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The Definitive Guide™ To

Converged Network Management

Ken Camp

Introduction to Realtimepublishers

by Don Jones, Series Editor

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Chapter 1: Introduction to Unifying Network Management and Converged IP Communications

The convergence of voice and data networks has been evolving and gaining momentum for several years. Although networks today have not converged, they are moving toward convergence in several ways. Many organizations are implementing Voice over Internet Protocol (VoIP) in an effort to cut communications costs or leverage the competitive advantage of integrated services, and VoIP implementers often focus on voice quality and interoperability—important factors in the delivery of Quality of Service (QoS). However, convergence really means much more than that.

Today, converging networks and net-centric applications are changing everything about network management. Network administrators need to manage and monitor a wide variety of network elements. They have to understand more complex network events and respond more quickly than ever. Effective management for these integrated network technologies—data, voice, video, wireless, and so forth—is crucial to network operations. This guide will highlight the overall service management challenges facing enterprise business and identify common industry best practices for effectively managing an integrated, unified communications environment using VoIP. It will offer a series of systematic and holistic techniques for managing the total integrated network to ensure consistent service delivery and support for ongoing business operations.

The term convergence is generally used in reference to the integration of telephony with data services and applications as well as video onto a single network. This single network is frequently assumed to be the Internet, but the convergence of services is bringing voice and data networks closer together in many ways. These technologies all used dedicated, separate resources in the past but can now share resources and interact with each other, creating new efficiencies for business.

The IP data network is evolving much further than just VoIP. Video technologies are blending and overlapping VoIP. Video VoIP (VVoIP) is becoming an accepted business service. With new operating system (OS) evolutions ahead and increased difficulty in air travel, many businesses are seriously exploring video collaboration as an alternative approach to traditional travel.

Beyond video, mobility is a prime business consideration in today's business environment. The evolution of wireless technologies to broadband services increased productivity for mobile workers. Another looming aspect of convergence is the convergence of the wired enterprise data network with the wireless cellular networks. This fixed mobile convergence (FMC) will surely add momentum as the technologies and handsets mature.

A trend is underway in which voice and data communications are merging. The irresistible logic is that digitized voice is just another kind of data, so why not carry it on the same data links that handle all your other ones and zeros? The economies of convergence can be considerable—there is no need to build and support separate voice and data infrastructures when you can have just one. That combination of infrastructures presents the problem that convergence aims to solve: data people, who haven't worried about voice in the past, have to worry about it now. Similarly, people who used to specialize exclusively in switched voice circuits must adapt to the new environment.

From enterprise business to small business to consumer, the end user doesn't care what network delivers services. The ability to work from any single device, anywhere, any time is more in demand today than ever in history. This chapter will review the broad aspects and implications of convergence in several forms.

Convergence Covers Many Areas

Five years ago, when people spoke of convergence, they really meant the evolution to VoIP or IP Telephony (IPT). Over those years, convergence has emerged with many different shapes and serves multiple purposes. It means different things to different people. Let's briefly explore the different nuances of convergence to fully understand what is happening in the network technology evolution.

Infrastructure Convergence in the Network—the Wiring and Circuit Convergence

Infrastructure convergence has taken place over nearly 20 years in different forms. The earliest examples of infrastructure convergence were brought on within the corporate building, driven by advances in LAN technology and improvements in digital PBXs. This initial convergence was the shift to a single Category-5 unshielded twisted-pair (UTP) wiring run to every work cubicle. Wiring plans simplified to bring voice and data services on a single wired infrastructure in the office. This premise-based convergence was driven by cost and convenience, but it set the stage for further network infrastructure integration.

Large enterprise business struggled for years with the separation of telephony and data services. Organizations bought circuit connections from carriers in staggering numbers:

- Primary Rate T-1s (PRIs)
- Basic rate ISDN lines for telecommuters and small offices
- POTS lines for home offices
- Point-to-point circuits to connect PBXs
- Frame Relay T-1s, both full and fractional
- Point-to-point circuits for SNA and other data needs
- POTS lines for dial-up to Remote Access Service (RAS)

Billing and administrative tracking of all this circuit connectivity placed a huge burden on enterprise business. The validation of circuit billing and Centrex service billing as well as, in some cases, trying to validate billed minutes of use, is a huge, labor-intensive effort.

Reducing the number of circuits became an obvious solution. Virtual circuit technologies such as Asynchronous Transfer Mode (ATM) and Frame Relay demonstrated the practical potential for the large enterprise business to receive all voice and data services converged onto a single “fat pipe” circuit. Fiber optic cable began replacing copper wire in many areas as Synchronous Optical Networking (SONET), using lasers to transmit information, provided advances in carrying capacity of the physical media.

References for ATM and SONET and other technologies: Wikipedia provides an excellent basic framework for high-level explanation and history of many of the technologies described in this guide. There are hundreds of excellent technical books on the protocols and technologies mentioned.

Service Convergence over a Common IP Networking Environment

For telephony carriers, infrastructure service convergence was also driven by Internet dial-up access. The Public Switched Telephone Network (PSTN) has been finely tuned and optimized to support voice traffic. An average telephone call lasts 3 to 4 minutes in duration. Dialup access to the Internet presented a new traffic engineering challenge to the telcos. The PSTN needed to support Internet sessions that might last for hours or days.

The PSTN is designed using blocking switches. What this means is that if every customer connected to a local central office (CO) picks up the phone at the same time, some will be blocked. It isn’t cost effective to over-engineer a central office to support 100,000 concurrent users when traffic studies have indicated that only 30 percent might be using the telephone at any given point in time.

Dialup access to the Internet forced a change in thinking about how the Internet and PSTN interact. As Figure 1.1 illustrates, they began as separate, disconnected networks, but dialup access introduced touch points between the two. As technologies evolved, a converging phase was reached with numerous high-capacity connections between the two. That is the current state of network technology.

The fully converged infrastructure network of tomorrow will blend the PSTN and Internet so closely that many users won’t even detect a distinction. Voice calls might route off the PSTN, a wired LAN connection using a VoIP phone on the desktop, or a wireless handheld smartphone that uses WiFi in the office then hands a call off to the cellular network when the user moves out of WiFi range.

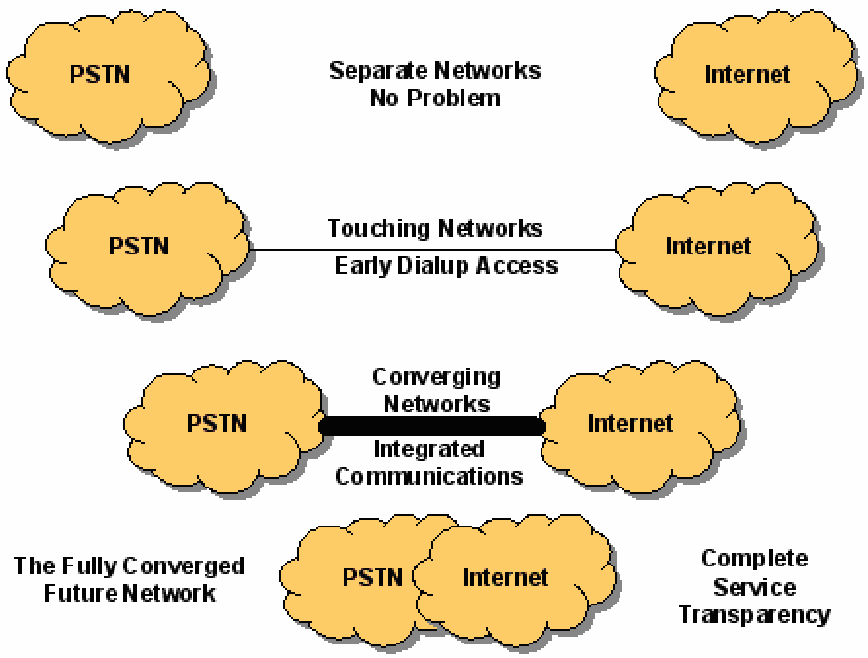

Figure 1.1: Network convergence evolution.

Device Convergence—the Convergence of Desktops, Laptops, Tablets, and PDAs

In business, the way people work has changed dramatically in the past 20 years. The text-based green screen used to work on a mainframe application isn't entirely gone. Now that function is often performed using emulation software on a PC workstation. The PC has evolved as well. Today, businesses and workers customize and optimize their workstations to provide the most effective, productive working environment.

Business has become finely attuned to the cost of real estate. This has driven many businesses to implement telecommuting programs. Reduction in computer size has brought small notebook computers into the workplace. Today, tablet PCs are becoming quite common.

This shrinkage in PC size has extended beyond the computer into the worker's purse of pocket. Personal Digital Assistants (PDAs) became very popular, and today are blending into the smartphone. Mobile phones today have more computational power and memory than earlier PCs.

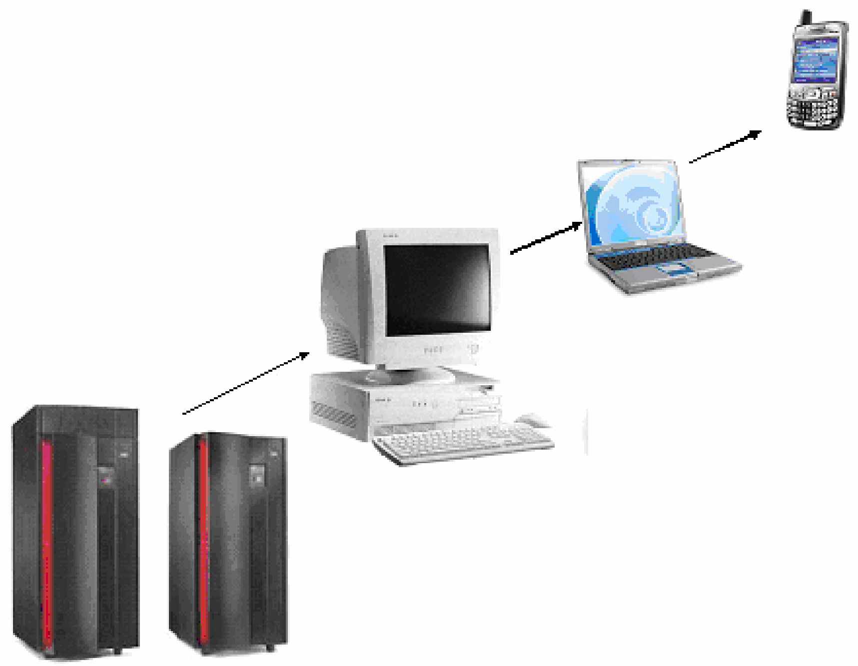

Figure 1.2: Device convergence driven by shrinking desktop "real estate" and increased mobility.

It's also important to recognize convergence on the desktop. Real estate, even desktop real estate, is a costly commodity. Integrating voice services into the desktop workstation, laptop, or even the PDA/smartphone is another facet of convergence.

Application Convergence—Integration of Enterprise Business Applications with Network Services

Perhaps one of the most interesting and potentially productive areas of convergence is the integration of applications and services. The Web no long relies solely on HTML. Today XML-based Web services provide communications between front-end applications and back-end servers. Software development tools such as AJAX and Ruby on Rails as well as plug-in tools such as Flash provide an increasingly rich end user experience in using Web services. Combining these advancements in development tools with enterprise applications provides a new frontier of service convergence that might lead to process re-engineering efforts in workflow.

Major business applications include Customer Relationship Management (CRM) systems that can integrate all aspects of the relationship life cycle between businesses and customers. Enterprise Resource Planning (ERP) systems aid in supply-chain management for manufacturing organizations. They help monitor inventory control, quality assurance processes, and product life cycle management.

Fixed Mobile Convergence

A more recent development in convergence is the idea of embedding WiFi technology into mobile handsets. Business users all carry a mobile telephone. There is a clear trend emerging in the form of a fixed and mobile telephony convergence. The goal is to provide both services on one handset—it is a critical strategic issue across the telecommunications industry for the fixed wireline carriers, the mobile wireless carriers, and IP network providers. When this convergence is accomplished, these operators, regardless of service, will be able to provide voice access to the fixed line infrastructure of the PSTN. Beyond voice conversations, there is a real independence in the end user's device. Location, access technology, and terminal all become irrelevant because they all work.

As Figure 1.3 shows, using FMC features, a phone call might originate via a softphone on the PC in a worker's home. When the worker leaves for the office, the call could be transferred to the mobile handset, perhaps using WiFi over a home wireless network. While driving to the office, the call could be continued over the cellular connection to the PSTN. Upon arriving at the office, this call might automatically be passed from the cellular network onto a corporate WiFi connection to minimize cellular airtime. Finally, the worker might switch the call over to a rich media desktop client in the office to collaborate over an application, shift to video conferencing, or share some other network resources.

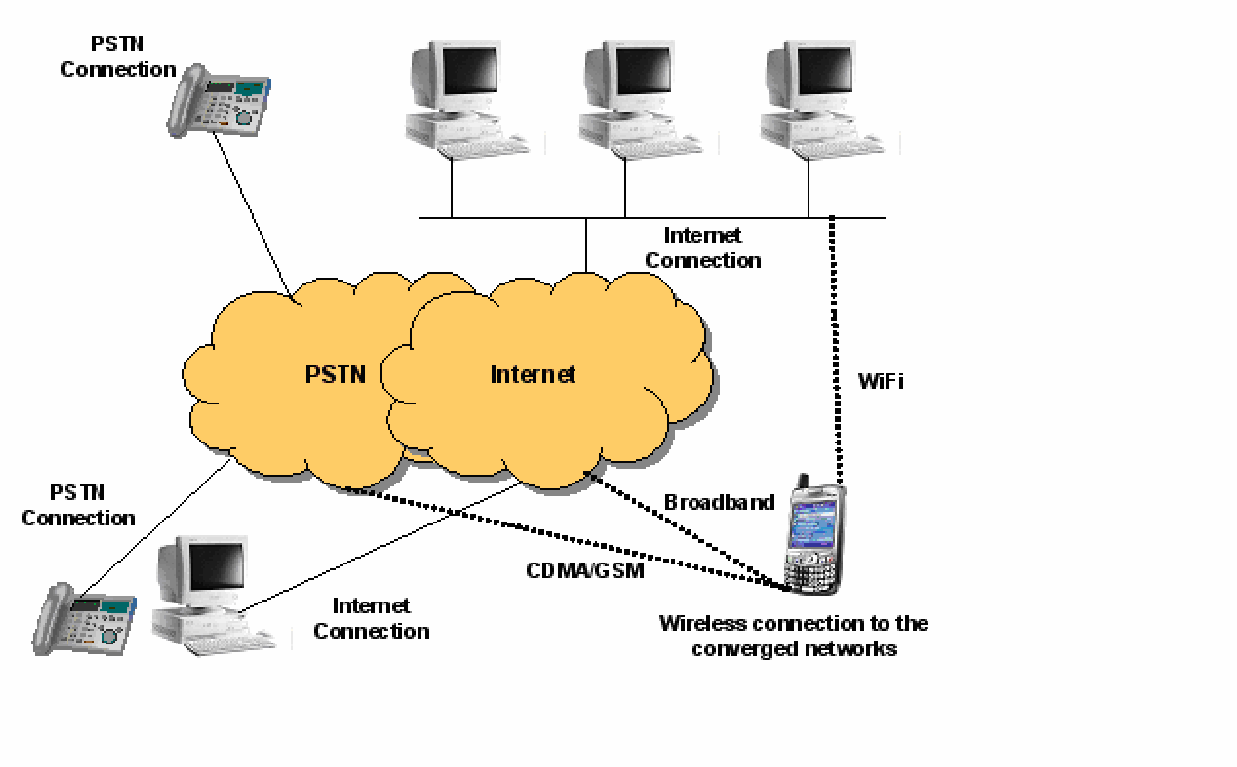

Figure 1.3: Example of FMC.

Key drivers for traffic migration between fixed and mobile networks include the fact that customers gain the ability to choose which network carries voice calls. This choice can be made based on both cost and convenience. Users making calls from home or office locations (where both are available) can select the transport network that suits their needs.

Will increasing dependence cause consumers to use mobile devices at home? Absolutely—business users already do a tremendous volume of mobile calling. The continual downward spiral in per-minute costs can drive consumers to rely more and more on mobile services. The rising availability of broadband data services delivered to these handsets assures the spread of new uses. According to one survey conducted in Britain, the penetration levels for mobile telephony are as high as 70 percent in several developed countries. Mobile telephony may present a viable alternative to traditional fixed telephony for many users.

From the mobile operator's perspective, increased indoor usage of the mobile phone will drive acceptance. Coverage, capacity, and spectrum constraints might limit their QoS. Using convergence of technologies to embrace the unlicensed, short-range wireless protocols today and the longer range WiMax technologies on the horizon, provide an inexpensive means for them to increase capacity and provide service moving ahead. It's in their best interest to actively embrace the convergence of fixed and mobile services.

FMC depends on the handsets to make a wireless connection to an access point and to both make and receive calls via the fixed line infrastructure. Short-range wireless technologies using the unlicensed frequency bands give fixed line service providers the opportunity to create new services without the investment in expensive spectrum licensing or complicated frequency planning.

The two most likely technologies for wireless interface in the 2.4GHz unlicensed band are Bluetooth and WiFi. For FMC, there is a key limiting factor—or success factor: the handset. Compatible handsets are appearing in growing numbers. Bluetooth is available in more than 25 percent of new models. Bluetooth is suitable as a wireless media for FMC and was designed to handle voice service, but it's not the sweet spot. WiFi integration into the handset is quickly becoming a key differentiator for manufacturers. This is a trend that will rapidly expand to the majority of new handsets over the next year or two.

VoIP as a Converged Service

IPT is one of the most visible, talked about technologies in the marketplace today. Improvements in performance and cost reduction issues are huge drivers for enterprise business adoption. There are several factors that make IPT a suitable technology evolution for business telecommunications. Because there are so many different business needs ranging from small companies with home offices to very large enterprises spread around the globe, there is no single "one size fits all" solution for telephony requirements. This should not come as a surprise, because there isn't a single solution for using existing telephony services and equipment.

Voice vs. Data

Voice telephony services require a long holding time to support call durations averaging about 4 minutes per phone call. The signal is sensitive to delay and jitter because it is a real-time interactive communication. In the PSTN, you deliver this service using circuit switched technology. In some cases, large businesses have implemented private telephone links, such as T-1 tie lines, between branch offices to provide internal telecommunications from site to site.

Data communications don't have long holding times or durations. They're described as bursty and unpredictable in nature. The durations of data transfers may be quite short, especially with increases in available bandwidth. Data communications are frequently not a real-time, two-way, interactive session like a phone call. Email and Web browsing, for example, are not as time-sensitive as voice. Data services have generally used all the available bandwidth for a very short duration. That's beginning to change as QoS becomes an integral part of network operations.

In the past, it wasn't uncommon for enterprise businesses to have two or three completely separate networks for conducting business. Voice traffic was often handled through a PBX connected to the PSTN. Real-time interactive data was transmitted via some type of packet network, either IP or Frame Relay in many cases. Large file transfer between mainframe systems or server farms might have been carried out across a dedicated point-to-point connection.

The Cost of Doing Business Drives VoIP

One of the major early drivers behind convergence and VoIP was cost, the job of calculating the true total cost of separate voice and data networks is a complex process. The network infrastructure cost is one factor. Telephone services have historically been billed based on minutes of use. A phone call requires that a circuit be established for the duration of the call, so this billing system made sense in the past. Data networks are different, and most commonly bill for either the bandwidth provided or some guaranteed carrying capacity (referred to as Committed Information Rate—CIR).

Equipment cost can usually be treated as one-time capitol expenditures (CAPEX).This cost is tied to buying required equipment, routers, computers, telephone systems, wiring and cabling, and so forth. CAPEX costs are closely scrutinized when businesses invest in new network solutions, but they only represent one facet of the total cost of ownership (TCO).

Operational expense (OPEX) is another factor that is too easily underestimated. Given that the topic of this guide is converged network management, it will be digging into a variety of holistic management issues and techniques. The labor effort required to provide operational support for additions, moves, and changes to the network are difficult for many organizations to quantify. Reorganization is for many enterprises, business as usual, with constant upheaval. The costs to support this continual churn over the lifetime of the network often far outweigh the one-time equipment costs. For the enterprise managing multiple networks, telephone and voice, this OPEX may be duplicated for each network. It is very common for an organization to have a support person or staff for the telephone system and another for the data network. In the past, these support personnel required different skill sets, but convergence, even in the state achievable today, can change that.

Administrative costs comprise still another facet of networking that may be underestimated in some enterprises. There are the basic costs of processing monthly invoices for payment. Beyond simple accounts payable, there is administrative effort involved in the ongoing monitoring of network performance compared with monthly billing. Billed minutes of use in the voice network and committed information rate delivered have to be compared with Service Level Agreements (SLAs) and validated against traffic reports. The same staff that manages day-to-day operations might perform this analysis. Although often overlooked and perhaps not high in terms of actual cost, this function can potentially yield high return, particularly in networks that are volatile and changing frequently. Holding providers accountable for meeting the terms of their contracts is a labor-intensive but crucial component of both voice and data network services.

The telephone and data network(s) in an enterprise need holistic treatment, like a living organism. The health and well-being of these vital business resources must be constantly monitored. Enterprise business constantly evaluates and assesses employee performance. Network performance must be proactively managed in a similar fashion. That means vigilant monitoring throughout the life cycle of the network to ensure maximum performance.

Holistic management of the business network includes working closely with voice and data service providers. It’s far too easy for changes in a dynamic enterprise to happen without any correlation to network services. The result is that the network reality diverges from the requirements. They move in different directions driven by changes in the business.

> I’ve helped many clients conduct network analysis and assessment focused on identifying potential OPEX reduction. In one such case recently, we quickly discovered that the company had downsized operations by half in the preceding year. What was troubling was that the network billing didn’t reflect any reduction. The circuit cost component of their OPEX hadn’t changed. When a remote branch office closed, this company had completely failed to have the network provider disconnect circuits and stop billing. Several thousand dollars were paid to a provider over the course of a year for idle circuits that carried no traffic. Although this might seem a laughable example of really poor management, that isn’t the case. This type of problem is far too common.

Given the technologies generally available, is it practical to build one single network infrastructure that will support all traffic and service types required to do business? A few years ago, the answer to that question was probably no. Four or five years ago, only the earliest adopters of leading-edge technologies were able to take advantage of integrating multiple services into a single environment. Today, networks are actively converging. Right now, for many companies, unifying communications onto a single integrated infrastructure, converging services into single workstations, or integrating applications and services makes sound business sense. As technologies collide, keep in mind that what didn’t work yesterday may be viable today, and what doesn’t work today, could well be the de facto standard of tomorrow.

Technical Definition vs. Market Definition

It's perhaps an important distinction to identify the differences between VoIP and IPT and what they might mean in the marketplace. From a purist perspective, VoIP is simply the process of digitizing and packetizing voice. Figure 4.4 shows a common arrangement for making a VoIP call. Although IP telephone calls can take on many forms, this example represents an IP phone call in a form that doesn't directly involve the subscriber in any way. The telephone set in this example is a traditional telephone connected to a traditional PBX.

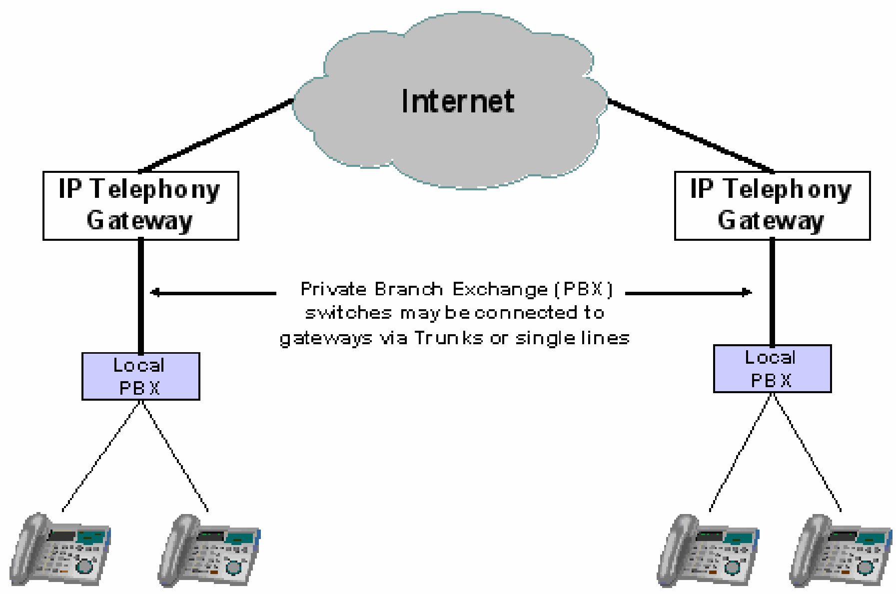

Figure 4.4: A variation on IPT—the gateway between business and the PSTN.

The phone call begins when the subscriber lifts the receiver and a dial tone is supplied from the local PBX. In this case, when the caller keys in the telephone number, that information is forwarded to an IPT gateway. The gateway provides a conversion point for traditional voice traffic to be converted to IP and vice versa.

The originating gateway has to convert standard PSTN signals and correlate the dialed telephone number to the IP address of the terminating gateway that serves the called party. This signaling information is packetized and sent to the IPT gateway at the far end. The receiving gateway must decode the IP packets and convert the signals back into traditional voice format for the PBX on the remote end. When the called party answers the telephone, a complete two-way conversation can occur.

Some benefits to this approach are obvious. Others are more subtle. This method of VoIP implementation has been employed by both corporate enterprises and commercial VoIP service providers (also called an Internet Telephony Service Provider—ITSP):

- Using a packet network to transmit allows the sharing of resources rather than the dedicated resources for voice and separate resources for data. Although this visual shows the Internet, many enterprises use their internal wide area packet network to deliver this service.
- If the Internet is the packet network used, the service provider can now provide two distinctly different but necessary services from one consolidated backbone infrastructure. This approach may drive the need for QoS in the packet network.
- Because the ISP or ITSP is not a regulated telephone company, the requirement for payment of access charges doesn't exist. Thus, the ITSP can deliver calls cheaper. However, it also means that the local exchange company doesn't receive remuneration while its circuits are being used in many cases. This is good for one part of the industry, but bad for another part. Telephony traffic is shifting off the PSTN and on to the Internet.
- In this particular implementation of IPT, the end user doesn't have to convert to a VoIP phone. The end user doesn't have to do anything. They might not even know the telephone call is being carried over a VoIP service. The Internet is a large and growing network, but the user experience for a voice call may best be served using a telephone set. There is no added complexity of soft phones or specialized instruments for end users in this model. This could prove a viable approach for common calling services such as pay phones, hotel phones, and public phones where other, more advanced services aren't required.
- The initial startup cost, or barrier to entry, for a VoIP service provider is very low in comparison with the startup costs for a new telephone company using traditional PSTN technologies. IP packet switching technologies provide for greater efficiencies at a lower cost.

Variations on implementing the converged services network range from straightforward to very complex. When evaluating how services converge within an enterprise, it's vital to do so with an eye to not just the business needs but also the ease of support and network management. Some other variations on implementations might be:

- IP Centrex services from a managed service provider—This service could be delivered to the corporate network over one single network connection, with voice and data services broken out inside the company network.
- A managed IP-PBX inside the corporate network—The key difference between this approach and that shown in Figure 4.4 is the potential for VoIP telephones or integrated softphones within the corporate network.
- Internet users might utilize some form of commercial gateway—Notable early examples of this were http://www.dialpad.com and http://www.net2phone.com. You can expect to see an increase in Web services providing this sort of service for Web only, or thin client, connections. In particular, as developers find new ways to embed VoIP services within the browser, you can expect to see more of this functionality.
- Some companies are using PC-based software to provide internal telephone calling and collaboration services between employees on the corporate network—This LAN-based VoIP is efficient within the network, and essentially free of charges. This may be some of the early use you see for the new Microsoft Live Communications Server. Calls to the PSTN might require employees to have a traditional telephone using the traditional network.
- A large multi-location enterprise deploying unified communications might implement multiple gateways at geographically dispersed office sites. Calls might be directed to the nearest gateway to the called party, minimizing toll and long-distance charges. This "least cost routing" approach has long been used in large enterprise voice networks.

VoIP or IPT?

Voice digitization is not a new technology. Digitization began inside the telcos in the 1960s utilizing what is called T-1 service today. During this time, T-Carrier, as it was called then, was deployed to provide trunking capacity between telco central offices. Analog transmission technologies were in widespread use at the time, and digital technologies enabled network performance improvements in the PSTN that benefited both customers and the phone companies. The following section explores the basics of a telephone call to set the stage for a discussion of network requirements and call quality issues in later chapters.

A Telephone Call Simplified

When a person makes a telephone call, there are several basic steps that must occur whether the call is a videoconference, a fax transmission, or a voice conversation. These tasks are performed by the service network in each case.

When you speak, your vocal chords vibrate, generating sound waves. These sound waves are an analog signal. These sound waves travel through the air, and your ears convert them back into signals your brain can understand. The world around you is very much an analog place.

The telephone converts the sound waves from your spoken conversation into electrical signals that can be transmitted over copper wires. The telephone set at the receiving end converts that electrical signal back into analog sounds waves you can hear.

In the telephone network, that electrical signal is converted from analog into a digital signal, a stream of zeros and ones defined by changes in the electrical state of the circuit. Thus, to carry on a telephone conversation, your analog voice must be changed from sound waves made up of vibration of air molecules into an electrical representation of the analog wave. This analog wave then must be digitized and transmitted across the network. At the other end, the signal is converted back to analog, then back into sound waves at the earpiece of the telephone handset.

Converting the Analog Signal to Digital

The telephone network today is made of digital central office switches connected by digital trunk circuits. It makes sense to transmit digital signals. The local loops (also called the *last mile)* that connect subscribers and most phones are analog. Analog-to-digital conversion must be performed somewhere in the network. This conversion is typically accomplished using a technique known as pulse code modulation (PCM). In order to perform PCM, a coder and decoder (more commonly called a *codec*) are needed. PCM takes samples of the voice conversation 8000 times per second. These samples are then converted into an 8-bit word, resulting in a 64Kbps sample (8 bits × 8000 samples per second = 64kbps). Each of these 8 bit samples can be coded into one of 255 different possible combinations. The 8 bits of binary data can represent values from 0 to 255, but all zeroes cannot be used in this coding scheme.

This guide will touch on the 64Kbps line rate more than once. This line speed is what a standard voice channel in the PSTN provides. In the United States, a common time-division multiplexing transmission scheme is used to transmit 24 voice channels as a digital stream of data over a single circuit. This is the basic voice service provided over a T-1 circuit today, with 24 voice channels delivered over 1.544Mpbs (due to some overhead). This design of digital facilities and voice circuits is part of the Synchronous Digital Hierarchy throughout the world. Other parts of the world don't use T-1 circuits, but their approach is similar.

By today's standards, 64Kbps really doesn't seem like a lot of bandwidth, but the technical reality is that, depending on which codec is used, it's far more than necessary to carry a voice conversation.

> 📖 A future chapter will touch on coding schemes used to sample and compress audio traffic streams.

Network economy of scale and simple economics drive the industry to work toward reducing the bandwidth required for a phone call. The less bandwidth required per conversation, the more conversations the network can carry at one time. Adding carrying capacity through codecs and compression minimized capital investment in network equipment. This can potentially drive down the cost of carrying a phone call.

When Does VoIP Become IPT?

The PSTN is a complex and mature implementation of technology that has evolved over more than a hundred years of use. Though technologies are changing and advancing quickly, the Internet is quite immature by comparison.

The PSTN has what is referred to as the Advanced Intelligent Network (AIN). 800 number services provide access to databases of caller information. Caller ID, call waiting, call hold, and conference calling are common features available to almost all users. E-911 services are constantly evolving with technology in order to identify the caller's location to within a very small geographic radius. This mature network has been optimized over time to provide the best quality voice and a variety of services that are now taken for granted. The PSTN does far more than transport voice traffic. It provides a comprehensive and robust suite of telephony services.

The IP networks, or Internet, certainly can't provide that rich set of services by itself. VoIP has been achievable for a number of years. Simply packetizing a voice signal is easy. There is far more to IPT than just carrying voice traffic in IP packets. The full, rich telephony feature set has to be incorporated before users could give serious consideration to IPT as a viable service in the production network for enterprise business.

IPT solutions can now bridge the gaps between the PSTN and the IP network. Using the TCP/IP protocol suite, a network such as the Internet can now provide many aspects of the traditional telephone network. The IPT market has seen a substantial growth rate over the past several years. Vendor products have improved, but so has basic network technology.

The issue of VoIP vs. IPT may be one of semantics, but it represents a problem for business people trying to make a decision. One vendor will talk about VoIP, another about IPT, and a third will now refer to unified communications. All the scenarios described in this guide are IPT in some form.

It's clear that if the end user connects to the PSTN, they'll require some kind of a telephone. If they connect to the IP network, they'll need an IP device of some kind. As business managers, it's important not to get hung up in the semantics vendors or service providers use. It's prudent to focus on the services being provided to ensure they support business needs.

Intended for PC-to-PC Voice Messaging

Pure IPT between computer users over the Internet was a popular, early hobbyist's communication technique. This form of VoIP has been accomplished by users meeting in a chat room. Today, it's performed by a staggering array of software solutions including voice-oriented solutions such as Skype, Gizmo Project, and GoogleTalk. There is also a whole family of instant messaging (IM) programs including ICQ, AOL Instant Messenger (AIM), MSN Live Messenger, Yahoo, and others. IM users can easily activate a voice session over the Internet in most popular client solutions today. Many of these VoIP solutions that began as PC-to-PC calling tools now have links in and out from the PSTN as well.

Early prophecies for success in IPT viewed the technology as a tool for consumers to combat the high price of long-distance telephone service. Those arguments seemed to hold water at that point in time, but they overlooked the technical advances and growth of mobile telephony. Today, many wireless providers offer nationwide calling at rates cheaper than imaginable 10 years ago. VoIP still aids in reducing the cost of international long distance, but the current primary driver is more likely to be service integration than cost reduction.

Never Intended to Replace Global Telephony

VoIP wasn't developed with the ultimate goal of replacing the PSTN—not initially. But VoIP disruption has occurred and is still gaining momentum. Delivery of voice service is no longer tied to the vertically integrated Class-5 telco switch. This disruption has been accepted as the trend to the future of telecommunications.

IP evolution is replacing monolithic, proprietary solutions and broadening into Web and application services. Innovation is increasing at a faster rate in a dynamic and highly competitive environment.

"Free Voice" Was Compelling and Simple

The idea of free voice calls spurred hobbyists and early adopters to experiment with an array of new ideas. The drawback is that this PC-centric group of innovators and early adopters provides only a limited market. They represent an application and integration proving ground, but not a mass market.

Pushes the Edge Over Convergence

What will be the accelerant for VoIP and unified communications? The ongoing debate between cost reduction and enhanced utility will continue into the foreseeable future. Today, the market sees far too many "more of the same" or "me too" solutions. This limits the advance of both user experience and penetration. The VoIP developer community has often been providing equivalent features to the telco environment. VoIP as a replacement for PSTN dial tone is neither useful nor well received in today's market.

If applications will fuel the future growth and demand, one option might be to pursue integration inside what have been termed the *walled gardens*. An integration layer between proprietary applications and content providers will expand the reach of converged services to a broader market.

Vertical communities of interest are early adopters, but early adopters really only provide fuel for developers. Markets emerge over time through an established pipeline of applications and services. You never know at the beginning of the development cycle what the "killer app" will be. Communities are the early adopters and innovators that help create the future. A rich multi-media, real-time telephony experience exists within the technology, but it's just taking root and beginning to expand. One thing is clear: Voice will no longer be driven by the telephone. Voice services of the future will be driven by the context of the business application.

Multiple Play

Multi-play is a popular marketing term describing the delivery of multiple communications services by service providers that traditionally only offered one or two of those services. Triple play and quadruple play are commonly used to describe the combined delivery of high-speed Internet, television, telephone, and mobile phone services, respectively:

- Dual play—The dual-play service is a marketing term for the provisioning of the two services: high-speed Internet and telephone service over a single broadband connection. This has most frequently been used with cable providers.
- Triple play—Three-way convergence is inextricably linked to the underlying communication infrastructure. A prime example of this would be packaging communication services in a form so that customers can purchase television, Internet, and telephony in a single, bundled service subscription.
- Quadruple play—So far, the quadruple play service remains elusive. It's the triple play service of broadband Internet access, television, and telephone with the addition of wireless service provisions. Advancements in WiMax and other leading-edge technologies are rapidly improving. Transmission over a wireless connection link at combinations of speeds, distances, and non-line-of- sight conditions may soon make it possible to never connect to voice or data services by a wire to anything, even while at home.

New Devices

New devices are driven by technology advances, user expectations, and how people use their VoIP and mobile phones. There is some relevance evolving in the market surrounding how often people change phones. Many consumers move to the next new smartphone every 6 months in the mobile environment, but in business, many people have been using the same telephone set with the same features for many years.

You get new cell phones regularly because they enhance your productivity in some way. This leads providers to think about how people interact with their phone set. The comparison of how you change cell phones sets the stage for other unified communications options.

Although vendors continue to see rising shipments of dedicated IP phones quarterly, there are other business drivers to consider. Three words describe this new driver—relevance, context, and presence. The type of telephone set you need varies depending on your context. In some cases, you'll be in a meeting or unable to talk and an instant message might be the most relevant communication method. At other times, you can't be interrupted at all. Whether it's a PC-based softphone, a traditional desktop telephone, or a mobile handset, each offers strengths depending on the context being used.

Today, people work in distributed, virtual, mobile teams. The work day doesn't start when you enter the office, and it doesn't end when you leave. You need to manage your communications flow throughout the day. There are hundreds of features and services in existing systems, but users generally don't know how to access them. Most people use a very limited subset of the available features and functionality. Features will become more intuitive. The context of your work day and how you're using other business applications will be more central to the development of converged communications devices as things evolve.

The key questions for device manufactures are:

- What do users expect from their phone? Users interact with many different devices all day long. Those interactions drive expectations.
- How can we add efficiency via the telephone device?
- How can phones increase productivity?

Phones are moving from being feature focused to user focused. It's a paradigm shift on the desk and in your purse or pocket. Can a phone really improve or impact efficiency? Yes, but if it fails to provide access to features and functions, it can be a frustration and lead to lost productivity.

VoIP Can Include Advanced Applications

The convergence of real-time media carried over IP along with data is where we've arrived. That is the state of technology today. Now we're looking at how to converge these services onto application engines. Convergence with enterprise business applications presents another whole convergence layer. CRM, human resources (HR), and ERP systems all introduce interesting ideas when blended with telephony services.

Converged communications is the capability to integrate at different layers than just telecommunications. These other layers are still fairly separated into silos, or vertical groups. Workers today have multiple communications devices and soft phones. New capabilities such as real-time directories that provide presence and availability information, conference call setup, and rich media integration empower the end user.

Convergence integrates the power of multiple platforms that offer service capability in a unified environment rather than discrete applications or services. The Services Oriented Architecture (SOA) is moving toward opening up the services of one application for sharing with another application altogether. This isn't just VoIP. It isn't just integrating communications. It's service integration through software across multiple business applications. Telephony software in the converged environment can now act like a business application and interact with other business applications.

Many of the functions that the hardware enterprise PBX provided can now be performed in software. Integration at the software level, at the application level, is the key to opening up integrated services within IP. You can do new things when you integrate business applications rather than focus on technology.

Converged Communications Leads to Converged Applications

Specialized applications are becoming standard applications. Using the SOA information resources available to all participants (users, services, and applications) in the network as independent services may now be accessed in a standardized, converged way. Users don't need to change context from one application to another. The end user shouldn't have to shift the context of the core application in which he or she is working to make a telephone call. The developer communities are working to open up integration services to allow all that context switching to happen behind the scenes. The following list highlights real application examples:

- A Web browser-based interface is the simplest example. It can display state information to users. The browser can easily show present state in an address book or telephone directory, with click-to-dial capability.
- A mobile device that can run thin client or browser-based applications presents another example. This device makes it easy to authenticate as a mobile WiFi user and then display state information based on that IP connectivity. The mobile device simply becomes a thin client using Web services on the network. Again, directories and presence state are prime examples of integrating productivity tools.
- For large enterprises, think about the whole cumbersome process of voice and data adds, moves, and changes for new employees and people shuffling from cube to cube. That's historically a huge labor effort. Why not enable an HR system that interacts with the voice and data services network to automatically configure telephony services for new employees?

Converging business applications is absolutely where the entire business community is headed.

FMC

Mobility is a huge issue, and the advances ahead with FMC make this a subject of interest across many business sectors. The environment is ripe for mobile VoIP. The delivery of enterprise applications, information, and services to a mobile worker is vital today. One factor that has made mobile VoIP viable today is near-universal standardization on Session Initiation Protocol (SIP).

At a recent developer conference, informal polling of attendees pointed out that email usage on mobile phones was at about a 50 percent usage rate. We are becoming mobile data users more and more. Enterprise users are a huge piece of the broad mobile services market and have been since 1973. Although mobile usage is on the rise and ripe for convergence of VoIP and mobile services, user interactions and experiences are distracting and degrade from the overall the user experience.

Radio characteristics present another challenge. Current WiFi technology is very chatty in standby mode, draining mobile device batteries and shortening usable lifetime. Because of this, dual-mode devices tend to suffer from measurably greatly shortened battery life. There is industry work needed to make the WiFi device more effective as a phone.

A recent survey of mobile professionals produced some interesting results. The most active mobile users in business are either customer-facing workers or management. More than 75 percent were happy with coverage. About half felt that dual mode could improve coverage for better service. Churn in the mobile industry is driven by coverage issues more than any other factor. Most mobile carriers are now down to about a 2 percent churn rate.

Business today is very mobile. 70 million Americans use their cell phones for work. 70 percent of all cell calls start in WiFi-enabled areas (home, office, hotel, and so on). By 2009, the industry expects that 25 percent of new mobiles will be smartphones. One really interesting note from the survey about business and mobility: 67 percent of mobile professionals receive or make more than 25 percent of their business calls on mobile phones. Mobiles are very important devices to enterprise business. (These survey numbers come from a collection of surveys conducted by FirstHand, Nortel, RHK, and Gartner.) Mobile VoIP is much more than just terminating a SIP session on a small handheld device. It's about enabling a productive personal or business experience.

Summary

You can wait for evolution or you can begin using the technologies that are available today and ride the wave toward the fully converged networks of tomorrow. Even as developers are defining the architecture, boundaries, requirements, and interfaces for a shared unified communications network of tomorrow, the current solutions provide a wide open space to grow and evolve existing business services.

The chapters ahead will review key considerations for integration, business drivers that make sense, and productivity advantages. This guide will dig further into the key success factors for deployment and management of converged networks. It will investigate event correlation across multi-service networks, and dig into availability management and capacity planning in the world of unified communications. This guide will explore fault, configuration, performance, and security management in this exciting new world. Finally, it will take a look at asset and compliance management.

The next chapter will hone in on business processes most often impacted by convergence. In addition, it will examine the issue of call quality and how to deliver total quality network services, including VoIP in the converged network.