMs. Additional site-specific programming, such as a unique name and IP address, are entered and the final site-specific configuration is saved back onto the NCM server. In this manner, the new BCMs are installed with a standard configuration.

This ease of programming saves significant time and therefore, money because it takes just a few minutes, instead of the 8-16 hours that it takes to program a BCM from scratch. At a $50 per hour labor rate, that saves $400 to $800 per BCM.

- **Inventory** – Keeping an accurate inventory allows a business to quickly expand offices and add employees to keep up with business demands. However, an accurate inventory requires a business to keep meticulous records, which are still subject to errors. Inevitably, there are undocumented additions to any system. System administrators could use Unified Manager to query each system one by one, go through the fields and manually record all of the data. But, typically, they visit each site and perform a physical inventory. For a business that has sites all over the country, this process requires expensive air travel to visit each site. Or, a business has to pay a partner to perform the inventory for it.

- **Moves/Adds/Changes –** Ongoing moves, adds and changes can be expensive in a multi-node network. As a result, changes which impact a large number of nodes in a common manner are either not made or require a large amount of time/money to be used in effecting the change. NCM not only reduces the time to make these bulk configuration changes but also enables BCM systems to be configured dynamically to drive business objectives. For example, NCM can facilitate a cost-effective process for deploying weekly auto-attendant greeting updates to all BCMs in a network, ensuring that consistent and timely greetings are used to further business objectives.

## 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 customer-supplied Oracle database is available.

## NCM Configuration Tasks

Network Configuration Manager with BCM 2.5 FP1 or BCM 3.0 systems allows the network administrator to perform the following configuration 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
- 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 TFPT server
- Generate/view/print inventory and the following configuration reports:

    - System QuickStart Wizard Settings
    - DN list
    - System WAN Interface Settings
    - System List
- Launch Unified Manager.

BCM Network Management Configuration Manager can produce a complete hardware inventory and print out of 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.

With NCM 2.0 and BCM 3.0, programming changes can now be applied to all, or a subset of, BCM systems within a network from a centralized location. The changes that can be applied include:

- Applying software option keycodes
- Bulk device access password change
- Change Call Center greetings
- Change Auto Attendant greeting tables
- Change Custom Call Routing settings
- Change VoIP remote gateway tables
- Change call routes and destination codes
- Program data router configuration parameters and routing tables
- Add call filters/restrictions
- Manage SNMP trap generation settings.

NCM 2.0 provides the ability to schedule the listed changes and to define groups of BCM systems to which the changes may be applied.

NCM has been integrated with IVR to provide:

- 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 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 13-1***

| Standard Applications | |
|---|---|
| Telephony | √ |
| Auto Attendant | √ |
| Call Detail Recording | √ |
| Hunt Groups | √ |
| BRI S/T | √ |
| PRI | √ |
| Integrated QoS Routing Plus | √ |
| WAN Routing | √ |

Table 13-2

| Bundles | Capability |
|---|---|
| Bundle #1: Silver – 16 | Nortel Networks Call Center, Basic<br>CallPilot VPIM & AMIS Message Networking<br>BCM LAN CTE, 16 seats<br>BCM Attendant Console, 1 seat<br>CallPilot Unified Messaging, 16 seats<br>CallPilot FAX Messaging |
| Bundle #2: Silver – 32 | Nortel Networks Call Center, Basic<br>CallPilot VPIM & AMIS Message Networking<br>BCM LAN CTE, 32 seats<br>CallPilot Attendant Console, 1 seat<br>CallPilot Unified Messaging, 32 seats<br>CallPilot FAX Messaging |
| Bundle #3: Silver – 64 | Nortel Networks Call Center, Basic<br>CallPilot VPIM & AMIS Message Networking<br>BCM LAN CTE, 64 seats<br>BCM Attendant Console, 1 seat<br>CallPilot Unified Messaging, 64 seats<br>CallPilot FAX Messaging |
| Bundle #4: Gold<br>(Note that customer must have purchased a Silver bundle before upgrading to the Gold bundle.) | BCM MCDN & Q.SIG Voice Networking<br>Nortel Networks Call Center to Professional Call Center Upgrade<br>Nortel Networks Call Center Reporting<br>CallPilot FAX Overflow<br>CallPilot FAX On Demand<br>BCM IP Sec |

**Note**: Voice Messaging seats are not included in Silver or Gold bundles.

***Table 13-3***

| Options | Capability |
|---|---|
| Try-and-Buy Options<br>(Options available for 60 days from activation or until replacement.) | BCM CallPilot VPIM & AMIS Message Networking<br>BCM MCDN & Q.SIG Voice Networking<br>BCM DPNSS Voice Networking<br>BCM LAN CTE – Maximum seats<br>BCM Attendant Console – one seat<br>BCM VoIP Gateway – two trunks<br>BCM CallPilot Unified Messaging – Maximum seats<br>BCM CallPilot FAX Suite<br>BCM IP Telephony Client –eight seats<br>BCM CallPilot Voice Messaging – Maximum seats |
| BCM CallPilot Message Networking | Enables VPIM & AMIS Message Networking |
| BCM Call Center | Enables Nortel Networks Call Center |
| BCM Call Center Reporting | Enables Nortel Networks Call Center Reporting |
| BCM Professional Call Center | Enables Nortel Networks Professional Call Center includes Call Center Reporting |
| BCM Call Center to Professional Call Center upgrade | Upgrades from Call Center to Professional Call Center options available with or without Call Center Reporting |
| 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 Networks 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). |
| BCM IVR Host Communications Emulation | Activates host communications emulation on BCM to support access to PeriView management tool using a dumb terminal. |
| BCM TSP Seats 1.x and 2.x only | Adds 1, 4, 8, 16, 32, 64 Maximum TSP Seat License(s) |
| BCM LAN CTE 2.5 only | Adds 1, 4, 8, 16, 32, 64 Maximum Seat License(s) |
| BCM Attendant Console | Adds one Attendant Console License (max. five) |
| BCM VoIP Gateway | Adds 2, 4, 8, 16, 32 VoIP Gateway trunks (60 max). May require additional PEC IIIs |
| BCM CallPilot Unified Messaging | Adds 1, 4, 8, 16, 32, 64 Maximum Seat License(s) |
| BCM CallPilot Voice Messaging Mailboxes | Adds 1, 4, 8, 16, 32, 64 Maximum Mailboxes |

| Options | Capability |
| --- | --- |
| BCM CallPilot FAX Messaging | Enables CallPilot FAX Messaging |
| BCM CallPilot FAX Overflow | Enables CallPilot FAX Overflow |
| BCM CallPilot FAX On Demand | Enables CallPilot FAX On Demand |
| BCM CallPilot FAX Suite | Enables all FAX features |
| BCM Companion Wireless Portables | Enables 1, 4, 8 BCM Companion handsets |
| BCM Companion Base Station (two radios) (USA only) | Enables BCM Companion Base Station |
| BCM IP Sec | Enables BCM IP Sec VPN capability |
| BCM IP Telephony Client(s) | Adds 1, 4, 8, 16, 32, 64 IP Clients |
| BCM Point to Point over Ethernet (PPoE) | Enables PPPoE on a per site basis |
| BCM 3.0 Software Upgrade Authorization | Allows installation of BCM 3.0 software upgrade on Release 2.5 and Feature Pack 1 systems. |
| BCM Network Configuration Manager | Network Configuration Manager Software Authorization Code, enables multisite configuration management of BCM via centralized Network Configuration Manager software |
| SoftPhone Client CD | i2050 SoftPhone Client, enabling the i2050 on a per client basis (required if your total clients exceed 16 per site) |

# 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 13-4***

| Product | Server/Client Software |
|---|---|
| BCM Network Configuration Manager | Network Configuration Manager 2.0 (250 systems), server/client software and documentation to support centralized configuration management for up to 250 systems |
| BCM Network Configuration Manager | Network Configuration Manager 2.0 (1000 systems), server/client software and documentation to support centralized configuration management for up to 1000 systems |
| BCM Network Configuration Manager | Network Configuration Manager 2.0 (2000 systems), server/client software and documentation to support centralized configuration management for up to 2000 systems |
| BCM Network Configuration Manager | Network Configuration Manager 2.0 Upgrade (from 1000 to 2000 systems) |
| Interactive Voice Response (IVR) PeriProducer IVR Application Development Tool | IVR software and documentation for development of IVR applications for BCM |
| Interactive Voice Response (IVR) PeriStudio IVR Audio Development Tool | IVR software and documentation for development of IVR audio prompts for BCM |





# BCM 3.5 Updates

# BCM 3.5 Updates

This document supplements and overrides the BCM 3.0 Handbook. It identifies the enhancements and additions contained in BCM Release 3.5. As most of the information and instructions have not changed with Release 3.5, the entire handbook contents have not been updated.

The sections in this document mirror the chapters in the BCM 3.0 Handbook, so you can easily identify new features and enhancements in each area.

## What's New?

BCM 3.5 introduces over 50 new software features and 6 new hardware components. Enhancements and new features have been developed in the areas of core telephony, IP telephony, applications, data, management and serviceability targeted to the small and medium business and enterprise branch. These new features improve productivity, customer service and the overall simplicity of the BCM solution.

### BCM 3.5 Key Content

| Voice Packet/Circuit | Data | Networking | Applications | Management | Security | Platform |
|---|---|---|---|---|---|---|
| BST CAP (T7316E & T24)<br>255 Speed Dials<br>Autohold IC Page<br>Answer/DSS Key<br>Passive Call Log<br>Passive CLID<br>Doorphone<br>I2050 Audio Kit<br>MCDN Tandem E.164<br>Alpha Tagging<br>GATM<br>VoIP Codec Renegotiation | Dual v.35 WAN Card<br>Universal T1 DTM<br>PPP Fragmentation & FR Fragmentation | Centralized AA/VM<br>H.323 Interoperability with Succession 1000, M1 IPT and Radvision ECS<br>SIP Trunking & Data Service<br>Norstar IP Interoperability | IP Music on Hold<br>AA Holiday Schedule<br>AA Holiday Schedule<br>AA Park-n-Page<br>200 CP Unified Messaging Subscribers<br>CC Expected Wait Times<br>CC Agent Help<br>LAN CTE Enhancements<br>IVR Park-n-Page | NCM Path Distribution<br>NCM FTP Support<br>NCM Off Database Storage<br>Enhanced NCM Reporting<br>NCM VM Backup and Restore<br>NCM 60 day Try-n-Buy<br>BCM 60 Day Try-n-Buy<br>SNMP Trap Dial-out<br>Secure Dial Back<br>SNMP Trap Maintenance<br>BCM Line Monitor<br>Unified Manager Help | SSL/SSH<br>OS SP6<br>Password Policy<br>Denial of Service<br>BRU Data Encryption<br>Apache Upgrade | 3.0 Upgrade<br>3.0.1 Updgrade<br>APC UPS Integration<br>Brazil Localization<br>Channel Capacity increase from 16 to 32 |

BCM 3.5 strengthens its position in the enterprise branch and multisite space, adding leading features, enhanced multisite management, security and call center features.

# Introduction

BCM 3.5 is a key component of the Succession Enterprise strategy, a portfolio of solutions that deliver choice in voice service deployment with rich multimedia applications to enhance enterprise communications and drive more powerful customer interaction, resulting in higher potential revenue and lower costs by removing barriers of distance and location. Embodying all the attributes that made this solution a Best in Test winner, while improving reliability, flexibility and manageability, BCM 3.5:

- Builds on the award-winning capabilities of BCM 3.0
- Increases application performance
- Extends enterprise communications with seamless Succession call server networking
- Enhances management capabilities.

While BCM 3.5 incorporates a number of serviceability enhancements that make it easier to install, support and service, the service and support offerings for BCM 3.5 do not significantly depart from those of BCM 3.0. BCM 3.5 is primarily a software upgrade. Conveniently, this software loads directly onto BCM 3.0 or 3.0.1. For BCMs released prior to release 3.0, additional upgrade steps are necessary.

BCM incorporates sophisticated voice, data and management capabilities, so support staff must be well versed in all three technologies.

## BCM 3.5 Value Proposition

In an increasingly competitive marketplace, companies of all sizes need to find a strategic edge. BCM 3.5 helps businesses compete more effectively by delivering new technologies designed to increase employee efficiency, maximize the viability of existing resources and reduce operating costs.

## BCM 3.5 Key Benefits

BCM 3.5 offers businesses the following key benefits:

- **Cost reduction** – by consolidating voice and data over a single T-1 and centralizing management capabilities and voice mail

- **Productivity –** by saving time and improving efficiency with business-building features and applications like multimedia call centers and interactive voice response
- **Scalability –** by providing a solution that can meet the voice and data communications needs of sites with 20-200 users on a single, cost-effective platform
- **Security** – by providing integrated firewall and security measures like Network Address Translation (NAT), Basic and Stateful packet filtering, Point-to-Point Tunnelling Protocol (PPTP), VPN support with IPSec and Secure Socket Layer (SSL)

BCM 3.5 can help businesses gain a strategic edge over their competitors. By supporting both digital and IP telephony in a single, easy-to-manage, cost effective unit, this scalable platform enables organizations to adopt IP-based solutions at their own pace. And, it lets them preserve existing investments in Norstar telephony equipment.

# Telephony

As with previous releases, the telephony components of the BCM perform call processing and connect public switched telephone network (PSTN) lines to BCM telephones. Customers don't need to purchase an additional application server, or voicemail servers to handle the traffic.

A select number of telephony enhancements in BCM 3.5 target Nortel Networks global market. These include Alpha Tagging, a Global Analog Trunk Module and Brazil Localization.

## New Features and Enhancements

- Enhanced Business Series Terminals
  - T7316E
  - T24 Key Indicator Module
  - BST Doorphone
- Global market enhancements
  - Alpha Tagging
  - Global Analog Trunk Module
  - Brazil Localization
- Automatic hold during incoming page
- CLID alignment on multiple line appearances
- System speed dial increase
- Answer DN/DSS Shared Key
- Passive Call Log

### Business Series Terminals

Business Series Terminals from Nortel Networks offer one of the highest reliability ratings in the industry. Each device is equipped with an integrated LCD window that guides users through the phone's features while they are using it – so they can leverage robust functionality without spending time in training classes. Business Series Terminals boast rich features and enhanced capabilities that provide telephony solutions for a broad landscape of users—ranging from the needs of users with minimal feature requirements to the more demanding needs of high-volume users and business executives.

Cross platform compatibility with Business Communication Manager and Norstar* provides investment protection and offers a cost-effective migration path.

### T7316E (Enhanced) Telephone

Nortel Networks Business Series Terminals T7316E is a full-featured, expandable, multi-line telephone that has a two-line, 16-character-per-line 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. It is designed for high call volume positions requiring access to extensive system features. Typical users include supervisors, managers, executives and other business professionals.

**Key value propositions:**

- **Modular design** – allows the T7316E to be deployed anywhere as a feature-rich standalone solution; or, by adding the T24 Key Indicator Module, as a Central Answering Position for efficient call routing
- **Integrated Busy Lamp Field/Direct Station Select** – increases customer satisfaction while improving operator efficiency by utilizing multi-segment icons for accurate and efficient routing of customer inquiries
- **Audio control center** – simplifies telephone operation by clustering common audio features for fast and precise toggle action between handset, headset, and speaker
- **Built-in speaker** – improves communication by providing hands free audio capability for multi-tasking and group participation
- **Menu driven display with soft keys** – provides users with an intuitive interface for fast and accurate access to system features

**Product 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)

**Weights and dimensions:**

- T7316E: 1.06 Kg
- T7316E: 1.06 Kg , (L) 20.6cm x (W) 26.2cm x (H) 8.8cm

***BST Central Answering Position (T7316E+T24)***

Nortel Networks Business Series Terminals 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 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.

**Key value propositions:**

- **System powered** – reliability is improved and installation simplified by removing the auxiliary power supply and utilizing the BCM or Norstar system to power the BST CAP
- **Busy Lamp Field/Direct Station Select** – increases customer satisfaction while improving operator efficiency by utilizing multi-segment icons for accurate and efficient routing of customer inquiries
- **Multiple appearances of target line** – improves communication flow by providing the user with multi-segment icons for multiple target lines
- **Multiple hunt groups** – ensures real-time contact by providing a series of telephone lines organized in such a way that if the first line is busy the next line is hunted and so on until a free line is found
- **Multi-segment icons** – improves internal communication flow by providing a station status for system subscribers.

**T24 Product features:**

- BCM and Norstar compatible
- Multi-line w/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

**Weights and dimensions:**

- T7316E: 1.06 Kg , (L) 20.6cm x (W) 26.2cm x (H) 8.8cm
- T24 KIM: 30 Kg, (L) 19.6cm x (W) 9cm x (H) 8.5cm

The following table highlights configuration rules:

| | | Enhanced KIM (use KIM for multiple appearances of target lines) | | | Ordinary KIM (use KIM for BLF/DSS) | | |
|---|---|---|---|---|---|---|---|
| System Type | Max # sets per system | Max # sets | Max # EKIM on each set | EKIM (max #) | Max # sets | Max # OKIM on each set | Max OKIM |
| BCM *Rel 3.5* | 96 | 12 | 4 | 48 | 84 | 9 | 756 |

***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.

### Alpha Tagging

This feature displays the programmed system speed dial name for an incoming call, if the CLID number matches. In markets where name is received from the CO, that name takes precedence.

### Global Analog Trunk Module

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.

### Brazil Localization

Brazil Localization provides support for a Brazil profile, local ISDN protocol and support for Portuguese language on voicemail prompts and end user documentation.

### Automatic Hold during Incoming Page

A per set administration option has been added with BCM 3.5 to allow each set to automatically and immediately hold its active call when an incoming page is being presented to that set. In addition to holding an active call when a page comes in, a set will return the auto held call to the original active state once the interrupting page terminates. A short announcement "blip" sounds prior to the page. If the user is on a headset or handset, they will hear the page via the speaker at the same time as their conversation. If the user is on hands-free, the page will override the existing conversation. By pressing the held key, the user can terminate the page and return the call to its active state.

### CLID Alignment on Multiple Line Appearances

This feature allows the Calling Line ID (CLID) to be displayed on up to 30 sets having line appearance (Appear & Ring or Appear only), for an incoming external call on a physical or target line. Simultaneous CLID improves customer service.

### System Speed Dial Increase

This enhancement provides users with the option 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.

### Answer DN/DSS shared key

On BCM 3.5, an Answer DN key also functions as a non-user-programmable DSS key for the relevant DN. When an Answer DN is administered to a set key, pressing that set key when the key is idle (no Answer DN call being processed), will result in a DSS intercom call being made to the relevant DN assigned to that Answer DN key.

### Passive Call Log

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. The benefit of this feature is that call logging takes place without a line assignment. Typical applications include call centers and external interfacing employees.

# Data Capabilities

BCM 3.5 integrates sophisticated data capabilities, which have been enhanced for the latest release. These enhancements help to increase BCM's scalability and reduce costs.

## New Features and Enhancements

- T.38 Fax over IP
- VoIP QoS enhancement and Universal T1
- Dual V.35 WAN

### T.38 Fax over IP

This feature supports fax communications over the IP network to other BCMs or to a Meridian 1 head office. 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.

### VoIP QoS enhancements and Universal T1

BCM 3.5 provides an new 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 3.5 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 new QoS enhancements, the BCM routing WAN interface must use the Digital Trunk MBM interface. Prior to BCM 3.5, this interface was only capable of supporting circuit switched voice connections

**Benefits**

- The addition of Layer 2 Fragmentation reduces the latency introduced in VoIP communications on low speed data connections when larger non-VoIP packets have started their egress out of the BCM router and across the WAN connection
- Note that these fragmentation techniques do not require the re-transmission of the entire interrupted packet, so very little additional overhead is introduced to gain the QoS enhancements available by lowering the latency and jitter on the VoIP packet

- IP/UDP Header Compression reduces the size of the header information in the VoIP packets that egress the WAN link, which reduces the amount of bandwidth consumed by VoIP traffic

### Dual V.35 WAN

BCM 3.5 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.

# Messaging

BCM 3.5 includes several enhancements to Messaging features designed to improve customer service while reducing cost of operation. Messaging features now include centralized voicemail and Auto Attendant has been enhanced to include Holiday Schedule features and Park & Page.

## New Features and Enhancements

- Centralized Auto Attendant and Voicemail
- Auto Attendant Holiday Schedule
- Auto Attendant Park and Page
- Increased channel capacity
- Unified Messaging enhancements

### Centralized Auto Attendant and Voicemail

With Centralized Auto Attendant and Voicemail, the BCM acts as a central host for Norstar/M1/BCM connected over MCDN keycode. This capability will support up to ten networked platforms, CDP Private Number Plan, one level tandeming, MCDN features, voice messaging and auto attendant.

### Auto Attendant Holiday Schedule

This feature is an enhancement to the existing greeting schedule for the methods of operation based on time of day. It allows an administrator to set up an auto attendant schedule on a weekly basis, from Monday to Sunday, and includes an exception calendar that allows programming for holidays or special occasions. The enhancements to this feature allow administrators to make changes to the standard weekly schedule, if necessary. This means that administrators can create special greetings and menus for holiday periods (e.g. Christmas, Thanksgiving any time throughout the year. Scheduling can be administered via voicemail manager or NCM, and it applies to messaging, auto attendant and CCR.

### Auto Attendant Park and Page

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.

### Increased Channel Capacity

With previous versions of BCM, 32 channels are available for Voice Messaging, Call Center and IVR. Of the 32, call center and voice messaging can have access to 16 that must be shared (they are not assigned). IVR can grow to 24 channels, depending on the size of the application. If IVR is using 24 channels, only 8 channels remain for voicemail and contact center. With BCM 3.5, the total number of channels available to be shared between voice messaging and call center is increased to 32 channels, thereby improving application (voice messaging and call center) performance and simultaneous access.

### Unified Messaging Enhancements

This feature increases the current number of Unified Messaging clients from 100 to 200.  The BCM will require 512 MB of RAM to support more than 100 clients.

# Voice over IP

With proven Voice over Internet Protocol (VoIP) technology, BCM 3.5 can boost business performance and accelerate business success.

## New Features and Enhancements

- H.323 interoperability
- Media path codec renegotiation and administration
- Integrated IP Music on Hold
- Enhanced i2050 USB Audio Kit
- i2004 wideband demo
- SIP trunk & data services support

### H.323 interoperability

BCM 3.5 offers H.323 interoperability with Succession 1000M, Succession 1000 and RadVision ECS 3.0/3.2 Gatekeeper.

### Media Path Codec Renegotiation and Administration

This feature eliminates "dropped call" situations caused by codec negotiation limitations. With previous BCM releases, when a VoIP call is transferred from one terminal to another or one BCM to another and the new terminal or BCM gateway is not configured with the same codec as the original device, a call may drop. With this new feature, when an incompatibility is detected, the codec will be renegotiated and a compatible codec will be found, if available. Codec Renegotiation 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.

### 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 3.5. Common formats such as .wav or .ra formats are supported. With this feature, customers now have two new 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 3.5.

### Enhanced i2050 USB Audio Kit

This new USB audio kit for the i2050 Software Phone provides hard keys for easier feature access for IP telephony client users: answer, mute, hold, volume and goodbye.

### i2004 Wideband demo

This feature is for demonstration purposes only. It can only be demonstrated on i2004 hands free calls since the i2004 handset does not have wideband support at this time. The main benefit of this feature is clarity, as the wideband codec enhances the audio quality for crystal clear conference calling.

### SIP Trunk & Data Services Support

SIP trunk support will be piloted on BCM 3.5, 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 in this pilot.)

BCM 3.5 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.

Nortel Networks will continue to evolve SIP integration on our IP telephony platforms, including BCM, to enable future VoIP services and multimedia applications on platforms such as Multimedia Call Server (MCS) 5200 for the carrier environment and Multimedia Call Server (MCS) 5100 for enterprise customers.

# Voice Networking

Enhanced networking features on BCM 3.5 extend enterprise communications with seamless Succession call server networking.

## New Features and Enhancements

- H.323 interoperability
  - Meridian IP Trunk 3.0/3.0.1 interoperability
  - Succession 1000 interoperability
  - Norstar IP Gateway interoperability and Radvision Gatekeeper ECS 3.0
- MCDN Tandem E.164

### H.323 interoperability

BCM 3.5 will interoperate with the following Nortel Networks products using H.323 IP Telephony standards:

- **Succession 1000 Release 3.0, Succession 1000M/Meridian 1 (IPT 3.0/3.01):** 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 the IPT.
- **Succession 1000 Release 2.0:** BCM 3.5 will be interoperable with Succession 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**:

### 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.

# Call Center

Enhanced Call Center features have been added to BCM 3.5 and are designed to improve the customer experience.

## New Features and Enhancements

- Expected Wait Time
- Agent Help
- LAN CTE enhancements

### Expected Wait Time

This is a new routing step that, when encountered, will play an audio message for traditional voice-originated contacts, or, it will send a text message for chat inquires and Web-initiated voice calls. The message, whether text or audio, will provide the caller with information about how much time they can expect to remain in queue prior to being connected to an agent. 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.

### Agent Help

This feature 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 voice mail message.

### LAN CTE enhancements

LAN CTE is the interface used by third party developers to create specialized CTI (computer telephony integration) applications. BCM 3.5 delivers enhancements to LAN CTE that provide the ability to pass information between applications, additional set control and status, and increased application robustness. All of this adds to the breadth of third party applications that can be developed — and, in turn, increase the BCM's value to customers.

# Interactive Voice Response (IVR)

IVR on BCM continues to be enhanced. For the small or medium site customers that require an IVR solution, BCM 3.5 provides huge benefits. With IVR, businesses can increase their availability providing improved levels of customer service without increasing their costs. They can process more transactions and a business can be "open" 24 hours a day, 7 days a week.

## New Features and Enhancements

- IVR 1.1
  - IVR Park & Page
  - IVR Voice Mail Integration

### IVR 1.1

This new feature provides an interface that allows users to easily add new functionality that is supported by BCM but not by IVR. It simplifies the delivery of information captured (DTMF), retrieved (database look up) or to be sent from one application to another. IVR captures digits from the caller, performs a database lookup and makes information available to drive a screen pop. The toolkit allows IVR Park & Page and IVR Voice Mail Integration.

#### *IVR Park & Page*

This new 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 Voice Mail Integration*

This feature allows IVR callers to be sent to a voice mailbox and then sent back into the IVR script for further call options.

# Virtual Private Networks (VPN)

BCM 3.5 enables high-security Virtual Private Networks (VPNs) over the public Internet. VPNs can be easily established with a Nortel Networks Contivity Extranet Switch, Shasta 5000

Broadband Service Node or another BCM.  Secure Socket Layer (SSL) encryption is added in BCM 3.5 to secure data and hide passwords.

## New Features and Enhancements

- Secure Socket Layer (SSL) encryption
- Secure Shell (SSH)
- Other Security features

### Secure Sockets Layer (SSL)

In addition to VPNs, BCM 3.5 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. On BCM 3.5, 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.

### Security

BCM 3.5 includes a number of security enhancements, including Secure Socket Layer (SSL) encryption and other minor enhancements in the areas of control, audit, authentication, availability, confidentiality and integrity, that ensure voice and data traffic is secure.

### *Access Control*

This feature enhances password policies and password management. Password policy can be configured through UM for length, age, history and complexity. Upgrades to BCM 3.5 require password changes to meet password policy. In addition, NCM management of all BCM versions enforces BCM 3.5 password policy. The BCM Monitor password is stored using stronger encryption techniques.

Access Control on BCM 3.5 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. The new Apache authentication uses NT password DB (eliminates secondary internal password DB).
- **Shared Drives Disabled** Shared drives on the network will be disabled by default after upgrades to BCM 3.5 & Factory default for BCM 3.5.

### *WIN NT OS Hardening*

This feature provides Service Pack 6a (128 bit encryption version) and SRP (Security Roll-up Package), a collection of Security hotfixes since SP6a and other miscellaneous security hotfixes. It also provides a default DCOM setting to encrypt data for application that use default option. In addition, it offers more restrictive file access permissions on the registry, files and folders, as well as the elimination of OS components not used by BCM (POSIX & Outlook Express).

#### *Apache Web server upgrades*

BCM 3.5 involves an upgrade to Apache Version 1.3.27, which helps to reduce virus vulnerabilities.

#### *Denial of Service (DoS) prevention improvements*

BCM 3.5 offers improved prevention of Denial of Service (DoS) due to malformed packet and stress testing of data network, thereby reducing system overload.

#### *Event logging & user level privilege improvements*

The maximum size for WinNT event logs has been increased to 3 MB each for the Application Log, Security Log and System Log. With BCM 3.5, Unified Manager user levels are mapped to NT User Groups that are non-admin. NT administration capabilities are not available to non-admin users through other interfaces such as Telnet. Dialup user group members are given read-only access rights in Unified Manager.

#### *Limiting BCM information revealed publicly*

With BCM 3.5, 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.

### BRU Data Encryption

BRU data encryption facilitates the secure transfer of BRU files. The encryption algorithm is based on a static password using RC4 40 bit encryption to provide any system with the ability to apply the restore. This type of encryption prevents malicious users from viewing sensitive data in the BRU backup files through simple text viewers, such as notepad. The data is encrypted during the backup operation and is decrypted during the restore operation.

### CDR Security Enhancements

CDR data contains sensitive telephony data which is transmitted to servers for collection. The data needs to be transmitted securely to ensure data confidentiality.

Enhancements include:

- For the CDR Pull method, the data is sent through a secure SSL interface.
- For the DCOM (real-time) interface method, the data is encrypted within the DCOM protocol but this DCOM encryption is not available for Win9x client OSs.

# System Management and Software Options

BCM 3.5 adds new monitoring capabilities as well as additional management features. These enhancements include Network Configuration Manager 3.0, a powerful, global, template-based solution that simplifies the management of large networks containing hundreds, or even thousands, of units.

## New Features and Enhancements

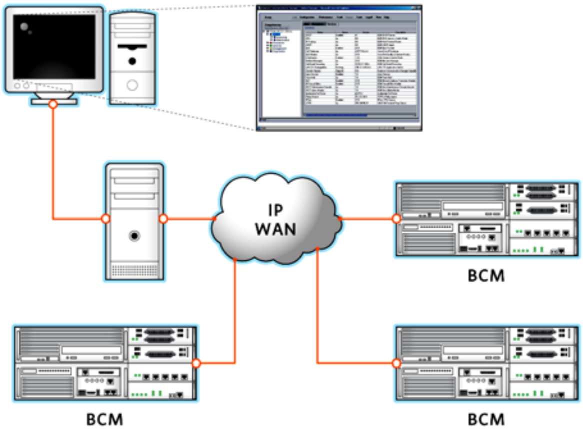
- Network Configuration Manager 3.0
- BCM fault management
- SNMP Trap Dial-out
- SNMP Trap Maintenance
- Secure Dial-back
- Unified Manager Help
- BCM Line Monitor Enhancements
- BCM Monitor Security Enhancements
- OAM Configurable Security Options
- Server Message Block and Secure Channel Improvements
- APC Smart UPS integration

### Network Configuration Manager 3.0

Network Configuration Manager 3.0 is a multisite management application that provides centralized configuration and system management capabilities for a multi-site BCM network. NCM enables customers and/or channel partners to significantly reduce the total cost of ownership of their BCM systems by reducing the time it takes to make bulk-programming changes. With BCM 3.5, the NCM 3.0 software includes support for patch distribution, voicemail backup & restore, enhanced reporting and support for new BCM features.

Network Configuration Manager for BCM meets the needs of enterprises with replicated sites that are organized to manage the network themselves. Customers who do not wish to, or are not equipped to, manage the network themselves can choose to have their networks managed by a Nortel Networks Channel Partner.

The NCM solution for BCM consists of three elements: (1) NCM server software, (2) client software, which allows multiple network operators to simultaneously use NCM, and (3) the BCM resident multi-site management authorization keycode. Multisite management is activated by keycode on each BCM.




NCM provides the ability to apply programming changes to any or all BCM systems within a network from a centralized location. The following changes can be applied, with new 3.5 features in bold:

- Centralized database and configuration changes
- Launch of Unified Manager and Desktop Assistant Admin ProAE
- Auto attendant greetings and hours of operation
- Call center greetings
- Custom call routing tree settings and telephony call routing information
- Application of software option keycodes
- Bulk password changes
- Deploy, upload or remove IVR scripts
- Deploy, replace, remove, upload audio files
- **Distribute software patches**
- **Transfer files via FTP**

- **Store data off the NCM database**
- **Backup and restore voicemail files**
- **Enhanced reporting capabilities**

***NCM Patch Distribution***

NCM includes the capability to distribute patches to multiple BCM systems employing the BCM patch process. This includes the ability to schedule deployment of patches to BCM systems, the initiation of the BCM patch process on designated BCMs, and confirmation within the NCM audit log of 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 Voicemail Backup***

With BCM 3.5, voicemail is included in the NCM backup and restore process, so voicemail 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 File Transfer Protocol (FTP)***

This feature allows files greater than 32 MBs to be transferred between an NCM and FTP server, thereby simplifying network administration and reducing back-up.

***NCM off database storage***

NCM provides file archive storage outside of an embedded database, minimizing network downtime in conditions when a server fails. Typical applications involve multisite or multi-BCM networks with an NCM back-up server.

***NCM data back-up procedure***

This feature enables users to quickly backup NCM specific data and archives. Additions include support for new 3.5 features, such as integrated IP Music on Hold, increased system speed dial lists (new wizard), Auto-Attendant Holiday Scheduling (new wizard), Auto Attendant Park & Page, SIP trunking (new wizard), SNMP Trap Dial-Out, Secure Dial-Back, and UPS client. This procedure reduces the occurrence of lost data by performing immediate backup and archive (opposed to waiting for a schedule to run).

#### *NCM Configuration Report Enhancements*

This enhancement expands and customizes the report generation capabilities in NCM. The report generation includes modifications to the existing five standard reports to support new information being imported into the NCM database. The report configuration tool allows the user to select which fields from the NCM database they wish to be included in the report.

This feature increases network performance by allowing users to easily export selected data into a spreadsheet or a third party tool such as Crystal reports, in order to customize the report format.

#### *NCM Try-and-Buy Keycode*

A key code 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 Fault Management

This capability has been enhanced to improve the reporting of events via SNMP by adding telephony events, event descriptions and IDs to minimize duplication.

### SNMP Trap Dial-out

A dial-out connection can be automatically established when a fault condition occurs, so that the event information can be delivered to a centralized NOC. Inexpensive remote management can be offered in sites without an always-on IP connection.

### Secure Dial-back

Secure Dial-back will allow the BCM to respond to a dial-in call from a network administrator by invoking a secure dial-back to a predetermined number in order to enable the administrator to set up a dial-up connection for remote management.

### OA&M Configurable Security Options

Several OA&M security settings are configurable via Unified Manager and provide administrators the ability to harden or loosen security settings as required for their environment. Typical applications for this feature include the government or military institutions.

The configurable security settings include:

- setting the Authentication type
- Option to Clear the page file on shutdown
- SMB signing level for client and server
- Domain Secure Channel signing and encryption options
- Enabling and disabling SSL web access
- Setting Web encryption level
- Password policy configuration
- Lockout policy configuration and control.

Additional Unified Manager security related capabilities include:

- Certificate upload capability
- PuTTY download
- Telnet server enabling/disabling
- Modem enabling/disabling
- Dial-in access.

### Server Message Block and Secure Channel Improvements

SMB is used in drive shares and named pipes, and RPC communication used by client applications such as Unified Manager, LAN CTE and BCM monitor.

SMB sessions are configurable through Unified Manager and in BCM 3.5, policy changes have been made to make them more secure:

- Idle SMB sessions are disconnected after 15 minutes
    - Re-established connections using a single sign-on to automatically re-authenticate the user
- Null SMB session (no user ID or password used in the connection) will not have any access or enumeration capabilities for file shares and named pipes
- Digital signing allowed, if possible through client negotiation
    - Digital signatures allow for the detection of man-in-the middle attacks and message authentication prevents fake message attacks
    - Provides a compatible interface to Win 95/98/Me (default OFF)

- Secure channel utilized during BCM to Domain controller interaction
  - The CDR and CTE client applications are the only BCM applications that are supported with a Domain controller.
  - Secure channel set to digitally encrypt, if possible, through client negotiation

### APC Smart UPS integration

This feature allows for the graceful shutdown of BCM under power failure conditions and the ability to provide a warning message under these shutdown conditions. The UPS is configured to shut down the BCM two minutes before the battery is drained. The ability to turn the power off after system shutdown is only provided on BCM200/400 and will not be available on BCM 1000, due to the BCM 1000 hardware (Power Supply) limitation.

APC Smart UPS will be supported on BCM 3.5 through the serial cable UPS interfacing solutions.

Both the Serial solution and APC's PowerChute Plus management software will be integrated with the Unified Manager UPS configuration software and the UPS status monitor software. The Serial interface supports only one UPS to BCM connection.





# Appendix and Glossary

# Appendix

# 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 Companion component that is mounted on walls and ceilings to provide a radio link to an office or other area where Companion 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**

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.

**T7316 telephone**

The Business Series Terminals T7316 telephone that has a two-line display with three display buttons, 16 memory buttons with indicators and 8 memory buttons without indicators.

**M7324 telephone**

The Business Series Terminals M7324 telephone that has a two-line display with three display buttons and 24 memory buttons with 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**

If users have several calls waiting at their telephone, they can invoke the Call Queuing feature to answer them in order of priority. Priority is given to incoming calls, followed by callback and camped calls.

**Callback**

If a user parks, camps or transfers a call to another telephone and no one answers it, the call will ring 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.

**Central Answering Position (CAP)**

An M7324 telephone that has been designated as a CAP under CAP assignment in System programming. The CAP provides backup answering and can be used to monitor the telephones within a system.

**Central Answering Position (CAP) module**

A module connected to an M7324 telephone that provides 48 additional buttons that users can deploy as autodial buttons or feature buttons. A maximum of two CAP modules can be connected to a single M7324 telephone.

**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.

**client**

A client is a computer system or process that requests a service of another computer system or process. A workstation requesting the contents of a file from a file server is a client of the file server.

**cold start**

A cold start occurs when all system programming is lost. This can happen after a major event such as an extended power failure.

**Companion 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.

Companion Wireless provides mobility in the workplace. Calls that used to ring just at your telephone set can also appear and ring at your portable.

**Companion portable telephone**

Handheld wireless telephones that allow complete mobility within the reach of Companion 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 indentifier (DLCI)**

The DLCI is used to identify 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**

After a specified number of rings, this feature 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 dialup connection is a temporary connection between computers that is established over an analog or digital phone line.

**Differentiated Services (DiffServ)**

DiffServ is 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 indentifier.

**DN**

See Directory number.

**DNS**

See Domain Name Server.

**domain name**

The domain name is used to organize Internet names into manageable groups, such as nortelnetworks.com.

**Domain Name Server (DNS)**

The domain name system or Domain Name Server is 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)**

DPNSS is 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)**

DHCP is 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.

# E

**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**

Event messages are stored in the system log and displayed during a Maintenance session. They record a variety of events and activities in the system.

**exceptions**

See Overrides.

**Expansion Media Bay Modules**

Expansion Media Bay Modules 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)**

Extended Data-Out (EDO) is a form 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**

FAX 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**

Filtering is 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 frame is a unit of data transmission in a local area network.

**frame relay**

A frame relay is 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.

# M

**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.

**Meridian 1 ISDN Primary Rate Interface**

A protocol used between members of Nortel Networks 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) voicemail 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.

# N

**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 Companion 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**

Lets 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 server s 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 Business Communications Manager 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 Meridian 1.

**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**

See Simple Network Management Protocol.

**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**

See Transmission Control Protocol/Internet Protocol.

**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.

# U

**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.

# V

**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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