IP Telephony and Converged Networks

An Applications and Technologies Overview for Mid-market Companies

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Introduction

Though much of the buzz about converged voice/data/video networks has focused on larger firms, mid-market companies are taking advantage of convergence with equal success. In the process they’re learning that IP Telephony can be the catalyst that makes major data network investments affordable. In today’s economy, not only do these investments have to be affordable, they have to have quantifiable Return on Investment (ROI) projections with aggressive payback schedules.

Increasingly, customers want to add voice to the many services and applications that travel over their Internet Protocol (IP) networks. Often the force behind the convergence decision is not the telecom manager but the Management Information Systems (MIS) director, the Chief Information Officer or the Chief Financial Officer. When these decision makers sit down to plan a major data system investment and upgrade - for instance, to integrate supply chain communications or implement an enterprise resource planning (ERP) system - adding IP Telephony to the project can strengthen the business case, and add sufficient value to make the entire project feasible.

Indeed, the business cases for some projects have been so compelling that companies have replaced existing LAN/WAN networks and nearly-new PBXs in order to capitalize on the capability and efficiency of a single multi-purpose voice/data/video network.

This paper explores the trends developing around IP telephony, discusses some specific instances in which mid-market companies have implemented IP Telephony and other voice services as part of a comprehensive networking solution, and then moves to a discussion on the ROI-rich applications of a converged network.

We continue with a discussion of the technology of converged networks, starting with a comparison of traditional circuit-switched and packet-based networks. We’ll explore the components and applications found in a converged network, discuss ways to move into converged networking, and then conclude with a look at some of the issues encountered in converging your networks.

Mid-market Drivers for Converged Networks

Given that conventional circuit-switched networks already deliver high quality voice communication, while IP networks require special care in design and operation to match that voice quality, it's reasonable to ask why businesses are focused on converged networking. The answer is that converged networks promise to deliver a compelling set of benefits, and the ROI provided by those benefits creates a convincing case for implementing IP Telephony.

As companies increasingly adopt an enterprise-wide view of their voice and data needs, the logic of using a single network becomes clear. The ROI for converged networking is compelling. Customers implement converged networks to reduce costs throughout the organization. Simplified network administration, improved access to business applications and reduced costs to connect multiple locations, are but a few of the reasons driving the decision to integrate networks.

New developments have advanced convergence technology to the point where we can build converged VoIP systems that deliver the reliability, call quality and features companies need to operate effectively - and in the process save money and open the way for new business applications.

As a result, the momentum for converged networks is growing fast. A survey of 500 U.S. firms by market analyst Phillips InfoTech predicted that purchases of new IP telephony lines will increase at a combined average growth rate of 190 percent, to nearly equal sales of traditional PBX lines by 2003.

In addition, a majority of both small and large firms expect to adopt IP Telephony within just a few years. InfoTech found that 91 percent of large firms and 61 percent of small and mid-sized firms expect to implement IP Telephony by the year 2004. The InfoTech survey also found that data decision makers were far more likely than telecom decision makers to be convinced of the benefits of IP communication.

IP Telephony - The Experience of Wheeler Trucking and Heart Partners

This expression of strong interest in IP Telephony dovetails with our marketplace experience. Increasingly, mid-market firms are investing in solutions that combine voice and data on a single IP-based infrastructure. The experience of a growing regional trucking company and a partnership of cardiologists - let’s call them “Wheeler Trucking” and “Heart Partners” - provides a good example.

Serving a multi-state region through eight locations, 150-person Wheeler Trucking had acquired a new IBM computer as part of a program to host its enterprise-wide business applications. A discussion of wide area network (WAN) support with Wheeler’s information technology director made it clear that the company wanted to create a single, standardized infrastructure across the business.

A detailed network assessment was conducted, producing a recommendation for a complete network redesign, including replacement of the existing LAN/WAN equipment and a nearly-new PBX. As part of the solution, an IP-PBX and voicemail system were installed at Wheeler’s corporate headquarters, connected to the remote locations through a voice-enabled data network.

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Wheeler's new system has generated significant cost savings and operational efficiencies. Linked to headquarters over managed IP circuits, Wheeler's remote offices now use the central call processing and voicemail capabilities without needing their own local key. PBX or voicemail systems. Intra-company calls are carried on the spare capacity of the data WAN, avoiding the public toll network with its long distance charges.

Administration of Wheeler's new system is also far simpler and less costly - a fully converged network typically yields a 20 percent savings in network administration. With a single architecture, Wheeler now manages all of its multimedia equipment and services centrally, with one group of technicians and one set of procedures. The complex administration a PBX requires for moves and changes is no longer necessary, since IP telephones can be moved from place to place as easily as a computer. End users now have one Ethernet wire to their desktop for both telephone and computer. Moves can be handled by the user, instead of a technician.

In the case of Wheeler Trucking, IP Telephony was not part of the original request for proposal. Wheeler believed its newly purchased traditional voice system was adequate to meet its needs. As Wheeler learned of the added benefits inherent with a converged network, however, IP Telephony became a key to the success of the company's enterprise-wide strategy.

For Heart Partners, an opportunity to add two new clinics coincided with a need to upgrade to a new NT version of the healthcare provisioning application that served its five offices and 100 associates. Supporting the new software would require an upgrade of the data LAN/WAN, and that created an opportunity to further capitalize on the investment by moving to a single, unified voice/data network. Heart Partners installed a centralized IP PBX and voicemail, connected to 90 IP telephones in the offices and clinics through a common data network.

This IP Telephony solution produced significant savings. The small support staff was easily able to administer a single converged system. With nearly $200,000 a year in reduced network charges, Heart Partners saved enough to also acquire firewalls, security systems, applications servers and carrier services without exceeding its budget.

Advantages and Applications of Converged Networks

In choosing IP Telephony, mid-market businesses are taking advantage of the capabilities made possible by converged multi-service networks, which as detailed below, are driving mid-market firms to bring their voice and data systems together:

Centralized Administration -- Managing one network, with one wire to the desktop and with centralized administration, offers great potential for savings. With one network there's no need to employ separate administrators with different skill sets for separate networks. According to the editors of Convergence magazine (formerly Computer Telephony), convergence is central to the enterprise, where all businesses are interested in improving efficiency and reducing costs.

Lower Costs for Adds, Moves, and Changes -- Because the phones on an IP Telephony system are IP devices, it is not necessary to bring in a technician to move a phone. End users can do the job. It can be as simple as unplugging the phone from its current location, and plugging it back in at its new location.

Reduced Toll Expense -- By sending intra-company voice calls over the data network or a Virtual Private Network (VPN), instead of the public toll network, converged communication systems can significantly reduce the charges a company pays for long distance calls.

This approach is particularly lucrative for international calling, where per-minute rates remain high, and for smaller companies that can't command the deep discounts large firms enjoy on public network (PSTN) toll rates. In the highly competitive U.S. market, competition has driven long distance costs so low that toll bypass alone may not justify the implementation of IP telephony. Where a data network is already in place, however, it may pay to transmit voice over that network even within the U.S.

Remote Offices -- IP telephony systems extend the features, functions, and applications of a central PBX to serve a branch office location.

The remote location receives access to all the capabilities of the large, IP-capable communication server, without the cost of installing a separate PBX at the remote site. System administration is centralized for additional savings.

Improved Options for Teleworkers - Remote workers -- including telecommuters, "road warriors" and home-based customer service agents -- can call the office over an IP connection and conduct business almost as if they were on site. "Softphone" programs turn a laptop or desktop computer into a full-featured telephone that uses the call management capabilities of the host PBX to deliver such capabilities as hold, transfer and call conferencing. Other programs can incorporate remotely located customer-service agents into a central Customer Relationship Management (CRM) system. By enabling agents to work effectively at home, such systems can help contact center operators improve agent retention and reduce the cost of turnover.

Open Standards and Enterprise-Wide Applications -- Because IP communications systems are based on widely-accepted industry standards and "speak the language" used in data communications, they simplify the task of adding new applications from a variety of sources, and implementing those applications across the business. Standards-based converged systems make it far easier to integrate CRM and other eBusiness applications across the enterprise. Such initiatives are far more difficult where vendor-specific proprietary standards are in place.

Network Scalability -- IP networks can be more readily expanded as a business grows and needs change. Rather than having to add additional station cards, an IP Telephony system can be expanded simply by plugging the IP endpoint into an available switch port on the network.

Unified Messaging -- Standards-based unified messaging systems bring voice messages, e-mail and fax together on a single system and enable users to access, manage and respond to their messages from anywhere, by computer or by phone. Because all messages are stored on the Microsoft Exchange server, companies avoid the cost to purchase and administer a separate voicemail server.
Some of the same benefits are achieved by "integrated" systems, which store voicemail, fax and e-mail messages on separate message servers and use software to link the systems and give users "one call" access. Such systems enhance user productivity and, because they store messages on at least two servers, may provide enhanced reliability compared to single-store systems. The downside is that the multiple servers must be administered separately.

Desktop Integration to personal information manager and contact manager programs. By using features like Computer Telephony Integration (CTI), an end-user can utilize their ACT!™, Microsoft Outlook®, or GoldMine® contact register to make calls, speeding the flow of work and simplifying data management.

Web-enabled contact centers -- Companies are learning to integrate their customer relationship management (CRM) technologies so that contact centers, toll-free 800 numbers and Web sites work together to improve customer service and sales. For instance, by setting up IP telephony gateways to channel IP calls from the Internet into the contact center, companies can enable PC users who are properly equipped to browse their Web sites and at the same time talk to agents over an IP voice connection. Such an arrangement can help the company save money on 800-number charges, close business instantly and strengthen customer service.

IP Screen Phones -- Screen phones feature large color displays that allow users to view applications and corporate directory data, surf an intranet or the Internet and manage telephony features such as speed dial lists and call logs. Users can operate visually, rather than using the telephone keypad.

XML and Java-based applications -- The open architecture of a converged platform enables software developers to create new and compelling applications. Using the Computer Telephony Integration (CTI) links these platforms offer, as well as the large, pixel-based telephone displays, developers can deliver rich, new and often highly customizable functionality to address specific needs. For instance, these new display sets can serve in such environments where the footprint or operation of a PC would serve as well as a single unit phone. Such situations could include at the hospital bedside, in public or private waiting areas and in the executive suite. Other examples include:

- Directory Software is now available to automatically synchronize the data in IP telephony systems with other corporate LDAP (lightweight directory access protocol) directory data. For example, an administrator entering a new employee's name in a corporate directory database can automatically trigger the communications server to provide a voice message box and telephone extension.

- Customized applications can turn a screen phone into a multipurpose display. Clients in a lounge area, for instance, could use the phone to check flight schedules, order tickets or other "yellow pages" functions. Office users can display company directory information and dial with a touch. Display applications for lawyers and other professionals can make it easy to capture customer information for later use in billing.

- Security-driven applications can activate phone features or display specific graphics based on such factors as user ID or time of day, and enable or prohibit their use based upon a combination of factors.

Since these applications can be developed with standardized XML and Java programming environments they can be re-used across a variety of platforms, making them more economical to acquire and support.

Network Comparison: Switched Circuits and Packet Networks

Most of today’s enterprise communications systems consist of two separate networks:

Data Network -- A packet-based LAN links desktop computers and servers dedicated to data communication. The connection medium may be copper twisted pair cable or fiber optic cable. Many LANs today use the Ethernet protocol. LANs are connected to other enterprise locations and the rest of the world using the Public Switched Telephone Network (PSTN), leased private lines, VPNs (Virtual Private Networks), frame relay, and the Internet.

Voice Network

A traditional enterprise circuit-switched voice network is controlled by a switch and call-processing server called a Private Branch Exchange (PBX). The PBX makes circuit connections to desktop telephones using a network of copper twisted pair wiring. The enterprise voice network is in turn connected by trunk lines to the Public Switched Telephone Network (PSTN).

Let’s compare the operations and advantages of traditional circuit-switched voice networks with those of packet-based IP networks.

Circuit-Switched Networks -- Circuit-switched networks have operated successfully since the telephone was invented more than a century ago. They are well suited to voice communication.

In circuit-switched communications, switching systems set up a dedicated physical connection or circuit for each call — a direct connection between two points, carrying full duplex (bi-directional) voice at 64 Kilobits per second (Kbps), a speed that delivers acceptable voice quality — what the industry calls "toll quality" voice. Full duplex communication enables both parties to speak and hear the other party at the same time, without blocking either party.

The switched circuit is dedicated to a call throughout that call’s entire duration. When the call is complete, the circuit is disconnected. The physical medium — which could be a pair of copper wires, a segment of spectrum in a fiber optic link or a channel in a high-speed circuit, such as a T-1 or international E-1 line — is then available for use on another call.

Because circuit-switched systems dedicate a circuit to each call, they generally deliver high voice quality. Unfortunately, such systems tie up those network facilities whether the parties on a call are speaking or not. On a switched circuit, silence — even the brief silences between words — occupies the same band-
width as speech. And since the circuit is dedicated, other users are unable to use that circuit, even during the silences. So circuit-switched systems are not very efficient in the way they use network facilities.

**Packet Networks** -- Packet communications systems are designed to overcome such inefficiencies. Several packet communication systems exist - each with distinct characteristics.

In all packet systems - including Frame Relay, ATM (Asynchronous Transfer Mode) and Internet Protocol (IP) - information is digitized and organized into "packets" (also known as frames or cells) that contain addressing and other information, in addition to the payload of data. Packets are then transmitted in the direction of the destination computer.

As the packet makes its way toward the destination, routers along the way read the packet's address and send it along the way. Unlike circuit switched communications, however, the packets do not necessarily travel the same path. Because they travel various routes, packets can arrive out of order - those sent later in a message can arrive at their destination earlier than others. It's up to the computer at the receiving end to check for missing packets, request a re-transmission if packets are missing, and reassemble the packets in the correct order.

This "Connectionless" networking can be very efficient. Because the bandwidth available on a circuit is shared, packets from multiple communications can travel over the same circuit at one time. The "silences" are filled with information.

What's in a Converged Network?

A converged multiservice network is made up of a number of hardware and software components. Following is a brief discussion of each of those components.

**Transmission medium** - Networks use either conducted media (wires or cables) or radiated media (wireless, satellite or infrared) to connect transmitting and receiving devices over a distance.

- The least costly and most common conducted media system is unshielded twisted pair (UTP), made up of two thin copper wires, separately insulated and twisted around each other. Though UTP was installed originally for analog voice communications, it supports digital transmission as well. Over short distances of up to 100 meters or so, Category 5 UTP cable can support bit rates of 100 Mbps (megabits per second). UTP is subject to electromagnetic interference, though shielded cables are available for use where such interference would otherwise hamper communications.

- Fiber Optic transmission systems use thin glass fibers to conduct bursts of light and transmit extremely large amounts of digital information. They are immune from electromagnetic interference. This is the technology of choice for high-speed backbone networks used by communications carriers. Fiber systems are also used in the local loop to serve large customers, office buildings and office parks, and are available - though less commonly used - in enterprise local area networks.

- Coaxial cable systems are commonly used in cable television networks. Coaxial cable is capable of high speeds of 100 Mbps (megabits per second). Unlike UTP, coaxial cable systems are immune from electromagnetic interference, though shielded cables are available for use where such interference would otherwise hamper communications.

- Radiated media systems include satellite communications, focused terrestrial microwave systems (which use the horn-shaped and bowl-shaped antennas often seen on promontories and tall buildings) and various microwave and infrared systems used for short-haul communications in the local loop - the network segment linking the telephone central office with homes and businesses.

Of the various radiated media transmission systems, those used most commonly in converged enterprise networks are Wireless Local Area Networks (WLANs) that use radio frequencies in the Ultra High Frequency (UHF) or Extremely High Frequency (EHF) range.

**LAN Switches** -- LAN switches are intelligent hubs with basic routing capabilities, able to read the target address of packets and forward them to the appropriate port. Switches may be located at the workgroup level or in the network backbone. Switches that are capable of supporting switched Ethernet capability, and VLAN configurations are optimal.

**Routers** -- Routers are intelligent devices that support connectivity between LANs, and also between LANs and various WANs. They typically provide connectivity, addressing and switching functions and operate at the bottom three layers of the OSI model - the Physical Layer, Link Layer and Network Layer. Their relatively high level of intelligence enables routers to route traffic based on consideration of the network as a whole. Routers also have the ability to route data based on network policy. Thus they can provide various levels or quality of service (QoS), based on such factors as the user, the terminal, and the type of payload. For instance, network policy can be established to ensure voice packets receive the high priority and reduced delay required for quality audio.

**IP Telephony Gateways and Gatekeepers** -- IP telephony gateways (also known as circuit-switched gateways) are network devices that convert voice information from traditional circuit-switched analog form to packet-switched digital form, reversing the process at the other end of the call. In an enterprise network, for instance, a gateway could be used to convert voice circuit-switched voice signals from a PBX into packet form for transmission over the...
enterprise LAN or private network. Similarly, gateways are used as the mechanism to interface to the PSTN.

Gateways typically provide a number of analog or digital port connections on one side, and a 10 Mbps or 100 Mbps Ethernet interface connection on the IP network side. Most enterprise-level IP telephony gateway networks operate in a point-to-point mode, with matching gateways from the same vendor on both ends of the transmission path.

Codecs and signal compression -- Signal conversion is an important gateway task. Gateways incorporate digital signal processors called codecs (coder/decoders) that perform multiple functions, such as analog to digital and digital to analog signal conversion, signal compression and echo cancellation.

Signal conversion is necessary to convert analog (wave form) voice into digital form, and reconvert it into analog form at the receiving end.

The conventional method is Pulse Code Modulation (PCM). In PCM coding, voice volume is sampled precisely 8,000 times a second, and each sample is encoded into an eight-bit digital value, but not compressed. PCM requires bandwidth of 64 Kbps to achieve "toll quality" voice clarity - it's not full voice fidelity, but quite acceptable.

Compression is a key factor in sending voice successfully over a packet network. By compressing the voice signal, we can reduce the bandwidth required to transmit the signal. The signal is then decompressed by the codec at the receiving end.

In addition, the receiving codec may employ various intelligent continuity algorithms to reduce or eliminate the effect of delayed packets, jitter and packet loss by stretching the voice frames received earlier and blending them with frames received later.

These compression/decompression algorithms, such as G.729 or CS-ACELP and G.723.1 (ACELP MP-MLQ) can achieve compression rates of 4:1 or even 10:1, while still achieving voice quality that approaches that of PCM and is acceptable to the listener.

Call Processing Software -- Though the performance of an enterprise communications system is mission critical, users often take for granted the intricate operations a voice system must perform to deliver even routine call processing functions, such as three-way conferencing or call transfer. In a circuit-switched PBX, the thousands of lines of software code controlling these operations have been developed, tested, fine-tuned and improved over a period of decades. Thanks to this extended development period, circuit-switched PBX switches and software are robust, reliable and highly functional.

Users expect the same features and functions and the same level of reliability in an IP-based voice system, regardless of whether the call processing software resides on an IP-enabled PBX, a PC or a special purpose IP communications server. Users care little about the technology that delivers this performance, though technology makes a big difference to those who engineer and manage IP-based voice communications systems. Today IP systems are able to match the features and reliability of circuit-switched switches, and capture the savings, administrative advantages and operational flexibility promised by VoIP communications.

Terminals -- Whether they look like standard telephones or like computers, desktop terminals play an important role in the performance of the entire system. After all, terminals serve as the point of interface between the human user and the electronic communications system. Their function is critical.

For instance, IP terminals incorporate the codec that converts analog voice into a stream of IP packets, and reconverts the incoming packet stream into sounds the ear can understand.

VoIP networks incorporate two basic terminal types:

**Hard phones** look and work much like the circuit-switched phones widely used in business. They have the same kinds of feature and function buttons, display screens and handsets, and may incorporate such features as speakerphone, infrared port, computer connection port and headset jack.

A new addition in the hard phone category is the screenphone, a high-end IP telephone with a compact color screen capable of displaying corporate directory information, speed dialing lists and data from the Intranet and Internet. Screenphones are designed for executive use, and for providing graphical information to customers in such settings as airline lounges, financial services offices and hotels.

Softphones are desktop or laptop computers set up with special software that enables the computer to connect to the call server and function in place of a separate desktop telephone. Users speak and listen through a telephone headset or speaker and microphone, and operate the terminal with mouse, keyboard and on-screen display. The softphone program may offer a choice of user interface displays, either showing a telephone keypad on the screen, or a Microsoft Windows computer-style display. Extensive call control features are available, including multiple call appearances, caller ID, conference, speed dial, send all calls, and message waiting.

Quality of Service and Policy Management -- Because packet-based voice communication can be easily disrupted by packet delay, VoIP networks must be capable of minimizing delay by assigning higher priority to voice packets as they travel across the network. To achieve acceptable QoS, network elements must be able to tag voice packets with Class or Grade of Service markings, so they will be recognized and given priority as they move through the network.

For both the LAN and WAN, centralized policy management is the key to building converged multiservice networks that deliver the required QoS, security and cost savings. Policy management software allows network managers to define specific rules, or policies, governing how network resources will be used. For instance, you can establish rules that assign priority to delay-sensitive voice traffic and video traffic to ensure service quality meets your standards and supports your business goals.

Network Management Software -- Network management applications simplify the task of configuring, monitoring and troubleshooting complex networks. Such systems provide...
information systems managers with real-time information about network performance and enable them to control the variety of products in the network from a single point - in some systems, from any terminal with Web access.

**Key Issues in VoIP Networking**

**Standards** -- As of this writing the industry has not agreed on a single set of technical standards for VoIP communication. The debate over rival standards and proprietary technologies may continue for several years. While this situation has not halted the move to VoIP, the uncertainty around standards means purchasers should act carefully in choosing equipment and its associated standards, with the understanding that the interoperability of equipment from differing manufacturers may be difficult to achieve.

Among the standards under discussion:

H.323 standards 1 through 4 - Developed by the International telecommunications Union (ITU), the H.323 standards govern the transport of audio, video and data over IP-based networks. H.323 defines codec techniques, call control and channel setup specifications. The standard also includes several codec algorithms, such as G.711, G.721, G.722, G.723.1, G.728 and G.729. H.323 is currently the leading VoIP standard. A number of major suppliers have adopted H.323 for IP telephony.

Session Initiation Protocol (SIP) - A signaling and control protocol for creating, modifying and terminating multiparty sessions such as Internet conferencing and telephony.

Media Gateway Control Protocol (MGCP)/MeGaCo - MGCP provides signaling and call control for VoIP gateways and endpoints. The protocol is intended to simplify the tasks IP terminals must perform in a VoIP network, and open the way for less complex, less costly terminals.

**Voice Quality** -- Because they were designed for data, and never originally intended to meet the requirements of real-time communications such as video and voice, IP packet networks must be carefully engineered and managed if they are to provide voice communications of acceptable quality. Network planners must ensure that the network's performance in such areas as latency (packet delay), jitter (variation in delay) and packet loss does not hamper voice transmission quality.

**Delay and Jitter** -- Data networks can tolerate some delay, but they're intolerant of packet loss. In a data network, even if some data packets are delayed, communication is considered successful once all data packets have successfully arrived at their destination. Packet addressing ensures all packets arrive and are reassembled in the right order. To lose a packet, however, is to corrupt data and impair the completeness and quality of communication.

Packetized voice communications, on the other hand, can operate successfully even with some loss of packets, but cannot tolerate excessive delay. Packet delay of more than 100ms (milliseconds) can create discontinuity that's readily audible and disturbing to the listener. Jitter - the variation in packet delay caused by changing conditions in the network - is also disruptive.

Packet delay and jitter are only two of the factors network managers must deal with as they design IP networks to carry voice communications. We'll discuss these and related issues, such as Class or Grade of Service, in detail later in this paper.

**Quality of Service and Class of Service** -- The converged multiservice network must not only provide reliable transport for a variety of types of traffic. The network must also provide a consistent quality of service across the variety of LAN and WAN topologies a packet might encounter on its way from source to destination. This requires the consistent cross mapping of the prioritization schemes of the LAN and the WAN.

Most priority strategies use Class of Service (CoS) by assigning a priority level to a packet or frame. CoS is a classification method only and does not ensure a Quality of Service level. CoS is the method used to limit delay and other factors so that the network produces a resulting Quality of Service.

The LAN prioritization scheme that is the most viable at this time is the IEEE 802.1p/Q standard, which defines the open standard for tagging data frames and assigning CoS in a VLAN (virtual local area network).

**Quality in the Wide Area Network (WAN)** -- The WAN story is not so simple, with alternate options such as Differentiated Services (DiffServ), Multiprotocol Label Switching (MPLS) and several others. The predominant emerging standard for IP-based WANs is DiffServ.

Sometimes viewed as rivals, DiffServ and MPLS are in fact complementary developments that approach the QoS challenge from two different network perspectives.

DiffServ is a Layer 3 solution that addresses QoS requirements in a connectionless environment. Its main purpose is to standardize a set of QoS building blocks with which providers can fashion QoS-enhanced IP services. DiffServ is meant to be implemented at the network edge by access devices and then supported across the backbone by DiffServ-capable routers. Since it operates purely at Layer 3, DiffServ can be deployed on any Layer 2 infrastructure. DiffServ and non-DiffServ routers and services can be mixed in the same environment.

MPLS is a strategy for streamlining the backbone transport of IP packets across a Layer 3/Layer 2 network. Although it does involve QoS issues (MPLS can indicate Class or Grade of Service), that is not its main purpose. MPLS will help to build backbone networks that better support QoS traffic, but it entails significant changes in existing network architecture.

DiffServ and MPLS are independent developments that can function with or without each other's help. There is hope that they can be used together as access (DiffServ) and backbone (MPLS) counterparts. In the near term, DiffServ tackles IP QoS head-on, and it provides mechanisms for achieving both access QoS and backbone QoS across the network.

**NAT (Network Address Translation)** -- Routers using Network Address Translation reside between the corporate LAN and the Internet. NAT performs several useful functions in IP networks connected to the Internet:
Before embarking on a project to converge voice and data networks in any business, it's important to assess the capabilities of your existing network infrastructure. Most LANs are highly complex systems comprised of a variety of products, often from different manufacturers, installed at different times and sometimes designed to meet varying standards. Not all data networks or network components are capable of performing at the levels required for quality voice communication. Without significant improvements, some LANs are simply not ready for convergence. Network managers should clearly understand the technological, operational and financial challenges ahead before embarking on a LAN convergence initiative.

The challenge grows when you add the WAN to your converged networking mix. Both the VPN and the public Internet pose distinct challenges for voice communications.

While network administrators may have day-to-day operations of today's systems well in hand, the move to a converged network raises significant new issues. This is the time to decide whether the "home team" - both the human element and the network infrastructure -- has the bandwidth to surmount these issues. It may be wiser to request expert network assessment from a vendor experienced in overcoming the challenges of VoIP, convergence and LAN/WAN integration. Many vendors offer these services.

Transmission over the public Internet without using a VPN can also be problematic. Variations in network congestion can cause delay and packet loss that reduce voice quality below acceptable standards and make the public Internet an option for voice communication only when issues of low cost and convenience take precedence.

Security -- The public Internet with its shared infrastructure provides cost savings when compared to leased lines and private network solutions. However, those factors make Internet access a security risk as well. To reduce these risks, network administrators must use the appropriate security measures.

Encryption and Tunneling -- The "private" in virtual private networking is a matter of separating and insulating each customer's traffic so that other parties can't access or compromise the confidentiality of data. IPSec tunneling and data encryption does this by carving private end-to-end pipes or "tunnels" out of the public bandwidth of the Internet and then encrypting the information within those tunnels.

In addition to IPSec, an IP Layer 3 standard, there are two standards for establishing tunnels at Layer 2: the Point to Point Tunneling Protocol (PPTP) and Layer 2 Tunneling Protocol (L2TP). Neither which includes the encryption capabilities of IPSec. The value of IPSec beyond these solutions is that it allows for native, end-to-end secure tunneling and, as an IP-layer service, it also promises to be more scalable than the connection-oriented Layer 2 mechanisms. IPSec provides a robust architecture for secure wide-area VPN and remote dial-in services.

Authentication -- A VPN platform should support a robust system for authenticating the identity of end users.

Firewall Technologies -- A firewall is a network interconnection element that polices traffic flowing between internal (protected) networks and external (public) networks such as the Internet. Firewalls can be used to "segment" internal networks as well. Firewalls are only one part of an overall security strategy. They do not guarantee security by themselves, and must be complemented with other security measures, such as user authentication and encryption, to achieve a complete solution.

Bandwidth -- The provision of adequate bandwidth is an important issue in VoIP networks, where bandwidth restrictions can cause delay and degradation of service. In remote access situations, for instance, using analog dial-up (bandwidth < 56Kbps) may provide insufficient bandwidth to provide toll-quality voice.

Best practices in Converged Multiservice Network Design

Network Assessment -- Professional network assessment is strongly recommended. A network assessment program should accomplish the following tasks, either as a turnkey operation or with participation from the customer, to inventory the network and its capabilities:

- Identify all equipment in the customer's network, as well as physical and network layer information, device connections, network topology and device configurations through a site configuration survey.
Optimal voice quality is possible when LAN or WAN bandwidth is wholly controlled by the customer, so that bandwidth can be optimized for voice or video communication.

Reliability and redundancy -- Reliability is essential as customers load more and more of their mission-critical business applications on to a single, converged, multi-service network. Adding voice on to the network only increases its importance. One key to reliability is redundancy, and this can be achieved by proper network design. There are different ways to achieve the same goal. Some systems are designed to provide redundancies within cabinets; others are built in modular form and can be arrayed or clustered to deliver comparable results. An effective network design will place reliability in context with other requirements, such as accommodating growth, providing flexibility, controlling cost and meeting other key business criteria.

How to approach and provision -- The process of planning and implementing a VoIP network should include these steps:

1. Identify and evaluate your business problems to arrive at a solution that addresses both your immediate needs and your longer-term goals. This includes evaluating the return on investment provided by the logical options, such as an incremental versus an integral approach to convergence.

2. Be sure to assess the impact of your organization’s IT plan for network growth and utilization, new e-business/enterprise applications and business locations, as well as their specific voice services requirements.

3. Conduct a network assessment to evaluate the status and capabilities of your current network and determine what changes will be required.

4. Provision your network based on these evaluations and assessments.

Suitability for voice -- Not all data networks are suitable to be upgraded to provide the QoS support required for IP telephony. Even those networks capable of being suitably upgraded may not match the price/performance of an all-new replacement network. A network assessment should provide much of the information you need to help you make the rebuild/replace decision.

Use of silence suppression -- Silence is another form of information to a codec, and unless the silent periods in a conversation are suppressed, the codec will use network bandwidth to send packets across the network that say, in essence, no sound is happening. Voice Activity Detection (VAD) software prevents encoder output from being sent across the network when there is silence. The use of VAD can greatly conserve bandwidth, a benefit that is particularly important in the WAN.

Delay -- Packet delay or latency is the length of time it takes a packet to cross the network. Each network element, including switches, routers, distance traveled, firewalls and jitter buffers, will add to packet delay. Users will start to experience difficulties carrying on a conversation when the one-way delay exceeds 50 milliseconds (ms). In some less critical applications, however, such lower quality may be acceptable.

Delay is somewhat controllable in a LAN, since the enterprise has complete control over the infrastructure. In the WAN, inherent delays are encountered that may not be manageable. Service quality over the public Internet is highly variable and unpredictable, as any Web surfer can attest. Large delays are an unavoidable fact of life with most Virtual Private Network (VPN) products.

Jitter and Jitter Buffering -- Jitter is a measure of the variance in delivery time between pack-
ets. If high enough -- greater than 20 ms -- jitter can create noticeable problems with voice quality. To compensate, many vendors implement a jitter buffer in H.323 voice applications. The buffer smooths packet flow by holding incoming packets for a specific time before forwarding them for decompression. Unfortunately, this also adds to packet delay. The jitter buffer should be sized with an eye to how it will perform with other network elements, such as network topology and router queuing methods, that affect jitter in the network.

Packet Loss -- Packet loss occurs when packets are sent, but not received at the destination. To ensure good quality in a VoIP network, packet loss should be less than 0.2 percent. Related factors to consider:

- Packet loss is more noticeable in a constant pitch tone, and less detectable in the varying pitch of the human voice.
- Losing a group of contiguous packets is more troublesome than random losses.
- Loss of packets with larger voice payloads is more noticeable than loss of packets with smaller payloads.

Duplex -- A network with shared segments - a hub-based network, for instance - can deliver lower voice quality due to excessive collisions between packets. A network that is fully LAN switched end to end provides full duplex (it can send data in both directions at once) and is ideal for VoIP traffic.

Infrastructure compatibility -- One of the most promising aspects of open multiservice IP networks based on open standards is their potential to readily accommodate shared applications across the enterprise. Unfortunately, the standards governing multiservice networking and IP telephony are not universally agreed on. Some suppliers continue to use their own proprietary standards, while others choose one or another of the various open standards under development.

This unsettled state of affairs means multiservice networking equipment can't be simply purchased "off the shelf" to automatically operate with other existing or new equipment.

Purchasers must carefully appraise the standards and capabilities of the vendors they're considering to ensure they don't "buy trouble."

Conclusions

Both big companies and mid-market firms want the same things from their voice and data networks: economy and performance. Before making any investment, companies of all sizes need to feel confident of their expected payback, ROI, and ability of these converged platforms and applications to solve the business issues they face in the market today. Smaller companies are finding that converged voice/data/video networks form the basis of enterprise-wide solutions that simplify administration, trim long distance costs and make it far easier to integrate applications across the whole operation. As information systems managers review the advantages of a single network that handles all of their needs, their ability to add IP Telephony to the mix can provide savings that make the whole project pay off.

As they would prior to any complex undertaking, companies convinced of the benefits of converged networks must understand and plan solutions for the numerous technical issues of VoIP networking before embarking on a convergence program. Network providers experienced in voice/data convergence are well equipped to assess needs, evaluate the capabilities of current networks, recommend hardware and software and implement the solution. Some even offer comprehensive post-installation network management services.

Converged networks have become an accepted solution that can both reduce operational and administrative cost and increase an organization's flexibility and integration, providing the return on investment that managers today demand.