Implement Voice over IP and start saving money today!

XO®

XO Communications® VoIP FOR DUMMIES®

With case studies on how VoIP improves productivity

A Reference for the Rest of Us!®

FREE eTips at dummies.com®

Timothy V. Kelly
Executive Director, National Center for Converging Technology Research

XO: The Proven Business VoIP Provider

What is VoIP?

Does it really work?

Do calls get dropped?

Is VoIP calling really free?

Before you bet ***your*** business communications on VoIP, you need the answers to these and other critical questions. Who can answer those questions?

We're XO Communications and we sponsored this special edition of *VoIP For Dummies* to help answer some of those questions. It can help explain the "what" and the "how" of VoIP. When you're ready to ask "who?" we hope you'll turn to XO Communications.

XO Communications has been providing communications services exclusively to businesses for nearly ten years. Our services range from local to long distance, Internet access to private line, Ethernet services, VPN, web hosting, and more. Our XOptions and XOptions Flex bundles provide a complete flat-rate solution of services for businesses. And, while new VoIP providers are appearing every day, XO is the proven business VoIP provider who:

- **Has carried VoIP traffic since 2000**
- **Now carries nearly 10 billion VoIP minutes per year**
- **Serves businesses exclusively, with nearly 200,000 business customers**
- **Sold over 3,000 XOptions Flex VoIP bundles in less than one year after introduction**
- **Owns and operates our own award-winning OC192 IP network backbone**
- **Has deployed Sonus softswitches for state-of-the-art IP telephony**
- **Carries VoIP traffic for many prominent and emerging VoIP carriers**

XO serves over 70 major metropolitan markets nationwide, and offers 24 x 7 customer care.

So, when it's time for you consider the "who" behind your VoIP solution, contact XO Communications.

www.xo.com/LearnVoIP

XO Communications® VoIP FOR DUMMIES®

by Timothy V. Kelly

Wiley Publishing, Inc.

XO Communications® VoIP For Dummies®

Published by
Wiley Publishing, Inc.
111 River Street
Hoboken, NJ 07030-5774
www.wiley.com

Published by Wiley Publishing, Inc., Indianapolis, Indiana

ISBN-13: 978-0-470-08640-7

ISBN-10: 0-470-08640-8

Manufactured in the United States of America

10 9 8 7 6 5 4 3 2 1

1O/SW/QW/QW/IN

Publisher's Acknowledgments

We're proud of this book; please send us your comments through our online registration form located at www.dummies.com/register/. For details on how to create a For Dummies book for your company or organization, please contact dummiesrights&licenses@wiley.com.

Some of the people who helped bring this book to market include the following:

Acquisitions, Editorial, and Media Development

Project Editor: Jan Withers

Editorial Manager: Rev Mengle

Business Development Representative: Karen Hattan

Composition Services

Project Coordinator: Kristie Rees

Layout and Graphics: Heather Ryan, Alicia South

Proofreaders: Laura Albert, Dwight Ramsey

Special Help: Dave Dobel, Martin Weiss

Publishing and Editorial for Technology Dummies

Richard Swadley, Vice President and Executive Group Publisher

Andy Cummings, Vice President and Publisher

Mary Bednarek, Executive Acquisitions Director

Mary C. Corder, Editorial Director

Publishing for Consumer Dummies

Diane Graves Steele, Vice President and Publisher

Joyce Pepple, Acquisitions Director

Composition Services

Gerry Fahey, Vice President of Production Services

Debbie Stailey, Director of Composition Services

Table of Contents

Introduction

VoIP (pronounced *voyp*) is the name of a new communications technology that changes the meaning of the phrase *telephone call.* VoIP stands for *Voice over Internet Protocol,* and it means "voice transmitted over a computer network." VoIP technology enables you to merge your current telephone service into your data network, thereby reducing costs, simplifying infrastructure, and minimizing maintenance efforts. VoIP gives you greater control, flexibility, and features that can make your employees more productive.

Internet Protocol (IP) networking is supported by all sorts of networks — corporate, private, public, cable, and even wireless networks. Don't be fooled by the "Internet" part of the acronym. VoIP will run over any type of network. Currently, in the corporate sector, the private dedicated network option is the preferred type. For the telecommuter or home-user, the hands-down favorite is broadband.

Traditional telecommunications and Internet Service Providers are racing to provide businesses and residential customers with VoIP services which, because they are so efficient, are rapidly replacing the traditional services you are used to buying. In a word, telephony is going digital, and digital means getting your infrastructure IP-enabled.

As you can imagine, VoIP is a win-win for everyone. The added flexibility and quicker response times translate into greater customer satisfaction and increased productivity throughout your organization.

About This Book

If you are a manager or owner of a small or medium-sized business who needs to decide whether to make the switch to VoIP, or if you are an IT person looking to help your boss make an informed decision about integrated networking, this book

provides an excellent place for you to begin. It's also an excellent starting place for end-users who are new to VoIP.

With this guide, you can find out how VoIP works and how it sizes up to telecommunications technology that was previously considered to be irreplaceable. By the time you finish Part V, you'll see why many businesses throughout the world have turned to VoIP and integrated networking as their main system for data, voice, and video transfer.

You may read this book from cover to cover, which is what I recommend, seeing as it's a pretty fast read. If you are in a hurry, however, feel free to dip into whatever part or section best suits your needs and return to the rest of the book when you have more time to enjoy the read.

Recognizing the Role of XO Communications in VoIP Convergence

As your business experiences *convergence* — in which voice, data, and Internet services become integrated into a single service — you need to make critical decisions that will impact the cost, efficiency, and productivity of your business. VoIP services come in a wide range of options, so you want to work with a proven communications partner that understands both VoIP technology and your business needs.

XO Communications is a recognized leader in VoIP services. XO has built an IP-based, nationwide, fiber-optic network to meet the communications needs of businesses like yours and carried over 10 billion minutes on its network last year.

Convergence has already started. Service providers are investing in the next generation of VoIP services. You can expect new telecommunications features and services, some not possible today, to be introduced because of the capabilities of an IP-based service. Even today, the benefits are compelling enough that many companies are making the move to VoIP services now in order to gain productivity and cost savings.

How This Book Is Organized

Each part of this book focuses on a different aspect of VoIP, as described in the following sections. As I mention in the preceding section, you may read the book cover to cover, or skip around to find the information you need when you need it.

Part I: Getting Down to Business with VoIP

Part I introduces you to the basics of VoIP. You get the rundown on essential terms and the general workings of the technology. This part describes some of the cooler things you can do with VoIP.

Also included is a case study that demonstrates some real-world applications of VoIP.

Part II: Simplified Management with VoIP

In Part II, you discover how switching to VoIP can reduce your operating costs — and the effect is immediate. You save the cost of leasing lines and regulatory fees, and calling features are actually free.

To help set VoIP in context, Part II compares its features and cost structure to the four main non-VoIP telephony models. I'll let Part II do the talking, but I think you'll agree after reading it that VoIP is the most cost-effective choice.

All this information is fantastic, but how do you sell it to your boss? Or, if you are the boss, how do you make the best informed decision possible? Look no further. This part gives you all the information you need, including the new "Future Four" ways to make VoIP work for your company.

Part III: Three Phases to VoIP Migration

Part III outlines the migration path recommended by XO, an industry-leading vendor of converging technology services. You can also turn to this part to find out how VoIP improves your network's overall quality of service in the course of the transition. You can also follow the highlights of a case study of one company that developed a hugely successful VoIP migration strategy.

Part IV: Ten Reasons to Switch to VoIP Now

The reasons to switch to VoIP are countless, depending on how far you want to project the future of the marketplace. Part IV describes the ten best reasons to make the switch. It covers everything from projections for the future of the telephony industry to the flexibility that VoIP features provide your organization.

Part V: Ten VoIP Myths

This part dispels the top ten myths about VoIP. Get the truth about VoIP here and now.

Case Study

This extended case study illustrates how XO Communications adapts its services to make VoIP work for the unique requirements of a multi-location, midsize company. Flip here if you want to begin with a real-world example of the benefits of using VoIP.

Appendix: Glossary

Confused by a term you encounter while reading this book? Turn to the glossary and your bewilderment will fade into the past. (It's also a great tool for understanding VoIP marketing brochures and white papers.)

Icons Used in This Book

Throughout this book, I occasionally use icons to call attention to material worth noting in a special way. Here is a list of the icons along with a description of each:

If you see a tip icon, perk up — you're about to find out how to save some aggravation.

This icon indicates technical information that is probably most interesting to IT professionals.

Some points bear repeating, and others bear remembering. When you see this icon, take special note of what you're about to read.

Where to Go from Here

The most important thing to keep in mind whenever you are exploring a new technology is how it fits into the larger picture. Take a global view. Specifically, always be thinking, "How will this feature increase my company's efficiency?" Or, "How will an integrated network help promote collaboration across my company?" Of course, you may also be wondering how you'll save money with VoIP.

Consider the direction of the telephony industry. The move toward VoIP is happening, right here, right now. If you are a decision-maker in your company, you need to plan for how you will remain competitive in this changing market. If you're an IT professional, you need to research the available technologies so that you can make recommendations to your boss and implement, if necessary, a VoIP system. End-users need to be prepared to make the switch if their company adopts a VoIP system, or if they get transferred to a new location that already has such a system in place.

This book is a great first step to understanding the basics of VoIP technology. My best advice? Flip the page, and keep on reading!

Part I

Getting Down to Business with VoIP

In This Part

- How IP Telephony and VoIP work
- IP-based features
- Getting to know your IP phone

Technological innovation is hurling itself upon us once again. This time it is coming in the form of improving the way we make telephone (voice) calls. It brings with it several new capabilities that really change the meaning of the phrase *telephone call.* VoIP is the name of this new communications technology. *VoIP* stands for *Voice over Internet Protocol.* Basically, VoIP means "voice transmitted over a digital network."

VoIP is often referred to as *IP Telephony* because it uses the latest innovations with the popular and familiar IP protocols to enable enhanced voice communications throughout the enterprise or small-to-medium-sized business. IP networking supports corporate, private, public, cable, and even wireless networks. IP Telephony unites an organization's many locations — including mobile workers — into a single, converged communications network.

And by the way, don't let the "voice" part of the acronym fool you — IP Telephony calls go above and beyond the call of duty. When it comes to placing telephone calls, IP Telephony provides a range of support services and features unequalled in the world of telephony.

What Is VoIP?

VoIP, or *Voice over Internet Protocol,* means basically what the acronym states: Voice travels over the Internet. When VoIP was first developed, it worked only with the Internet and nothing but the Internet so help us all. Today, VoIP operates over most distributed network types, including those used throughout the corporate sector. But the "I" has stuck with the acronym. The "P" represents the term *Protocol*. Protocol refers to the type of rules that the network uses to send and receive signals. These signals are the high and low electrical or optical pulses often represented by the more familiar 1s and 0s of digital networking.

IP Telephony works by converting voice communications into data packets. Conveniently, it runs on the popular Ethernet LAN (local area network) technology, which currently supports over 96 percent of all company-operated LANs.

The old school: Circuit-switched telephony

Before digital networking took off, everyone had to use the one and only Plain Old Telephone Services (POTS). POTS runs over a network called the Public Switched Telephone Network (PSTN). The PSTN has been around since Mr. Bell invented the telephone. That is why most companies today have POTS-related systems in place. These POTS telephone systems use the old tried-and-true method of telephone service known as *circuit-switched.*

Believe it or not, a good illustration of POTS and PSTN is the experiment where your fifth-grade teacher had you take two tin cans and a length of wire to create an archaic telephone system. As strange as it seems, this antiquated method of telephony is the principal means underlying the operation of POTS operating over the PSTN.

What changes in the real POTS-based telephony system is the number, length, diameter, and type of wire or cables used. These elements have grown immensely in variety and type.

In addition, the types of telephone equipment have changed dramatically both at the customer end and at the carrier provider end. But POTS telephony continues to use "circuit-switched" rules (or protocols) of operation.

The new school: Packet-switched telephony

Unlike circuit-switched POTS, which always requires use of the Public Switched Telephone Network (PSTN), VoIP technology has enabled telephony and other new and novel features and services to run over dedicated and wireless networks including even your computer network. These newer network types use packet-switched protocols.

Packet-switched VoIP puts voice signals into packets. Along with the voice signals, VoIP packets include both the sender's and receiver's network addresses. VoIP packets can traverse any VoIP-compatible network. Along the way, they can choose alternate paths because the destination address is included in the packet. The routing of the packets is not dependent on any particular network route.

In a circuit-switched network, the destination address is not included in the signals; routing directions are determined physically by the actual POTS line. So the routing must follow a specific network line just as a train follows a designated set of railroad tracks. If the line is down, the call cannot go through.

In a packet-switched distributed network, if one of the network lines supporting the enterprise network is down, the packet can switch while in route between locations to another working route to keep the call up. Using VoIP, voice signals can be *packetized* like computer data packets. This enables companies to consider using the same network infrastructure to support both data and voice applications. Companies can consolidate their physical networks (while maintaining redundancy in their routing patterns) and build an enterprise-class communications network with the latest advanced IP-based features.

VoIP makes possible other services that older telephony systems cannot do. The VoIP protocols, or simply *IP* as many call it, are *interoperable.* This term means that the IP protocols will work well with all kinds of networks. VoIP is valuable because it fundamentally operates the same way in all types of networks. IP protocols are also highly *portable.* This means they will work with any IP-enabled end-user device such as an IP telephone, computer, or even a Personal Digital Assistant (PDA). VoIP works everywhere! Because VoIP is interoperable and portable, it makes possible many new end-user applications that redefine how we make telephone calls. Because VoIP is highly "networkable," it is useful to distinguish the two major network types used by most if not all companies.

Understanding How IP Works

IP Telephony works by taking traditional voice signals and converting them to a form that can be easily transmitted over a local area network. Thus, the heart of IP Telephony is the same as traditional data networking with computers. IP-enabled phones handle the voice-to-data conversion well, but don't be misled — implementing VoIP doesn't mean that everyone has to use IP-enabled phones. The best VoIP providers implement IP Telephony in a manner that protects your investment in existing telephone equipment, even if you have analog telephone stations.

All IP phones have one important thing in common: a built-in network interface card (NIC), just like the card a computer uses. The NIC is critical for any network device because it provides the device with a physical address and a way to communicate over the network.

The physical address supplied by a NIC is called a *MAC address.* MAC stands for *media access control.* The MAC address uses a standardized address, usually represented by six hexadecimal numbers separated by dashes. For example, the following is a valid MAC address: 00-0A-E4-02-7B-99.

To support IP Telephony, a server is typically dedicated to run the software used to manage calls. Servers are just like personal computers, except they have more memory, speed, and capacity. The server stores the database that contains all the

MAC addresses corresponding to all the IP telephone extensions assigned to users. Depending on the size of the LAN and the number of users, you may use more than one server. For example, some LANs running IP Telephony dedicate a server to handle voice mail.

Depending on the size of the LAN, one or more devices known as switches are installed. These *switches* are boxes that have a series of ports into which all LAN-addressable devices ultimately connect. (Examples of LAN-addressable devices include computers, printers, wireless access devices, gateways, and storage devices.) Usually the switches are set up in the communications closets around the LAN, and they operate 24 x 7. All the switches are interconnected, often with fiber-optic cable.

In a nutshell, all network devices, including your IP telephone, must physically connect to the LAN through a port on a switch.

Calling Just About Anywhere with VoIP

Maybe ET still can't call home with VoIP, but you can make IP-enabled phone calls as easily as old-fashioned telephone calls and with even greater mobility. The efficiency of IP happens behind the scenes, as this section explains.

Calling on the LAN

When you want to call a coworker at your same location, you dial the phone number corresponding to the person's name. The signals are packetized and sent to the managing server, where the packet picks up the MAC address of the person you're calling. Next, the packet is forwarded to the switch, then to a particular port on that switch, and finally to the IP telephone associated with that MAC address that's connected to the port. The coworker's telephone rings. When the coworker picks up the receiver or answers the call, a virtual connection is established between the coworker and yourself for the life of the call. IP Telephony does all this at lightning speed.

Calling on the WAN

The process of calling a coworker at an offsite location varies only a little from calling on the LAN. The call is still initiated in the same way. But because the coworker is connected to a different LAN, the local server sends the call packets to a special gateway switch located on your LAN which sends your call through the company's WAN (wide area network) to the destination LAN and telephone. This is where IP Telephony technically becomes VoIP.

Each LAN in a multilocation network is connected to the larger WAN. If you're located at the company's headquarters in Pittsburgh, and you call a coworker located at the office in Los Angeles, your call begins as an IP Telephony call on your LAN. It then travels from your LAN through a gateway, switch, or router that is programmed to repacketize your call and encode the VoIP packet with additional information, such as the address for the destination LAN.

Network gurus refer to the process of packetizing your voice telephone call as *encapsulation*. A good analogy for this fancy techno-term is putting a letter into an envelope for mailing. The difference is that these encapsulated packets contain the content of the telephone conversation in digitized form.

To participate in the company's VoIP WAN, each LAN needs at least one edge device, such as a router, a switch, or a gateway. An *edge device* is just that — a device that sits on the boundary, or edge, of your local network and provides a connection to external networks. Depending on the company's network design, these edge devices can even have multiple interfaces that connect them to more than one outside network. The edge devices take care of all the IP Telephony traffic going off-LAN by encapsulating the signals into packets, encoding the packets with the correct addressing information, and forwarding the packets out onto the WAN, where they make their way in a packet-switched manner to their respective destinations.

When the packets arrive at the destination LAN, the edge device on that LAN breaks down the VoIP packets and forwards them internally to the server that manages IP services. From this point, the rest of the process is similar to IP Telephony services described in the preceding section: The phone rings, the person answers, and a virtual circuit is established between the caller and the receiver.

Making calls to the PSTN

The same gateway device that enables you to call a coworker internally over the company's WAN also enables you to call people not located anywhere on the company's data network by routing your call over the PSTN.

If you're calling someone not on your LAN but in the same city (also known as your *local calling area*), the LAN edge device routes your call to the local telephone company's central office for completion over the PSTN.

In fact, your VoIP service can also route an off-network call to a distant city through the company's LAN and WAN and then hand off the call to the local phone company in the destination city for completion.

Gaining Flexibility with VoIP

VoIP is not just about making and receiving telephone calls; it's about a whole new way of communicating. Sure, it includes telephone calls, but there is so much more to the VoIP telephony picture. VoIP integrates most if not all other forms of communication. You can even run videoconferencing to your desktop.

With VoIP, your company enjoys increased productivity and customer satisfaction. These improvements are typically realized through the flexibility offered by enhanced calling features. A few calling features, such as voice mail and call transfer, have been around in the POTS world for quite some time. On the other hand, integrating data, voice, and video applications to run over a single network and work with wireless phones are more recent innovations made possible by IP Telephony.

One of the big stories with VoIP concerns the many new and exciting features that increase your ability to be agile and mobile. You no longer have to say "I've got to get to a phone!" VoIP can be on your desk, computer, mobile phone, or PDA. It can be hardwired or have no wires at all. This flexibility is astounding to those familiar with traditional telephony.

If you have a mobile user base, be sure to check out IP soft phones. A *soft phone* is software that works on a laptop computer or pocket PC and provides most of the functionality of a traditional desk phone. If a user can connect to a network, the soft phone provides a way to reap the benefits of IP Telephony regardless of location.

Escaping oversell and ensuring VoIP savings come with service

XOptions Flex, from XO Communications, is an integrated voice and data solution using Internet Protocol that can help small and midsize companies dramatically cut phone and Internet costs by taking advantage of VoIP. As its name implies, the service is flexible to fit the needs of different size companies, so you don't have to buy into a more expensive solution than you need. For example, XO showed a multi-location mortgage company how VoIP could reduce costly bills and improve their Web-based services.

Century Mortgage Corporation's business is helping clients find the best mortgage to finance their homes. Headquartered in Norcross, Georgia, with additional branches around Atlanta and in Virginia, the company employs approximately 100 people. The company offers interactive Web tools for clients to compare mortgages and check loan status, and it relies heavily on phone, fax, and Internet communications to conduct business in 11 states.

Peter Chen, IT Manager, wanted to reduce Century Mortgage Corporation's telephone and Internet bills, which amounted to over $8,000 per month — plus additional long distance charges. Making the situation more frustrating, the company's service provider seemed to have no interest in helping Century Mortgage identify a more cost-effective solution.

Chen set out to find a better voice and data solution for the Norcross headquarters and two branch offices. He searched for a new provider that would

- Accurately assess the company's voice and data needs and present the optimal solution.
- Significantly reduce excessive monthly telecommunications costs.
- Provide on-site service when help was needed.

XO quickly made Chen's short list of potential service providers.

Working with his account manager at XO, Chen quickly realized the two T-1s and two primary rate interfaces (PRIs) in the Norcross office were overkill. "We had been oversold by our previous service provider," Chen said. "We were paying for much more than we needed."

Century Mortgage replaced the previous service with XOptions Flex, choosing a business-class VoIP package of a single T-1 and eight voice lines all on one circuit. The company deployed the same configuration at two other offices, helping increase their data speeds from 256K to 1.5 Mbps. With long distance included in the bundled service, the variable cost for toll calls among offices and with clients was eliminated.

Not only did XOptions Flex prove to be a cost-effective solution, Chen was also impressed with XO's local sales and support team. He said, "After being virtually ignored by our previous service provider, it was very important for us to work with a company that would show up in person when we needed service."

Key features of the XOptions Flex package include:

- A single flat rate for local, long distance, inbound toll-free, Internet access, and Web hosting. (Flat rates vary by market; limits apply.)
- The ability to use existing telecommunications equipment already on premises.
- Dynamic Bandwidth Allocation, which assigns additional bandwidth to data throughput when phone lines are idle.
- The reliability and security of XO's national IP network and softswitch technology.

Because Century Mortgage's customers call, fax, and use the company's Web site, bandwidth was a concern. XOptions Flex offered the speed and quality of service required, while delivering a huge savings to the company.

After deploying XOptions Flex, Century Mortgage reduced its monthly voice and data costs from over $8,000 per month to under $2,500 per month — a savings of almost 70 percent. And XO can continue to expand with the company in the future; Century Mortgage has already ordered it for a new branch office.

Part II

Simplified Management with VoIP

In This Part

- Reducing operating costs by switching to VoIP
- Integrating data and voice
- Improving efficiency
- Meeting your future with VoIP

In 1995, when VoIP was first introduced, many analysts projected savings for companies choosing VoIP over companies continuing to operate with POTS-related telephony systems. However, a small percentage of early adopters ended up frustrated with the early forms of VoIP. This was mostly because these early VoIP systems were based on using the Internet itself as the underlying network transport. Although the Internet can support many computer data applications, it is clearly not the transport of choice for the corporate world to vest its core telecommunications infrastructure in. (VoIP does work over the Internet, but the quality of service for a mid- to large-sized company is not the same or even close to the quality obtained from a dedicated private network.)

Even though VoIP in 1995 was not ready for prime time, it showed promise, and several leading manufacturers took VoIP in its early form and redeveloped it into a highly effective form of telephony system. As a result, VoIP today not only saves companies huge amounts of operating expenses, it operates over the companies' private, dedicated, packet-switched computer networks.

Moreover, VoIP brings to the table a number of slick calling options. VoIP call features include all of the features offered in POTS-related telephony systems, as well as many other features the likes of which have never been seen before in telephony systems. VoIP features enhance the collaboration of employees across your business/enterprise and ultimately reduce the operating expenses of the company.

Saving Money with VoIP

One of the big "aha's" with VoIP is that companies can enjoy an immediate savings on their regional toll and long-distance voice and videoconferencing charges. Before VoIP telephony, everyone was critically dependent on POTS running over the PSTN with no other options for their telephone needs. That is why the majority of companies today have POTS-related systems in place.

No more leasing POTS lines

But VoIP is changing this because there are several disadvantages to companies that use POTS-PSTN when compared with VoIP telephony. POTS-related lines are leased from the local exchange carrier, which can incur added expenses. For example:

- Each line usually has a nominal startup charge.
- Each line has a monthly recurring charge known as the *access cost.*
- For every POTS line, the company must pay monthly recurring usage charges for local, regional toll (includes intralata and intrastate), and long-distance (interstate) services.

  All recurring service charges are based on a rate per minute per line basis. For example, a company on the average may pay $0.10 to $0.64 per minute for its intralata carrier services. (*Intra* means within the same LATA but outside of the local calling area. *LATA* means Local Access Transport Area.) It may sound like it is a trifling kind of cost factor. When you add up all the minutes from every line in operation, however, the cost each

> month can frequently get into multiple 6- and 7-digit figures. The effect is even greater when your company has multiple locations that cross intralata boundaries within the same state.

Some say that VoIP does not really save a company much on toll charges. Nothing could be further from the truth. Usually they do not consider the hidden recurring cost factor. This secretive cost factor is the intralata, regional toll (also known as *local toll*) cumulative charges that their company suffers from on a monthly basis. Some may lump all toll costs into the category of long-distance, which is another mistake.

If your company has significant intralata toll minute volume in the aggregate (all minutes multiplied by all POTS-PSTN lines), you can reduce or eliminate these charges by converting to VoIP.

No more charges for calling features

With VoIP your company gains many more features, several that run over the network, and your call feature costs go to absolute zero.

With a POTS line, the Local Exchange Carrier (LEC) charges for calling features such as added uses of the POTS line beyond simply making telephone calls. These features include options such as voice mail, call transfer, and call forwarding. Sometimes these features are priced out individually (a la carte), and sometimes the LEC will bundle features for a discounted price.

Most companies use an internal telephone system, so call features are a moot point; their system can usually provide most if not all POTS-related call features. However, with pure POTS and CENTREX line models (these models are covered in the next section), call feature costs are highly relevant to the company's monthly telephony bill. Remember that features are priced by the individual line. If your company has hundreds or thousands of lines, the overall cost for all features for all lines can be astronomical.

The "Old" Final Four Meet VoIP

In order to reduce the monthly recurring charges (MRC) for POTS line telephony services, companies with 15 or more employees who need a telephone can acquire their own telephone system. Over the years, several conventional systems have emerged. All of them use the POTS model as their baseline. But each one greatly reduces the dependence on POTS lines and POTS line equivalencies. Also, they all provide the limited traditional features at no extra cost. As a result, companies seeking to use conventional POTS services generally use one of the four non-VoIP telephony systems models. I call them *The Old Final Four.* Here's how they work:

- **POTS:** Companies with fewer than 15 phone stations that aren't bothered by high regional and long-distance toll charges can stay with the POTS line model. With the POTS model, the company does not need to consider a model — everything depends on the carrier. Each employee has a phone. Each phone has a POTS line from the carrier. The POTS line model is the oldest of the conventional telephony models. It is sometimes called the *wireline model.*
- **KTS:** The second model is called a *Key Telephone System* (KTS). The KTS is often referred to as a *Key Station model* or simply a *key system.* It reduces a company's dependence on total number of POTS lines. It provides at no extra cost many of the traditional call features.
- **CENTREX:** The *CENTREX services model* is owned and operated by the carrier. *CENTREX* stands for *CENTRal EXchange.* CENTREX service provides the physical equivalent of a POTS line. The lines run from the carrier's switching equipment to each telephone station. The carrier is responsible for maintenance under a CENTREX model.

  CENTREX costs more per month per line but often can include many of the features without additional charges. Before VoIP, CENTREX was a great solution for startups or companies unsure of their strategic plans because they could gain all of the usual features along with POTS equivalent telephony service very quickly under

a month-to-month plan. When the company's plans become concrete, they terminate CENTREX and convert to a new telephony system.

- **PBX:** The fourth model is known as the *in-house PBX*. Before VoIP, the PBX was the mainframe of corporate telephony. PBX stands for *Private Branch Exchange* or *Premises Business Exchange* and it is the most expensive of the four models under non-VoIP approaches to telephony. However, it delivers the most value out of all four as well. Some key value points are that the PBX can use:
  - Dedicated high-bandwidth lines out to the carrier or to other locations on the company's network.
  - Interfaces to provide full-motion videoconferencing.
  - Extensive Call Management capabilities for setting up and controlling multiple call centers.

Companies using their own system can reduce the total number of POTS lines required by a factor of one line for every six to eight employees. The phone system's circuitry integrates multiple users over fewer lines. With the PBX, videoconferencing and other high-bandwidth applications can be integrated. Although companies can reduce the total number of lines required and therefore their total MRC, they still have to pay for local and toll usage. But with their own system, they are able to provide most of the traditional telephony call features at no extra cost.

This is a great savings by comparison to having no system at all, but not close to the savings attainable through VoIP. If you total all the savings from any of these older system models, it would amount to a mere fraction of what your company could save with a VoIP system. Remember, VoIP all but eliminates regional and long-distance charges. For many companies, these charges alone amount to thousands of dollars per month.

See Table 2-1 for a summary of The Old Final Four traditional telephony systems models.

Table 2-1 The Old Final Four

System	*Location of Equipment*	*Cost Structure*	*Comments*
POTS	Carrier lines run to company-owned phones.	Monthly recurring charges (MRC) per line, per phone. Regulatory fees apply to access line costs.	Call features are paid per month per feature. Relatively high cost on a per employee basis. Not well-suited for VoIP conversion unless toll-charge savings justify conversion costs.
KTS	POTS carrier lines run to customer company's KTS switch.	MRC per line, startup cost of KTS and phones. Regulatory fees apply to access line costs.	Most features are included at no extra cost (savings due to one POTS line for every 6 to 8 phones). Suitable for VoIP if company has substantial MRCs for regional, intrastate, or interstate toll carrier services.
CENTREX	POTS-equivalent carrier lines run to customer's telephone on a per phone basis.	Higher POTS-equivalent line charges, MRC per line. Regulatory fees apply to access line costs.	Little or no maintenance costs, though lines are higher priced compared to POTS. Suitable for VoIP if company has substan tial MRCs for regional, intrastate, or interstate toll carrier services.
PBX	Dedicated carrier transport lines run to customer's PBX.	Dedicated access lines. Highest MRC per line. Dedicated amount of bandwidth. Regulatory fees apply to access line costs.	All POTS call features are available free. Call center capabilities. Higher monthly maintenance charges. Highly suitable for VoIP if company has substantial MRCs for regional, intrastate, or interstate toll carrier services.

IP Telephony Converges onto the LAN

Unlike POTS, which under any of The Old Final Four models is vested in the costly circuit-switched world of the PSTN, IP Telephony runs on the company's computer local area network (LAN). In all previous telephony models, companies either had to acquire a totally separate system infrastructure, or they had to pay the local carrier higher costs for the privilege of using their lines and equipment, as in the case of CENTREX. Depending on the model used, there were lots of charges for local, regional, toll, and long-distance carrier services — not to mention regulatory fees based on the number and type of access lines used.

IP Telephony is unregulated. It runs on the company's computer network infrastructure. With IP Telephony (sometimes called IPT) there are little or no additional charges for the core infrastructure or the access lines thereto. IP Telephony brings an immediate productivity benefit primarily because it takes less time to satisfy customers. IP Telephony therefore enables a much greater cost-benefit, a higher Return On Investment (ROI), and a reduced overall Total Cost of Ownership (TCO).

IP Telephony is good for the company. It is good for the end-users in the company. Most important, it is good for the company's customers. Making a move to IP Telephony has never been more strategically appropriate than it is right now.

VoIP and toll bypass

IP Telephony is basically VoIP on the LAN side of the company's network. With IPT running on the company's computer network, the company's need for POTS-PSTN lines is drastically reduced. The total number of POTS lines needed is reduced as much as 95 percent. A small number of POTS lines may still be required to meet local ordinances, such as for automatic fire alarm systems and to make certain types of local calls from the company.

Consider the possibilities

The converged strategy presents an opportunity not possible with separated computer, telephony, and video networks. Companies moving toward convergence can realign staffing resources to create a more flexible, agile, and supportive organization. This action alone begins to foster a collaborative spirit across the company's enterprise. Former computer data network support personnel can now share job-related tasks. Former telephony personnel can now help convert their traditional telecommunications infrastructure into a computer network-based telephony system. Cross-training no doubt will be needed and desired. Managers of these data, telephony, and video systems can unify under the banner of convergence.

The old saying "united we stand, divided we fall" has direct relevance to any organization in today's marketplace. Convergence is now underway and will eventually sweep across most enterprises. Companies remaining on separated system networks for their computer, voice, and video needs will be falling out of the marketplace. Companies wanting to stay competitive will need to consider the move at some point.

Fortunately, IP Telephony and VoIP are at a maturity point where any company today can plan for the conversion with the assurance that it will be a cost-saving move and a productivity-enhancing strategy. If you cannot consider it for the cost savings, do it for the enormous enrichment of your company's productivity. Productivity increases are certain with IP convergence. They result from the entire set of added calling features and seamless applications. In addition, productivity results from the collaboration among employees that is fostered by convergence throughout the organization.

Another big "aha" with IP Telephony is when calls are placed to other locations within the same company. Calls that would normally need to travel over the PSTN and outside the local calling area to reach the more distant company locations can be handled more efficiently with IP Telephony. If the company has IP Telephony running at each of its locations and these locations are all connected over the company's private wide area network (WAN), all calls placed to any of the company's locations totally bypass the PSTN. Hence, you will hear the term for VoIP known as *toll bypass.* The PSTN is not involved at all in any such calls.

This fact may sound unremarkable, but it makes a huge difference in the company's toll-related monthly recurring service charges. For calls originating on the company's IP network that need to go to distant locations that are not on the company's IP network, the call would travel *on-net* to the company's closest location to the destination of the call, then at that location normally go *off-net* and convert to a local call over the PSTN. Such calls originating on-net that need to travel off-net are forwarded over the WAN to the distant LAN. A gateway device attaches the LAN on the inside and the PSTN on the outside. Through this gateway device, the call is passed to the PSTN. Instead of paying for a toll-based long-distance call, the company merely pays for a local call. IP Telephony and VoIP work together using the IP protocols to support telephony across the company's computer network. As a result, IP Telephony and VoIP reduce significantly or totally eliminate all toll charges and former POTS-PSTN telephony-related regulatory costs.

Integrating IT (and more) through Unified Network Management

The new term for putting a company's telephony systems and videoconferencing systems onto the company's computer data network is *converged network*. The term revolves around the fact that when this is done the former circuit-switched telephony and video systems are "converged" onto a dedicated packetized network. Another popular term for this is *integrated networking*. Integrated networks incorporate the use of computer data, telephony signals, and video signals onto the same network.

One Network versus Three

Despite all the advantages that exist for a company to move toward IP convergence in today's marketplace, you may be surprised at some of the reasons that are given for not even considering it. No doubt some companies will have to go into Chapter 11 before they realize the error of their strategic planning.

In the following sections, I explore some of the reasons for companies' reluctance to make the move to IP convergence, as well as some sound reasoning to counter their arguments.

If it ain't broke, why not improve it?

Most companies are reluctant to change systems simply because they are comfortable with what they have. "If it ain't broke, don't fix it." Yet many companies have experienced downtime with computer, telephony, and video networks. It is just a fact of network life.

No downtimes are more memorable than telephone system downtimes. Data and video network failures for some reason are always seen as a temporary situation and an easy fix. But if the telephony system network is down, it is a major crisis. Companies would rather tell their customers that they are wounded and unable to respond to their needs than to have them call in and get a busy signal or even a "fast busy" signal.

It took more than 100 years for the industry to shape the quality of service that now characterizes POTS-PSTN calling. Computer networks have been standardized for just 20 years. Sound management is required to consider, understand, and drive the strategic move to VoIP in a company.

Convincing your boss

Part of gaining the support for the move to IP convergence is to convince the company that it is the right move. You generally need sign-off by the people who manage the company's technology and, to a certain extent, the staff that reports to these managers. But you know and I know that by and large it comes down to convincing upper management.

The best way to appeal to upper management is to focus on the convergence as a cost-effective solution to an expensive problem. Map out your current expenses and lay those numbers side-by-side with the expense of IP convergence. The numbers speak for themselves: IP convergence reduces operating expenses enough to pay for itself in the near term, and it can make the company a whole lot of money going forward.

Another benefit that speaks to upper-level managers is the fact that implementing an integrated network brings the company together, makes all employees reachable on a higher, horizontal plane of communication. It promotes collaboration, enhances productivity, and ultimately leads to an increase in revenue.

Last, you'll need to provide your management with a seamless plan for transitioning to the new system. For the full scoop, see the next section.

Meeting the "New" Future Four

After deciding to switch to VoIP, your company has more options to fulfill its IP integration dreams. This section discusses four of the most frequently used VoIP configuration solutions.

Integrated VoIP service

Integrated VoIP Service is often the first step for many small-to-midsized businesses looking to switch from traditional telephone service to new VoIP service. This service allows you to keep your existing telephone equipment, even if it is not IP-enabled, and get many of the key benefits of VoIP, including better features, more flexibility, and significant cost savings.

A dedicated T-1 circuit connects your premise-based key, hybrid, or PBX system to the communications carrier's VoIP-enabled networks via an *integrated access device (IAD).* Instead of separate networks, all voice, data, and Internet traffic is combined on a single T-1 line. VoIP technology dynamically allocates bandwidth in real time to meet your voice and data needs. Sensitive voice calls are always prioritized, but unused bandwidth is made available for less-time-sensitive data traffic when available.

Hosted IP PBX service

An application server hosted in the service provider's IP network delivers a complete suite of enhanced PBX functionality over any IP connection. The service provider can connect your business simply by setting up and configuring your business

features and applications. Service providers target businesses with as few as five end-users to large enterprises with thousands of users distributed across the globe.

Enhanced functionality such as Auto Attendant or Attendant Console may be provided that enables the hosted system to answer calls with a recorded greeting, play a menu of connection options to callers, and then route the call to an appropriate employee extension or to a holding queue for a specific department, such as Sales or IT Support.

In a premise-based PBX system, each extension in the system is associated with a specific telephone location, usually a desk phone somewhere in the office. In contrast, in a hosted system, each extension is associated with a particular person. This person can take the call on any phone, anywhere in the world.

Service providers offer a *portal* (Web interface) where each end-user can manage his or her phone service. For example, employees typically can manage their own features and applications (including such valuable services as Find Me – Follow Me, Outlook Integration, and Presence Management) via this portal.

Managed (premise-based) IP PBX service

Most communications carriers now offer a turnkey-managed IP PBX service for organizations with a centralized headquarters but lacking the resources to manage, monitor, and maintain a VoIP system.

With a managed IP PBX Service, the communications carrier provides you the PBX, phones, and network as part of a bundled package. The carrier also provides installation, training, maintenance, monitoring, and *MAC (Moves, Adds, Changes)* support.

By signing up for a managed IP PBX service, organizations can generally lower their total cost of ownership, improve overall system reliability while freeing company resources to focus on their core business. The company gets one monthly bill and one person to call when there are problems.

SIP trunks

In a VoIP converged network with Session Initiation Protocol (SIP), organizations can choose from a variety of vendors to create a seamless, converged communication network. For instance, some equipment vendors use SIP to support trunking functionality. (*Trunking* is a way to make a network support a protocol it might not otherwise support.) Other vendors may use SIP to control gateways and calling features. The way in which SIP is used is entirely up to the equipment vendors. SIP allows their equipment to communicate with equipment from other vendors.

SIP trunks enable seamless voice and data convergence connected to an IP PBX that is owned and managed by the business. The SIP trunk provides both connectivity to the Public Switched Telephone Network as well as providing data connectivity to the Internet. SIP trunks simplify network management and optimize network expenditures.

SIP does to traditional telephone service what the World Wide Web does to the Internet. In fact, SIP is a cousin of the main protocol of the Web, hypertext transfer protocol (HTTP); both are text-based protocols.

SIP has emerged at the forefront of most, if not all, VoIP-related applications. It is faster, easier to scale, and easier to implement than previously deployed VoIP protocols. SIP is an open protocol which enables businesses to deploy an architecture that more easily supports the integration of multi-vendor telephony. In a converged network that uses SIP trunks, organizations can choose from a variety of communications carriers and equipment manufacturers to create a seamless converged communication network. Consequently, SIP has been embraced by the leading VoIP telephony manufacturers and is being built into VoIP hardware and software, including IP-enabled telephones.

SIP integrates with traditional circuit-switched interfaces and IP-switched interfaces. This integration enables the user to easily convert from traditional circuit-switched telephony infrastructures to next-generation IP infrastructures, including wireless networks supported by WiFi and WiMax.

Protecting Existing Telephony Investments (While Enjoying VoIP Efficiencies)

The good news is that integrating IP Telephony and VoIP onto your computer network can be done while keeping your conventional POTS-PSTN telephony systems operational. Because the two are physically separate networks, they can operate simultaneously.

If you work with a carrier company that supports IP Telephony and VoIP-based telephony, and a hardware vendor that provides hardware to support both types of networks, you can enjoy your conversion to VoIP and still have the security of the older system.

Typically, the provider companies offer reduced cost to keep the old running while you install the new IPT- and VoIP-based systems. When you are comfortable with your new converged and integrated network, you can plan for the removal of the old telephony systems and the termination of any non-used carrier services.

If your company has made a significant investment in telephony systems that were not IP-ready but IP-capable in the last couple of years, your company can now plan the move to IP Telephony and VoIP while still protecting the company's investment in IP-capable systems. This is another less costly way to reap the full benefits of your original IP-capable systems while you position your company for the eventual full conversion to IP Telephony and VoIP.

Your company can begin to save operating capital on toll-bypass, for example, to prepare for the costs of the full VoIP conversion status. This includes using, for example, digital desktop telephones that may have already been acquired. The telephones can connect to a now IP-Enabled PBX that could be the main telephony equipment connecting to other IP-Enabled PBXs over your company's WAN.

Whatever IP Telephony and VoIP conversion option your company may choose, you will be running a single network that integrates computer data with telephony voice and video if used. The requirements for managing the company's network become more unified versus divided. A single comprehensive network management system can be used to count every bit and byte on the network. Fault-isolation can be more readily processed because you do not need to troubleshoot what network the problem may be on. There is only one VoIP network with one or more distinct LANs running IP Telephony.

Because your company will unify its support staff into one department, the ensuing cross-training and convergence experience to be gained by all in this department can result in a reduction of the company's dependence on outside experts. In the short-term, your company may need to use outside contractors, or they may leverage their business volume to have their existing providers support their needs until the conversion is at or near completion.

Integration of the company's computer data, voice, and video systems strengthens the company's infrastructure. And, unification of the respective support staffs ensures that the company can succeed in the forthcoming converged marketplace.

Planning Your Future with VoIP

In a competitive marketplace, companies that are forward-thinking look at their competitors. Market projections based on a mere percentage of the total telephony marketplace indicate that the IP Telephony market could grow to as much as $15 billion a year by 2008. Companies are expected to make the move and have already begun to do so. This trend means one or more of your company's competitors are making the move and enjoying all the benefits. It also means that your company will be at a disadvantage if it does not undertake a strategic plan to convert to IP Telephony and VoIP. As collaborative companies with a unified workforce satisfy their customers in unprecedented ways, they are going to increase their respective market shares. Consequently, your company may not be able to afford to ignore IP Telephony and VoIP technologies.

Bandwidth on demand

Besides the movement of the market including your competitors towards VoIP, you need to evaluate a couple of significant technical benefits. First, IP Telephony and VoIP networks support the kinds of network transport services that run packetized services not only for computer data, but telephony voice as well as video where needed. These transports are usually dedicated lines of substantial bandwidth capacity.

Bandwidth is normally *channelizable,* which means that the bandwidth of the line can be divided into channels. The channels can be used *dynamically* (whenever they're needed for a specific application to run on them at any point in time). When channels are not needed, they go back into a pool of channels for other applications including data, voice, and video needs. This type of operation is often referred to as *bandwidth on demand.*

To achieve this type of bandwidth usage, the network architecture uses select types of terminating equipment called *Level Three switches.* Network service providers that supply the transports usually include or specify exactly what model of switches fit the bill. Bandwidth on demand is a function of the WAN network design that works very well with VoIP.

Scalability to size up or down as needed

Scalability refers to the degree to which your company can make changes to support growth, as well as increase access to and use of the IP Telephony and VoIP network.

On the IP Telephony LAN side of the network, each LAN uses an Ethernet LAN, which is highly scalable. New users, IP telephones, computers, and other devices can be connected to the LAN on a plug-and-play basis.

When an employee needs to move to a new location in the building, for example, their IP telephone and computer can be unplugged and taken to the new location, where they are plugged back in. Both devices relearn automatically on startup the identity of the employee. The devices are operational immediately.

Humanscale reduces costs with VoIP

Sometimes the greatest cost savings with VoIP comes with reducing the number of telecommunications providers. For example, Humanscale is a leading designer and manufacturer of ergonomic products for the workplace, including seating, keyboards, lighting, desktop and laptop holders, and extendable arms for computer monitors. The company has two manufacturing facilities in the United States and one in Ireland, with products available through retailers, contract furniture dealers, and from 14 sales offices. The company decided to switch from six telecommunications service providers to one service provider, XO Communications.

Humanscale wanted to take advantage of VoIP, but lacked the budget for a massive "rip and replace" overhaul of its existing phone system. The company elected to deploy XOptions Flex, XO's industry-leading VoIP services bundle, at three of its leading sales office locations: Baltimore, New York City, and Washington, D.C.

At each of these locations, Humanscale is connected to the XO IP network via a T-1 connection, with XOptions Flex providing an IP-based integrated voice and data solution. All three locations have local and long-distance minutes, dedicated Internet access, and dynamic bandwidth allocation, all for a single, flat, monthly price. (Flat rates vary by market; limits apply.)

XOptions Flex enables Humanscale to bridge the gap between TDM and IP voice services without having to invest in IP phones and new network equipment. Installing XOptions Flex at the three selected locations simply required an XO T-1 connection and an IP-integrated access device at each office. XO then worked with Humanscale to connect its existing analog phone system to the integrated access device, so employees have full use of the data and VoIP capabilities of XOptions Flex.

The XOptions Flex Web Administrator portal enables Humanscale to self-administer real-time changes to service features such as voice-mail passwords and calling capabilities for each employee at the three key locations.

Another advantage for Humanscale is being able to work with a national provider like XO that could support the company's locations across the United States. And when Humanscale is ready to deploy VoIP across all their offices, it can turn to XO to meet those needs as well. Humanscale estimates that it can save $270,000 per year with XOptions Flex and other bundledservices from XO — money it can invest in a new accounting software package one year earlier than originally budgeted.

Consider the benefit of this capability: No one needs to go to the telecommunications closet and reprogram their port numbers or change their network addressing information. This applies not only to a user on a single LAN that might exist on a multi-location, multi-LAN WAN; it includes all users anywhere on the WAN.

The VoIP protocols bring a certain degree of intelligence to the enterprise network that makes change a pleasure and a joy rather than a frustrating, time-consuming hassle.

Making easier moves, adds, and changes (MACs)

As a result of the high degree of scalability and the intuitive intelligence of VoIP networks, MAC changes are a thing of the past. Companies that still operate under one or more of the traditional telephony systems still must pay for expensive MAC changes: every time that an employee needs to move, when they need to add new users, or when telephony systems need to make changes in a user's telephony system profile.

Older telephony system technicians that complete these MAC changes bill out up to $150 per hour. Imagine if the company had to make a major set of moves or changes. At the least, these changes would be costly and time-consuming.

Larger companies may hire a staff of qualified technicians to do these MAC changes on a full-time basis. Under VoIP, MAC changes go away. Again, more cost and time savings to justify your move to VoIP.

VoIP carrier service providers can be differentiated in terms of service quality relating to compression and network speeds. How much of a commitment will the carrier make to ensure a high quality of VoIP service? You want to read in your service level agreement (SLA) how much speed and bandwidth you can expect. Reputable carriers usually list 256 Kbps as the minimum speed and as high as 1.536 Mbps as the maximum speed. If they state that it is "best effort" within this range, that is not so bad. If they state nothing or give no range, move on to a different carrier.

Part III

Three Phases to VoIP Migration

In This Part

- Convergence comes in phases
- Understanding quality of service

Whether your organization has a communication infrastructure that is multi-vendor and widely distributed or one that depends on a single vendor for your computer data, telephone system, and videoconferencing networks, you need a reliable way to integrate and optimize your network infrastructure.

Migrating to Converged Communication

XO sees the evolution and integration of corporate technology infrastructures in three phases. Naturally, companies will evolve portions of their data, voice, and video networks from one phase to the next according to their business needs. Given today's economy, organizations' business needs will no doubt compel them to be in more than one of these phases at the same time. The three phases are identified as follows:

- Traditional
- Converged Networks
- Converged Communications

Where everyone starts: The Traditional Phase

Companies operating in the Traditional Phase typically have separate physical networks for data, voice, and video (if used). Each location usually has its own LAN, and the company as a whole has a private, dedicated WAN running IP protocols for computer data. If a mid-sized business has a small number of locations, it may vest its WAN infrastructure in a Virtual Private Network (VPN) that optimizes access costs by using the Internet as the WAN transport.

Non-profits use XOptions Flex to reduce costs dramatically

East Orange Child Development, a nationally funded Head Start program, and East Orange Community Charter School are two non-profit organizations closely tied to the East Orange, New Jersey community. They offer children and parents critical educational and social services to help improve early education opportunities. With a heavy daily load of phone calls, faxes, and e-mails, telecommunications services are essential to the organization's successful operations.

The two non-profits needed to reduce burdensome monthly telecommunications costs while deploying flexible voice and data services. With constant communication taking place with parents and other community members, East Orange Child Development and East Orange Community Charter School were paying voice and Internet access costs of more than $6,800 per month, and they were using a DSL line, which didn't always provide the quality of service needed. Consultant Hasjonn Simmons, responsible for the organization's technology infrastructure, said that paying for local, long distance, voice mail, and other features for each separate line was complex and costly due to dealing with multiple vendors.

The organizations wanted to replace their multi-vendor solution with an integrated VoIP solution. After they performed a cost analysis, Simmons discovered that XOptions Flex offered the simplicity, flexibility, and value his organizations required.

Since deploying XOptions Flex in May 2005, East Orange Child Development and East Orange Community Charter School have paid a flat rate for voice and data services across multiple locations encompassing 58 phone lines and two T1 lines. Key features of the new system include:

- A single flat rate for local, long distance, inbound toll-free, Internet access, and Web hosting across five sites. (Flat rates vary by market; limits apply.)
- The ability to use existing telecommunications equipment already on premises.
- *Dynamic bandwidth allocation,* which assigns additional bandwidth to data throughput when phone lines are idle.
- *Online feature management* that enables Simmons to manage mail boxes and hunt groups — without having to place a call to vendors and wait for service.
- The reliability and security of XO's national IP network and softswitch technology.

The bottom line: Since deploying XOptions Flex, East Orange Child Development has saved 64.6% on telecommunications costs. East Orange Community Charter School has saved 59.7%. In addition, a flat-rate monthly bill from a single vendor rather than multiple variable bills from a number of different vendors makes it much easier to track costs and plan budgets.

The ability to use existing analog phone equipment for a VoIP solution reduced the need for new capital expenditures. Simmons said replacing the telecommunications infrastructure might have cost $90,000 or more, which would have been cost prohibitive for the organizations.

"We've been able to apply the savings to other parental and childcare programs that give back to our community," Simmons said. "In addition, the inbound toll-free number included in our flat rate has given parents a single contact number to reach us from anywhere with any question."

Telephone system needs are typically met through one or some combination of two or more of the Final Four models covered in Part II. Unlike the packet-switched network infrastructure of the computer data enterprise network, telephony system needs are ultimately met through the circuit-switched protocols of the PSTN. In-house PBX systems, which may be interconnected over dedicated lines using Time Division Multiplexing (TDM) protocols, is about as good as it gets.

Videoconferencing solutions depend on the size of the enterprise and the type of videoconferencing application needed, such as point-to-point, multipoint, or desktop. The videoconferencing needs of a business can be met by using dedicated or switched transports that run physically apart from the data and voice networks. Or these needs can be met by using the voice infrastructure (in the TDM world, underlying video requirements tend to follow those of voice) with some modification. For example, terminating equipment to support video would be needed at each location to support the application. But a video module can be used in the PBX to bring up a video call and to dynamically allocate bandwidth for the life of the video call.

In the Traditional Phase, on an interim or permanent basis, VoIP Gateways may be used to support POTS-related calling from the LAN side into the PSTN. Companies operating in this phase typically use cheaper, switched multi-channel transports such as a Primary Rate Interface (PRI) line to the PSTN. Quality of service equals that of POTS.

Making progress: The Converged Networks Phase

In the Converged Networks Phase, most enterprises build out their computer data networks to support IP Telephony on the LAN side at all locations and VoIP on the WAN side. As a result, one common infrastructure exists across the enterprise to support data, voice, and videoconferencing. This arrangement enhances the IP network to meet enterprise-class criteria, such as improving quality of service and increasing the reliability of real-time, mission-critical business and communication applications.

The organization benefits from a distributed communications architecture that minimizes the monthly recurring cost of transport access lines into both the dedicated and switched carrier services networks. Dynamic bandwidth allocation is optimized across all applications. In addition, the toll charges associated with the traditional regulated carrier services of the PSTN are minimized if not eliminated altogether.

In addition, the organization can begin to develop integrated data, voice, and video applications. Most if not all of the call features described in Part I become available across the enterprise. As the higher recurring costs of running separated networks are driven out of the budget, more operating revenues are made available for other business needs. As the organization deploys and leverages its IP infrastructure, it positions itself to integrate new applications as they become available.

Getting there: Converged Communications Phase

As enterprises become more distributed and business performance needs dictate enhanced user capabilities, converged communications applications are deployed. Converged communications lead to increased flexibility and cost efficiency due to modularization of components and applications. As solutions become more modular, their services can be deployed in a greater number of configurations and more easily integrated into multi-vendor environments.

Improving Quality of Service with VoIP

With traditional telephony, every call is given a dedicated channel for the length of the call. Although this setup contributes to high call quality, bandwidth utilization is low and calls are blocked when the network cannot find a full bandwidth channel available. Somewhere along the line, the folks running the PSTN decided that a breakdown in connectivity was preferable to a breakdown in quality — an expensive problem that VoIP services fix.

Coping with packet problems

When considering telephone *quality of service (QoS), toll* quality means the highest level of service. VoIP service can meet or beat traditional toll QoS if your communications provider has built and operates your service to the right standards.

The packet loss trade-off

Packet loss was an accepted part of the way things used to work. TCP, the network protocol used for Internet, e-mail, ftp, and many other types of traffic, is designed to use as much network bandwidth as it can. The sender sends TCP packets that are part of a session, gets a response from the receiver, sends more traffic, and waits for a response — and keeps doing this until the receiver doesn't say it received all of the packets. Then it sends fewer packets and the process continues until the entire file is transferred. What happened to those lost packets? They were dropped because of congestion.

This behavior makes Web pages load as quickly as possible, but it doesn't coexist with VoIP traffic very well, where the VoIP traffic is using the same amount of bandwidth all the time and expects packets to be delivered in a timely fashion.

What's really happening when there's congestion? When a router or switch has a packet to send, it decides what path to send a packet down and then lines that packet up behind other packets taking the same path. If it's a very fast path, such as one on a local area network, the given packet may have to wait behind only one or two packets before being sent, which won't affect voice quality.

However, there are conditions that can affect VoIP quality if not managed properly. Four main problems can potentially affect VoIP quality:

- **Packet loss:** May make the call sound noisy or choppy, or just make the speaker's voice hard to understand. Packets carrying the audio signal from the speaker to the listener are lost on the way. The most common reason for packet loss is too much traffic on one link on the path that the packet has to take in an unmanaged network, also known as congestion. Congestion over even a very short period of time can affect voice quality, and in a network used for more than just voice it's common for this to happen.
- **Delay:** Doesn't affect audio quality, but too much delay makes it hard to carry on a conversation. A caller may start talking when she hears silence, but if there is too much delay the person on the other end may have started talking already. These "collisions" in the conversation make the call a frustrating experience.

- **Jitter:** Refers to the difference in the length of time between received audio packets, which may create unnatural delays or gaps in the phone conversation. In a VoIP system, the sender is sending packets at a regular rate, say 50 times a second, or every 20 milliseconds. If a VoIP packet has to wait behind other traffic to be sent over a link in the network, it will take longer to get to the receiver than one that doesn't. A little jitter is to be expected, and all VoIP devices are able to deal with jitter up to a point by using a jitter buffer. The problem occurs when the jitter is more than the receiver is prepared to deal with, and it has to drop the packet. Most VoIP devices can adapt to jitter, and increase the length of time they wait for a packet, but if the jitter is too severe, this increases the overall delay that the callers hear.
- **Signal attenuation:** Refers to the degradation of a signal over distance and time. Did you ever drive out to the country or into the mountains with a radio on? As you get farther from the source of the signal, the radio fades out or you hear more static. This is similar to signal attenuation over the VoIP network. VoIP packets are represented on many transports as a change in voltage, and that voltage can degrade over time and distance. By the time the packets arrive at their destination, the signal inside the packets is no longer in its original form. If attenuation is extreme, the person you're calling may not be able to distinguish who you are or what you're saying. Attenuation can be reduced or eliminated by improving the speed at which packets are delivered on the network.

Prioritizing voice packets with VoIP

For VoIP telephony to have the same or better quality as the PSTN, your VoIP service must be implemented over a dedicated private carrier IP network that priortizes voice packets. VoIP services implemented without voice prioritization or using the PSTN suffer by comparison.

First, a VoIP endpoint (or sometimes another point in the network) classifies traffic as being VoIP traffic or not. Then as the traffic sent through the network is prioritized, voice traffic gets sent before non-priority traffic, such as Web traffic. As a result,

if there is congestion, the priority packets make it through first and other non-priority packets are lost. Also, the priority packets don't have to wait behind non-priority packets, thereby reducing jitter.

You need to be careful with how traffic is classified on the network. If anyone can classify traffic as priority, you may end up with all traffic in "first class," which has the same result as no traffic being treated as priority traffic. Normally a service provider limits either what traffic can be prioritized or who can send prioritized traffic.

Part IV

Ten Reasons to Switch to VoIP Now

In This Part

- Changing direction of the telephony industry
- Feature-rich, cost-effective alternatives
- Existing investment protection
- Flexibility and portability
- Enhanced network management
- Better utilization of personnel
- Productivity applications
- Better bandwidth utilization
- Reduced costs
- Experienced service

The reasons to switch to VoIP are countless, depending on how far you want to project the future of the marketplace. For now, here are the ten best reasons to make the switch.

Changing Direction of the Telephony Industry

Over the next few years, much of the $300 billion per year telecommunications industry will migrate and convert its equipment and carrier services to support packetized VoIP services on the WAN. It will not be long before traditional telephony systems providers are outdated.

As older providers lose customer base and revenue, they will streamline operations and eventually close their doors. The providers that stay in business will need to increase prices and therefore become noncompetitive. VoIP technology has become the strongest influencer in the telecommunications provider marketplace.

As VoIP emerges worldwide as the number one replacement for traditional circuit-switched telephony infrastructure, manufacturers of telecommunications gear will convert their product lines to meet customer demands for VoIP-enabled systems. The same holds true for network services providers. They will convert their core service offerings to give priority to VoIP-related services. In fact, this is already occurring with most major carriers. The demand for circuit-switched equipment and network services will decline. As a result, the cost to suppliers who stay in the circuit-switched niche will go up. These costs will need to be passed on to the customers.

In light of newer VoIP products and services, customers will want to convert to VoIP so that they have adequate support available from outside companies. Many companies will also want to develop the VoIP skills of their in-house personnel. In this way, companies can insure their long-term growth by reducing costs and increasing revenue. VoIP can save companies lots of money in operating expenses, but if you have a multilocation company, converting to VoIP does require planning and VoIP skills.

Feature-Rich, Cost-Effective Alternatives

Most traditional telephony calling features have made their mark on the industry. Features such as voice mail, call transfer, call forwarding, and three-way calling have become familiar to all of us. The costs of these features are either rolled into the cost of your company's private telephony system, or you pay for them a la carte.

All traditional telephony features as well as many new features and communications applications are available in the brave new world of IP Telephony. The number of calling features is overwhelming. And they all come with no additional cost because they are IP-based and are carried over the computer network.

Simple features such as being able to look at your telephony station and see a visual indicator that tells you whether someone in your calling group is "present" but at the moment on the telephone can help increase employee productivity. (Think how many times you wasted time calling people, only to get a busy signal or voice mail, not knowing whether or not they were at their desk.) The presence feature is just one of many features available with VoIP.

Or how about the ability to run a soft phone on your computer and do telephony using a point-and-click process with a headset? Such a capability would never be contemplated in traditional telephony because that world can't support computer-related applications in a seamless manner. Many other calling features are available in VoIP, all just as compelling to companies considering a change.

Existing Investment Protection

If your company has a traditional telephone system (such as a PBX or KTS) in place, you can protect your investment by adapting the system in the new VoIP network. The PBX system probably includes many digital telephone stations. These telephones can also be reused in the new VoIP environment.

Your company can migrate to VoIP while protecting your existing telephony hardware investments.

A *forklift* upgrade is when you get rid of everything from the older system and therefore lose your previous investment. The other approach is to use some or all of your existing equipment. With the right VoIP partner, you can avoid forklift upgrades to VoIP.

Flexibility and Portability

IP Telephony has spawned many applications that increase both the flexibility and portability of communications. For instance, a soft phone provides mobile employees with easy access to real-time communications and the same calling features enjoyed by stationary employees. Users have never had more telephone options for mobility. Wireless extension to cellular enables a "follow-me" feature so that employees can have calls ring at both their office and cellular telephones.

In a VoIP network, employees can travel to any of the company's locations, plug in their IP-enabled laptop, begin work, and make and receive telephone calls. Employees have, at their distant temporary location, all the rich features available to them at their home office location. The network automatically identifies the user and applies that user's profile information. Employees can even direct their calls to any digital desktop telephone at the temporary location. (The telephone does not even have to be IP-enabled.) Managers no longer have to make costly and time-consuming accommodations for computer data and telephony connections for a coworker visiting their location.

Enhanced Network Management

VoIP provides a foundation for comprehensive network management. As a result, the ability for you to manage every bit and byte that runs over your LAN and WAN has never been easier.

Likewise, you have at your disposal tools that find and fix network issues so quickly that managers rarely know anything happened. These types of tools can support local and remote network monitoring. In dedicated networks, near-perfect quality is provided. That's not to say that problems *never* occur, but with a VoIP network, your ability to detect symptoms and make changes to your setup in advance of any problems is greatly enhanced.

Better Utilization of Personnel

VoIP enables the realization of a converged network — data and telephony traveling over the same network. Gone are the days when you needed two different skill sets to maintain your networks (one for telephony and one for data). Although there are some skills unique to VoIP that traditional network engineers don't have, the underlying skills related to Ethernet networks and IP protocols are the same. This allows your company to maximize the training of your people and, in many cases, reduce the number of personnel you need in-house to support the network.

Productivity Applications

Many of the Web applications that previously ran exclusively over the Internet will now run over your private VoIP-based communications network. Your users can have their favorite Web page displayed on their VoIP telephone, or they can post special Web links on their telephone-based Web page. Many Web-based applications are candidates for running on your VoIP telephones.

Users can also add a video telephony solution, powered by IP video application software that enables a desktop PC or laptop to emulate an IP office phone. The quality of the video and audio that runs on the company's network, versus the Internet, is free from latency and jitter.

Better Bandwidth Utilization

Many people wrongly assume that when you add VoIP to an enterprise computer network, there won't be enough bandwidth available to support the change. The fact is that dedicated network transports supporting computer data or traditional telephony systems are about 30 percent utilized. Even though running both data and voice packets over the same network increases overall traffic, you must look at how the IP-based traffic operates.

On the LAN side, fault isolation provided by switching equipment maintains a steady mode of operation. If any chokepoints are identified, they can be remedied almost immediately by changing connection points or doing what the gurus call *load balancing*. But your IP-based management system will tell you this before it ever becomes a problem.

On the WAN side, the load needs more consideration. You usually have more than one site on the WAN side that may have users connecting to your site. In addition, the cost and overall bandwidth capacity of WAN transports are higher and recur monthly when compared with the LAN side.

For example, a T1 line has 24 channels. If you run traditional circuit-switched calls over the T1, you can maintain 24 simultaneous calls. The beauty of VoIP is that it is packetized, so

the same 24 calls could run through just a fraction of the T1's overall capacity. As a result, you gain multiple times the bandwidth equivalent with VoIP compared with circuit-switched telephony.

Reduced Costs

The cost reduction argument is compelling from a couple of perspectives. The argument is never more persuasive, however, than it is for companies that have a substantial volume of toll calls charged by the minute. VoIP can reduce local charges; that's a good thing. But VoIP also reduces or eliminates most other classes of toll charges and greatly reduces your regulatory fees. That is a great thing.

Depending on the number of locations your company has and how many toll boundaries your current calling plan covers, you can save big bucks. These savings are derived primarily from putting all your locations on VoIP and bypassing most if not all of your toll charges. If your organization has significant international calling, the same benefit accrues, except that your company can save even more on toll and regulatory costs.

Experienced Service from XO

You don't have to blaze your own trail into the VoIP frontier. You can work with a service provider with proven VoIP capabilities: with the right products, the expertise, and the business sense to help you succeed.

XO Communications can help you migrate to VoIP services simply, flexibly, and cost effectively — without throwing out your current telephone equipment investment. The basic VoIP bundle, XOptions Flex, enables you to extend the useful life of those assets while enabling you to take advantage of the productivity and economic gains of VoIP today. And, the XO VoIP product portfolio offers an array of premise or network-based solutions to meet your company's business needs.

Part V

Ten VoIP Myths

In This Part

- VoIP runs only on the Internet
- POTS is cheaper
- POTS is faster
- The quality of service is suspect
- VoIP-enabled phones are pricey
- VoIP calls can be intercepted
- 911 calls may not work
- VoIP is not ready for prime time
- VoIP call features are expensive
- You have to throw out all your old PBX telephones

If a new technology comes our way that brings with it the promise of reducing or eliminating tremendous monthly costs, it can be expected that supporters and stakeholders of the status quo are going to be concerned. Consider the effect of the horseless carriage on commerce back in the early twentieth century. As a new market for automobiles emerged, many counterarguments attempted to slow its growth. But by 1910, there were an estimated five hundred thousand cars and the industry was growing fast. Stable owners, for example, became garage businesses. Saddle makers got into the business of providing seats for the new cars. It didn't take long for society to adopt the automobile and adapt to its presence.

In a similar way, VoIP is disrupting the $300 billion telecommunications industry. VoIP has had to overcome many criticisms. This chapter addresses what many experts consider the most prevalent myths about VoIP.

VoIP Runs Only on the Internet

The most obvious myth about VoIP is that it runs only on the Internet. What can we expect? The term *Internet* is built into VoIP. However, VoIP requires and runs on the Internet protocol, but not necessarily on the Internet itself.

Internet protocol is not synonymous with the Internet. The Internet can be accessed from all types of networks. The Internet is not a network type unto itself; it is a network that is accessed by other networks and provides access to other networks.

VoIP runs on any network that can run the Internet protocol. This includes every network type known to man. (Part IV discusses different network types.) But just because VoIP can run the same protocols over any network type, this does not mean that that VoIP runs the same way on all network types. The protocols take care of packetizing the telephony voice signals, but the network type takes care of transporting those packets.

POTS Is Cheaper

You might be inclined to say that VoIP is much more expensive than its older POTS counterpart. This comparison needs to be considered from several perspectives.

First you should think about the chronology. As with any technology, early pricing is never cheap. VoIP is about ten years old, and costs will continue to decrease.

In addition, if you price VoIP simply by the cost of the top-of-the-line hardware, you may draw a false conclusion. For example, a videophone that runs on VoIP is much more expensive than a low-end VoIP phone. But then again, full-motion, real-time videoconferencing would cost a lot less than flying to a distant location for a meeting.

Moreover, with VoIP, you don't need an entirely separate telephony system as you do with POTS-related telephony. VoIP

substantially reduces the costs inherent in the traditional telephony infrastructure.

Finally, look beyond the hardware and consider the services delivered. Check out your monthly telephone bill and try to understand exactly, line by line, what you are paying for. For companies with one or more locations, this is more complicated than a consumer at home, but the process is the same — and just as illuminating.

POTS Is Faster

Some say POTS is faster than VoIP, but nothing could be further from the truth. When you evaluate the speed of VoIP, you need to consider what network or network types you plan to use to run VoIP. Private, dedicated networks running VoIP have proved as fast or even faster than POTS in supporting telephony.

At the other end of the continuum, if you run VoIP as the very first Internet hobbyists did back in 1995, you are going to get slow speeds. The decisive factor here is that the public telephone network and the Internet sit between the VoIP caller and receiver. Because both ends used dialup modems and POTS lines for access, delay was inevitable.

The Quality of Service Is Suspect

The same arguments for speed can be made for QoS — specifically, what network type are we talking about? If you test the QoS of VoIP by limiting the network type to the circuit-switched PSTN, POTS QoS wins hands-down. After all, the PSTN is tuned for optimal POTS quality.

The major requirement with VoIP is that the voice signals get packetized. Consequently, the QoS with VoIP is determined by the network type. If you run your VoIP network over dedicated digital lines, you find that the QoS is just as good if not better than POTS QoS.

VoIP-Enabled Phones Are Pricey

Phones should be evaluated on the features and applications that they deliver to the customer. Taking this approach, some POTS phones cost more than VoIP phones, and vice versa. Moreover, many digital POTS phones work with a VoIP network. Home users are also using older analog phones (with a carrier-provided adapter) to run with VoIP over their broadband service.

With some VoIP phone types, you can do videoconferencing and Web surfing. You can even transform your computer into a telephone at no cost. Can a POTS telephone do all that?

VoIP Calls Can Be Intercepted

Can VoIP telephony packets on a computer network be intercepted? Yes, they can. What does it take to intercept VoIP packets? It takes the same equipment and access that it takes to intercept computer data packets. How feasible is it? Not very.

After spending millions of dollars, the FBI developed a system called Carnivore that is essentially built on earlier network management technology known as a protocol analyzer. Basically, the device (a souped-up computer) plugs into a network much like any other network addressable device. It sits there and collects packets as they race by at the speed of light. The packets can then be analyzed for threats and other information, or so the theory goes.

If you're worried about such a device, keep the following in mind:

- A government agency at least as powerful as the FBI is required to gain access to a given network (excluding a trusted person doing it).
- Access must be physical. The person must have a key to the telecommunications closet or access to an office where they can plug into the network.

- Access is achieved through the network operating system, so the person must have a network access account.
- Network managers today have a variety of techniques to protect their packetized network traffic.

After all this, if you're still concerned about VoIP packet interception and security, consider the fact that anyone on the street can tap a POTS telephone line with a simple analog handset and a few wires. All they need is physical access to your line. They do not have to be inside your company; they can access the line from the street or in tunnels where the public access lines run.

A solid argument can be made that a packetized network has more security than the older circuit-switched network, particularly because you can also implement data encryption for VoIP.

911 Calls May Not Work

Remember that the 911 network was designed to be supported by and make full use of the circuit-switched network. VoIP uses packet-switched networks. This is a colossal difference that needs to be clearly understood.

You should make sure you understand how your VoIP service provider will handle 911 calls. Early VoIP services required you to maintain local POTS telephony lines to be able to make 911 calls. However, today most service providers now include 911 calls as part of their VoIP services.

VoIP Isn't Ready for Prime Time

We are beyond considering whether VoIP is here to stay when we can point to market leaders such as XO that is carrying over 10 billion VoIP minutes each year. The problems that plagued VoIP in the 1990s have been overcome through technology or worked around with different technology.

VoIP is ready and working now.

VoIP Call Features Are Expensive

Because VoIP comes with all the usual call features you find in a traditional POTS call plan — plus several more features that are unavailable in the POTS world — many think that VoIP features add enormous costs to the monthly bill.

Adding features to your POTS line does add costs. But nothing could be further from reality when it comes to VoIP. VoIP operates with the TCP/IP protocols, which are used on the Web. As a result, VoIP features can be delivered through the software without additional costs. They are free in all private VoIP networks.

You Have to Discard Your Old Phones

In the early days of VoIP, it quickly became apparent that the typical PBX (non-VoIP) environment could not run VoIP in its then-present state. Everything about the PBX was geared for circuit-switched telephony. The PBX back then could support dedicated access lines, but only if the lines were used to channelize the bandwidth for POTS line equivalencies. Because the PBX controlled all digital phones that came with the PBX, the conclusion back in the 1990s was that the PBX and its digital phones could not run VoIP.

Today, this potential drawback to VoIP has been wiped out. Through upgrades to the PBX, including connecting the PBX to the LAN, all digital phones that work with the PBX can now run VoIP. Companies today do not need to worry about losing their investments in PBX technology.

Case Study

XO Fits Belt Maker's Need for Faster Communications at Lower Cost

When an Italian belt manufacturer was looking to trim costs and improve international communications at the same time, XO Communications provided a tailor-made solution with XOptions Flex.

Dora Belts, located in Robato, Italy, designs and manufactures private-label belts for some leading U.S. retailers. Its New York City office is kept busy communicating with company headquarters in Italy and retail clients located across the United States. That means a lot of long-distance phone calls and many e-mails, often with large files attached containing product designs and images. The challenge to XO was to facilitate all levels of communications with cost-efficient services.

Determining the Optimal Mix of Telecommunications Services

To improve Internet speed and increase efficiencies, the company needed to take a comprehensive look at its existing telecommunications services.

Alex DeLeon, president of the United States division of Dora Belts, was dissatisfied with his DSL. Its "best effort" service was not meeting his needs for reliable bandwidth.

For example, when trying to share product images and files with clients and headquarters, he often had to split a large e-mail into three or four separate e-mails. Sometimes he had to send designs by fax instead.

Likewise, telephone bills were complex and costly. His office had different carriers for local and long distance, paying up to 17 cents per minute for long distance to Italy. Costs added up quickly with regular phone calls to designers, product managers, and clients.

DeLeon had heard about VoIP as a potential alternative and began to investigate a solution that would

- Provide a faster and more reliable Internet connection, allowing his office to communicate better with clients and company headquarters.
- Bundle data and voice services together into a single, simplified solution.
- Reduce monthly telecommunications costs.

Replacing Existing Services with an Integrated VoIP Solution

In September 2005, DeLeon contacted XO Communications, his current DSL provider, to ask about other possible solutions. Working with his account manager, he found the perfect solution in XOptions Flex.

XOptions Flex included a dedicated T-1 line for Internet access, which would provide superior service over DSL. Although the advantages of the T-1 line were clear, having five voice lines included on the T-1 circuit made the solution much more attractive to DeLeon. “We’ve definitely saved money having telephones and Internet access all on a single T-1 line,” DeLeon said.

Other key features of the new system include:

- The ability to use existing telecommunications equipment already on premises

- *Dynamic bandwidth allocation,* which assigns additional bandwidth to data throughput when phone lines are idle
- An *administration portal* that enables end users to manage voice and calling features, such as having voice mails sent to an e-mail account
- The reliability and security of XO's national IP network and softswitch technology

Reaping the Rewards: It's a Cinch

XOptions Flex has delivered increased reliability and performance of Internet communications while reducing the cost and complexity of all Dora Belt's telecommunications services.

Monthly cost savings

Moving Internet access, plus all telephone and fax lines to XOptions Flex, saves Dora Belts $400 per month on telecommunications costs. The company now pays 3 cents per minute for long distance to Italy instead of 17 cents.

DeLeon said other savings are realized because his increased ability to work over the Internet has cut back on the number of trips he has to take to Italy, eliminating some travel costs.

Increased performance

Increased bandwidth means data moves faster across the Internet. And that means users no longer waste time with awkward work-arounds.

"When we send photos or designs through e-mail we notice a big difference now. We no longer have to split a single e-mail into multiple e-mails," DeLeon said. "Nor do we have to send designs by fax or messenger."

Increased productivity

Dora Belt's employees accomplish a lot more now that XOptions Flex is installed. For example, because of the increased reliability of the Internet connection, DeLeon can stay in the office and view video archives on the Internet of fashion shows and other events important to his business.

Simplified billing

Dora Belts now receives a single monthly invoice with a flat rate for voice and data services. (Flat rates vary by market; limits apply.) "It's easy to read and doesn't fluctuate much month to month," DeLeon said. "I don't have to figure out three different bills every month."

Excellent service

DeLeon said his account manager at XO Communications was very responsive and a pleasure to work with. "Many companies might look good on paper, but it comes down to the people — and the people I worked with at XO made sure I was satisfied."

Appendix

Glossary

A

add-on charges: Monthly charges beyond access costs. Add-on charges are mandated by the government and administered by the carrier. These types of add-on charges are also referred to as *regulatory fees.*

B

bandwidth allocation: Digital bandwidth can be subdivided and allocated based on channels. For example, a T-1 line with an aggregate bandwidth of 1.536 Mbps can allocate eight 64-Kbps channels to the telephone system, eight channels to the computer data network, and eight channels to a videoconferencing system. All 24 channels can be used in a dynamic allocation pool. A pool assigns channels as needed for the life of the telephone or videoconference call and returns the channels to the pool when the call or session is over.

C

calling feature: Additional uses or applications of the telephone, the telephone line, or the network that carries the telephone call. Voice mail, call forwarding, and call transfer are examples of traditional calling features. VoIP telephony has all the traditional calling features plus a new generation of features that use the telephone and the network, such as presence, vemail, and displaying a Web page on your telephone screen.

circuit-switched: The traditional method of transporting a telephone call over the PSTN. Multiple devices known as switches are employed by the carriers to form paths, or circuits, over which telephone calls may be carried between a caller and a receiver.

convergence: The integration of switched and dedicated networks to support similar applications. For example, using VoIP on the corporate computer network to place a call on the traditional public telephone network.

D

digital telephony: Started first as a method for carriers to aggregate and transport POTS telephone calls on the carrier's network using DS-type transport lines. This same technology would later be redeveloped into privately owned telephone systems (PBX) that would be owned and operated by customers.

E

Ethernet: The oldest and most popular protocol used for establishing data networks. Ethernet is used in more than 98 percent of corporate America for LAN networking. Ethernet is increasingly being used as a Metropolitan Area Network (MAN) backbone standard. The fundamentals of Ethernet are modified slightly to support WiFi and WiMax, popular forms of wireless Ethernet.

F

frame: On the LAN side of the network, bit signal traffic is encapsulated and transported inside MAC frames. See also *MAC.*

G

gateway: A network device used to provide access between different types of networks. For instance, a gateway may provide access into an external network such as the PSTN, the Internet, or a private WAN. A PSTN gateway has a LAN interface on the inside and a PRI access transport line on the outside. It translates IP telephony frames from the LAN into circuit-switched POTS traffic for the PSTN and vice versa.

H

hard phone: A VoIP-enabled telephone that has an RJ-45 LAN interface port to connect it to the Ethernet LAN. VoIP telephones today come in all shapes and sizes.

hosted VoIP: A managed VoIP telephony service similar in concept to the traditional Centrex model. Synonymous with *IP Centrex, VoIP Centrex, hosted telephony,* and *hosted VoIP telephony.*

I

in-state toll: One of four traditional regulated toll carrier service categories, also known as intrastate.

interstate toll: One of four traditional regulated toll carrier service categories, also known as *long distance.*

intralata toll: One of four traditional regulated toll carrier service categories, also known as *local toll* or *regional toll.* Refers to calls in which the caller and receiver are in the same local access and transport area (LATA).

intrastate toll: One of four traditional regulated toll carrier service categories, also known as in-state toll.

IP address: An address comprised of four numbers, each ranging from 0 to 255, and normally expressed with each number separated by a period (such as 192.168.2.100). IP addresses are used to route network traffic from sender to receiver. The IP address is a major component field of a VoIP packet and is used to map the VoIP telephone call to a specific telephone number. In a VoIP telephony call, both source and destination (caller and receiver) addresses are used to establish and maintain the VoIP call.

IP Centrex: A managed VoIP telephony service similar in concept to the traditional Centrex model. Synonymous with *VoIP Centrex, hosted VoIP, hosted telephony,* and *hosted VoIP telephony.*

IP soft phone: Software that enables a computer to function as a VoIP telephone, including an on-screen dialing pad for point-and-click dialing.

L

LAN: Local area network. A data network limited to a small geographic area. A LAN can be as small as a couple of devices connected on the same network or as large as a campus-wide installation with numerous buildings and thousands of addressable devices on the same network.

M

MAC: (1) In network terminology, an acronym for media access control. The part of the network interface that controls physical access to the LAN through the MAC address. (2) In telephone system administration terminology, an acronym for moves, adds, and changes. MAC describes the most common type of maintenance necessary in traditional telephone systems.

MAC address: An address that uniquely identifies a network device. The MAC address is typically represented in hexadecimal notation, as in 00-04-23-58-90-6E.

N

network feature: Calling features provided in a VoIP network. For example, presence, Web surfing, and vemail are network features.

NIC: Network interface card. Provides the network device, such as a computer or a VoIP telephone, with its MAC address and the means for connecting to the LAN.

O

off-net: In VoIP telephony, refers to calls that must be carried on another network (usually the PSTN) external to the VoIP network.

on-net: In VoIP telephony, refers to calls carried on the customer's network.

P

packet: On the WAN side of the network, bit-signal traffic is encapsulated and transported inside packets. A packet can be best visualized as an electronic envelope for transmitting data.

packet-switched: Packet-switched networks such as a VoIP network use the addressing information contained in the packet to determine the route the packet takes to its destination.

POTS: Plain old telephone service. The most basic form of circuit-switched telephone service.

presence: One of the network features available with VoIP telephony services through a light indicator or software icon. If the presence light is on, the person is available on the network.

PRI: Primary rate interface. An ISDN transport line providing 23 B channels and one 64-Kbps D channel. The PRI has gained renewed popularity with the advent of VoIP networks that use gateways with PRIs to support Off-Net calls to the PSTN.

S

SIP: Session initiation protocol. An interoperable protocol in the TCP/IP family of protocols. SIP uses text formatting to set up and maintain communication sessions with various endpoints. These endpoints can include cell phones, desk phones, PC clients, and PDAs. SIP permits these various endpoints to operate as a single system.

T

T1: The standard for 24 DS0 channels having an aggregate bandwidth of 1.536 Mbps. A T1 line is also commonly known as a *DS1 line.*

TCP/IP: Transmission control protocol/Internet protocol. The family of interoperable protocols consisting of more than one-hundred-twenty protocols, each of which performs one or more services to support various network applications. The

early developers of the Internet agreed upon the name TCP/IP because, at the time, TCP and IP were considered the two most important protocols for any network connection.

toll bypass: A term that concisely describes how VoIP telephony service completely sidesteps the regulated, circuit-switched PSTN and all its associated toll usage charges by carrying telephone calls over private, packet-switched networks.

triple play: Refers to the capability to integrate data, voice, and video applications on the same transport.

U

UDP: User datagram protocol, an encoding protocol implemented at the transport layer of VoIP telephony and videoconferencing calls.

V

vemail: A network feature supported with VoIP telephony in which the user can elect to hear their e-mail or print a hard-copy of their voice mail.

voice mail: A popular calling feature that allows callers to leave a message in the event that the called party can't answer the call. Voice mail comes at no additional cost with VoIP telephony.

VoIP: Voice over Internet protocol. A network service that supports carrying telephone calls over packetized networks. VoIP reduces substantially or eliminates the need for a separate, circuit-switched telephone network to carry telephone calls.

VoIP Solutions from XO®

The XO IP-portfolio encompasses solutions for businesses of all sizes and at any stage of the VoIP evolution.

Product	Description	Best Fit	Capital Costs	IP Expertise	Flexibility
XOptions Flex®	Lets you combine voice and data traffic on a single circuit, while keeping the phone equipment you already have, and managing your services through an Internet portal.	The ideal entry-level VoIP solution, lets you keep your existing phone equipment.	☑ Low ☐ Medium ☐ High	☑ Low ☐ Medium ☐ High	☐ Low ☑ Medium ☐ High
Hosted IP-PBX **Available 1st Quarter 2007**	Provides the benefits using IP-phones, and an IP-enabled PBX, without the need to buy the PBX. Instead, the functions normally associated with a PBX are "hosted" in the service provider's network. Customers access and control their features through an Internet portal.	Businesses ready to invest in IP-phones, but who want to avoid the investment in a new IP-PBX.	☐ Low ☑ Medium ☐ High	☑ Low ☐ Medium ☐ High	☐ Low ☑ Medium ☐ High
Managed IP-PBX	The phone company provides connections to the Internet and Public Telephone Network, along with a state-of-the-art IP-enabled PBX, and IP-phones that are installed, and managed by the service provider. Customers pay a monthly fee that includes network access and the cost of installing, managing, maintaining and replacing the equipment.	Ideal for customers facing replacement of an existing PBX, who would like to avoid the capital costs of replacing expensive phone equipment.	☑ Low ☐ Medium ☐ High	☐ Low ☑ Medium ☐ High	☐ Low ☑ Medium ☐ High
SIP Trunks	Session Initiation Protocol (SIP) trunks are used in lieu of traditional telephone digital trunks when the customer owns and manages a premised-based IP-enabled PBX that combines voice and data services.	Suited for more sophisticated users who plan to purchase and manage their own IP-based phone equipment, and want maximum control of their equipment and services.	☐ Low ☐ Medium ☑ High	☐ Low ☐ Medium ☑ High	☐ Low ☐ Medium ☑ High

JOIN THE XO EVOLUTION

Does your current phone system have the future on hold?

Business communications is evolving and XO Communications is leading the way with the next generation of IP services. Our services range from local and long distance to Internet access, private line, Ethernet services, VPN, web hosting and more.

A leader in IP-based services, XO has been providing communications services to businesses for nearly ten years. We currently carry nearly 10 billion VoIP minutes per year. So don't let your business communications become extinct. Evolve to the next generation of IP-based solutions with XO Communications.

JOIN THE XO EVOLUTION

XO Communications
www.xo.com

Put your phone system on your computer network for maximum efficiency

See how to get started with VoIP, how it works, and why it saves you money

VoIP is techspeak for "Voice over Internet Protocol," but it could also spell "saving big bucks" for your business! This handy guide provides the scoop on the hottest advance in telecommunications in plain English. Find out how voice communications can travel online and how to choose the best way to integrate VoIP with your company's existing systems. This jargon-free approach helps IT managers understand the inner workings of VoIP and explains the benefits of VoIP features for end-users. Real-world examples throughout show how XO Communications can help companies use existing equipment while enjoying the benefits of the latest VoIP innovations.

Discover how to:

- *Use VoIP for your business phone service*
- *Simplify network management with converged networks*
- *Increase productivity with VoIP features*
- *Demonstrate VoIP's advantages to management*

THE DUMMIES WAY®

- *Explanations in plain English*
- *"Get in, get out" information*
- *Icons and other navigational aids*
- *Top ten lists*
- *A dash of humor and fun*

Get smart! @www.dummies.com

- *Find listings of all our books*
- *Choose from many different subject categories*
- *Sign up for eTips at etips.dummies.com*

ISBN: 0-470-08640-8
Not resaleable

For Dummies®
A Branded Imprint of
WILEY