IP Telephony
Contact Centers
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WHITE PAPER

Understanding VoIP

Leveraging Technology for a Competitive Edge

October 2005

This white paper shows how converging your traditional voice and data networks can save money and increase efficiency and productivity throughout your organization – just what it takes to remain competitive in today's marketplace.

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." Internet Protocol (IP) networking is supported by any type of network — corporate, private, public, cable, and even wireless networks—not just the Internet. The corporate sector usually prefers private dedicated networks; telecommuters and home-users generally favor broadband.

What does this mean in terms of your actual telephone? You can access your account on the VoIP network via a desktop telephone, a wireless *IP phone* (similar to a cell phone), or the soft screen dial pad of your laptop or desktop computer. With this setup, you can literally pick up your things and move to another location without having to forward your calls to a new telephone. What's more, you can access the Web from your IP phone, enabling you to get announcements and e-mail on the go. 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.

This paper provides an excellent starting place for managers who are looking to make the switch to VoIP or need to make an informed decision about integrated networking, as well as end users who are new to VoIP. It explains how VoIP works and how it compares to telecommunications technology that was previously considered to be irreplaceable. It demonstrates why many businesses throughout the world have turned to VoIP and integrated networking as their main system for data, voice, and video transfer.

The Avaya Advantage

The IP Telephony market is populated by many competitors, but only one stands out as the clear market leader: Avaya. To understand what it means to lead the IP Telephony market, you need to understand the significance of converging your traditional telephony systems onto your computer network to create an enterprise-level communications network. And you don't have to throw out your investments made in other telephone systems' hardware. You can do it the Avaya way, avoid forklift upgrades and reap the benefits of all of the new IP-based features and functionality available right now in the IP Telephony world.

IP Telephony the Avaya way focuses on using your existing telephony resources to build a vibrant communications network that enhances productivity. An Avaya system includes all the features you are familiar with—voice mail, call waiting, and call forwarding, to name a few—as well as many new exciting features, such as the Presence feature, which indicates at a glance whether coworkers are available to take your call, and the "follow me" feature, which lets your telephone number ring at whatever location you may be currently at on your company's network, whether you are in the same city or across the country.

Customers need reliability, and Avaya delivers it. If you are considering a converged network, you are probably eager to gain a simple-to-manage, business-driven architecture at a cost that is competitive with your current expenditures. Avaya can help you accomplish this goal -- they do it today in over 1 million companies around the world. And it supports over 90% of the FORTUNE 500®. The prestigious Gartner Inc. research and advisory firm listed Avaya in the Leaders Quadrant in the 2005 Magic Quadrant Report for North American Corporate Telephony. Focusing on enterprise migration to IP Telephony, the report positions vendors into one of four quadrants—Leaders, Challengers, Visionaries, and Niche Players—based on the companies' vision and ability to execute on that vision. According to Gartner, vendors listed in the Leaders quadrant are performing well today, have a clear vision of market direction, and are actively building competencies to sustain their leadership position in the market. Avaya is one of those Leaders.

Bottom line: You don't have to do it alone; the expertise of Avaya Global Services will deliver the results you are looking for with its extensive portfolio of professional services. Converge, communicate, and compete by putting communications at the heart of your business. Explore the possibilities with Avaya. Visit **www.avaya.com** to find out more.

Part I: Getting Down to Business with VoIP

Technological innovation is upon us once again. This time it is coming in the form of improving the way we make telephone (voice) calls, bringing with it several new capabilities that change the meaning of the phrase *telephone call.* VoIP is the name of this new communications technology. *VoIP* stands for *Voice over Internet Protocol,* which 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 make possible enhanced voice communications throughout the enterprise. 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.

How Does VoIP Work?

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. Today, VoIP operates over most network types, including those used throughout the corporate sector. *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 companies' needs for LANs.

Circuit-switched telephony

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

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.

Packet-switched telephony: From POTS to packets

Unlike circuit-switched POTS, which always require 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 signal; routing directions are determined physically by the actual POTS line. So the routing must follow a specific network line similarly to how a train follows a designated set of railroad tracks. If the line is down, the call cannot go through. In a packet-switched network, if one of the network lines 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 have begun to call it for short, 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 network types. 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 you can 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.

IP Telephony

IP Telephony enables voice communication over Internet Protocol (IP) networks. It unites an organization's many locations—including mobile workers—into a single converged network. It promises cost savings by combining voice and data on one network that can be centrally maintained. But more importantly, it brings advanced features and applications that enhance productivity throughout the organization.

A large percentage of calling patterns within corporations indicates that many calls will never leave the LAN. An IP Telephony call to a coworker at the same location would never leave the LAN. Similarly, a call to another department in your building would never leave the LAN. In these examples, the packets remain simply packets that travel over the LAN to the called person. These packets do not need to include other vital data inside them to direct the packet over longer distances to other locations. As a result, IP Telephony calls result in shorter-sized packets when compared to VoIP calling packets that would need to go off the LAN.

IP Telephony works a lot like the computer works on the LAN. But all users do not necessarily need an IP-enabled telephone. Avaya implements IP Telephony in a manner that can protect your investment in pre-existing telephone equipment (for example, enabling IP to digital and even IP to analog telephone stations).

You can acquire IP telephones in several different styles. But they all have one important thing in common: The IP telephone has a network interface card (NIC) built into it just like a computer must have a NIC inside of it to connect to the LAN. The NIC is the single most important component for any LAN device because it provides the device with its physical address on the LAN. This address is simply called the *MAC address.* MAC means *media access control.* The MAC address uses a standardized 6-OCTET address and is usually represented in hexadecimal. For example, the following is a valid MAC address: 00-0A-E4-02-7B-99.

To support IP Telephony, a server with a MAC address is typically dedicated to load the IPT software that is used to manage all the calls. Servers are just like personal computers except they have more memory, speed, and capacity. Because the server needs a MAC address, it has a NIC inside to provide the MAC address as well as a physical means of connection to the LAN. The managing server stores the database that contains all the MAC addresses, corresponding to all of the IP Telephony telephone extensions that will be assigned to end-users.

Depending on the size of the LAN and the number of users, other servers may be used. For example, some LAN running IP Telephony dedicate a server just for handling all the voicemail that will be stored and retrieved. Depending on the size of the LAN, one or more devices known as *switches* are installed around the LAN to form the core infrastructure of the IP Telephony LAN. These switches are boxes that have a series of ports into which all of the other LAN-addressable devices ultimately connect. Usually the switches are set up in the communications closets around the LAN, and they operate 24 x 7 x 365. All of the cabling typically runs from the user devices (such as the IP Telephony phone, computer, and servers) to the ports on these switches. Older switches supported computer data only.

If you plan to run IP Telephony with your computer data on the same LAN, make sure that you use IPT-compatible switches. As with any addressable device on the LAN, the switches used must also be MAC-addressable. All the other addressable devices, including your IP Telephony phone, must physically connect to the LAN via a port on one of the switches. All of the switches are interconnected, usually with fiber-optic cable. 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 are calling. The packet is then forwarded to the switch, the port on that switch, and finally to the IP telephone connected to the port corresponding to the coworker's MAC address. 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 of this at lightning speed.

The process varies a little when you place a call to a coworker located at a different site. This coworker is connected to a different LAN. The call would still initiate the same way. But instead of the managing server sending it to a switch located on your LAN, the call needs to go to the edge device used to connect your IPT LAN to the company's WAN. This is where IPT becomes VoIP. And this is where that second type of network comes into the picture.

Reaching out with VoIP

From the individual mobile end-user and small single-site LAN to the sophisticated, multi-location WAN that supports domestic and international connectivity, Avaya is a world leader in secure and reliable IP Telephony systems, communications software applications, and full life-cycle services.

Each LAN in a multi-location enterprise network is connected to the larger WAN. If you are located at the headquarters in one city, and you call a coworker located at the office in another city, your call begins as an IP Telephony call on your LAN. It then travels from your LAN through an edge device. Edge devices include products such as the Extreme Networks' Unified Access enabled switch. The edge device is programmed to re-packetize your call and encode the larger VoIP packet with the additional necessary information such as the address for the destination LAN or the mobile end-user. For a single location company, other options for the edge include using the Avaya G650 Media Gateway which connects directly to the PSTN.

The process of packetization is referred to as *encapsulation* by the network gurus. It is similar to 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. You would not be wrong to call it *Voice Signals Inside IP Packets.*

In order for the LAN to participate in the company's VoIP WAN, each LAN needs at least one edge device such as a router, level three switch, or a gateway. These devices, like all other addressable devices on the LAN, have a MAC address and a NIC to physically connect them to the LAN. But in addition, they each have an interface card that physically connects them to the company's WAN or some external network. Depending on the company's network design, size, and mission, these edge devices can have multiple interfaces that connect them to multiple outside networks. Or you can have multiple separate edge devices. These 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 destination LAN.

Unlike the MAC addressing on the LAN side, VoIP traffic on the WAN uses the IP addressing scheme. IP addressing currently uses a 4-OCTET format and represents the address in decimal numbers. For example, 192.168.2.4 is a valid IP address.

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

Instead of maintaining separate networks for computers and telephones, companies can converge both of these networks into one network using IP Telephony and VoIP. The whole new way of doing telephone calling using VoIP is even sometimes called *toll bypass*.

Gaining Flexibility with VoIP

VoIP is a win-win for everyone. With VoIP, customer satisfaction and productivity increases for your entire company.

A few VoIP features, such as voicemail 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. As a result, many new features under IP Telephony have become available.

As with any new technology tool, VoIP with all of its many end-user benefits is quickly replacing traditional POTS alternative technologies. Indeed, VoIP is even becoming a superior replacement for many former computer-only applications.

Say hello to VEMAIL

Before IP Telephony and VoIP, you had to access your voicemail by telephone and e-mail through computers. With VoIP, you can read your voicemail on your computer screen and listen to your e-mail through an IP-enabled telephone. The new term for this converged feature is *VEMAIL*. And this is just one of many new uses for IP-enabled telephony.

Surf the Web

Because VoIP operates with the same set of IP rules and protocols that support Web-based applications, it is possible to access the Web with an IP-enabled telephone.

Several varieties of IP telephones are now available. Some IP telephones have a large enough screen that you can display any Web page on your telephone, including a stock exchange ticker displaying your favorite stock symbols and current trading status or the current weather for your geographic area of choice. If you compare the IP-enabled telephone with any of the POTS telephones, you find a remarkable difference in the quantity and quality of features available with IP-enabled telephones. In an IP Telephony world, the features are available with no monthly recurring charges. Considering that many new features have come about with IP Telephony and VoIP, you'll want to consider by IP telephone type exactly what features are available.

Getting the most from your IP telephone

You can use an IP-enabled browser phone such as the Avaya 4630SW IP Screenphone to accomplish much of what you can do on your desktop computer. For example, Joann works for one of the top healthcare insurance providers headquartered in the Northeast. Her company has 17 locations connected over a VoIP supported Wide Area Network (WAN).Throughout her typical day, Joann uses an IP-enabled browser telephone to receive announcements, make phone calls, and send and receive e-mail.

Joann starts her day by checking her IP telephone's Web page for announcements. One morning, she read that her friend and coworker Rae Lynn had a baby boy the night before. She made a note to send Rae Lynn's family a card.

As part of her job, Joann reviews and approves/denies healthcare claims that do not fit the normal criteria for a final decision by the utilization review (UR) department. Much of Joann's communications relate to the status of the claims she is investigating. She regularly communicates with people located at her home office and other sites, usually the site of the claim(s) IP origin. Joann also interacts with staff from their company's huge healthcare provider network to determine the fine details of each claim she receives for disposition.

With the exception of any calls made in the local calling area, all Joann's telephone calls are carried on the corporate VoIP network. When the call is to a provider located off-net near one of the company's other locations, the call travels from Joann's IP telephone over the corporate VoIP network to the distant site's location where it goes over the company's LAN at that location, out the gateway there, and into the local calling area of that location. As a result, for all Joann's telephone calling, her resulting, monthly, off-net, billable telephony charges are minimal and for the most part are billed as local calls.

All the claims Joann's company receives are transmitted to their UR department via the Web. If a claim cannot be approved for payment upon receipt, the UR department forwards it electronically via the corporate VoIP network to Joann, with a copy to the medical director of the respective source location and a copy to the headquarters' medical director.

Joann works frequently with the medical director at the headquarters' location because of the technical nature of many of the claims she receives. On average, Joann calls this medical director 7 to 10 times per day on claim related matters. Therefore, she includes this medical director in her I telephone buddy group and makes full use of the "presence" feature alert indicator on her IP telephone. If the presence indicator is lit, she knows not to waste her time calling the medical director because he is on the telephone with someone else. Joann also has a presence indicator set up for her immediate report and the coworker that must fill in for her (and vice versa) when she is not at her desk.

Much of Joann's day is spent on her IP telephone. She uses it to process inbound or outbound e-mail from the company's various locations. Sometimes the content of a claim requires Joann to contact other personnel in the company. When she needs to do this, Joann accesses her browser based directory to retrieve the person's contact information and automatically dial their IP telephone number. Or, if Joann is on the road, she can use the Avaya Speech Access application feature to have the system automatically dial anyone in the directory by simply speaking their name.

Needless to say, Joann is a busy woman. About 30 minutes before her workday ends, she checks the weather advisory corner of the Webpage on her IP telephone. She wants to know whether she needs to bring her umbrella when she heads over to the subway station. She checks her voicemail and typically opts to have the remaining unheard voicemail messages printed so she can read them on the subway ride home.

Calling All Phones

With an Avaya approach to IP Telephony, companies can use their existing digital telephone station equipment to avoid forklift upgrades and be more selective and cost-effective with IP telephone deployment. Though telephone designs may vary from those on employees' desks to those in hallways or meeting rooms, any digital phone can support IP-based telephone calls on the LAN side, packetized VoIP-based calls on the WAN side, and local calls off the LAN and into the PSTN as needed.

In addition, digital and IP-based telephones are differentiated by the number and type of features they can support. Features that have been available on most digital telephone station equipment prior to the emergence of IP Telephony and VoIP include:

- Voicemail
- Call transfer
- Call forwarding
- Call waiting (also known as *call park* or *hold*)
- Multiple call appearances
- Three-way (or more) conference calling
- Redial
- Speed dial
- Message indicator to let them know they have voicemail

Migrating to IP Telephony does not have to mean that you replace digital telephone station equipment to keep these features. This equipment and its feature functions are interoperable in the new IP Telephony environment.

The Avaya approach to IP Telephony builds on existing feature sets by adding IP-based features and functions that transform the enterprise's infrastructure into a converged communications network. The added features and functions include:

- Employees connect their IP telephone into the company's LAN. In addition, they connect their computer into one of the ports on the IP telephone. In a startup company with no existing cabling plant, this reduces by one-half the number of cabling drops needed to physically connect all employees to the LAN. This feature can add up to thousands of dollars of savings for just one building or location. It also reduces the complexity of the company's cabling plant. The lower-end IP telephone types have just a single port to connect the phone itself.

- In the POTS world, the carrier companies provide the power from their equipment over their circuit-switched lines to the telephone. In companies using the conventional private telephone systems, the system (PBX, or Private Branch Exchange) provided the power to the telephones in the company. With quality IP telephones, the power is down line loaded from the LAN switch or the IP telephone can be plugged in at the user's desk.

- All IP telephones support the IP family of protocols (at least to a certain extent), so they are generally compatible with the Web. Not all IP-enabled telephones are physically able to support the full range of Web applications, but IP-enabled telephones that have HTML-based displays can support most Web-related enterprise applications.(HTML stands for *Hypertext Markup Language* and is the main programming language used to program Web pages.) IP-enabled telephones with HTML-based displays support the following features:

  – Dashboards (lights and indicators on the Web page area of your phone)

  – Web browsing

  – Corporate news and events

  – Weather advisory display

  – Employee productivity

  – Stock ticker

  – Support for end-user defined applications and links

Other special features on selected IP-enabled telephones include:

– Security alerts

– Access to corporate directory information via industry standard Lightweight Directory Access Protocol (LDAP) server

– Personalized ring patterns

– UNICODE support for native language display information.(UNICODE is a 16-bit code that translates every character of every language in the world.)

– Call log lists of incoming and outgoing calls

– Integrated speakerphone

– Infrared port for PDA and PC application integration

– Multiple call appearances

IP SoftPhone

The Avaya IP SoftPhone is designed to operate on a Windows based computer. It is essentially a piece of software to be loaded on the computer. Imagine dialing a telephone number from your computer by pointing and clicking the on-screen dial pad. Or as an alternative to clicking, you can simply say aloud the speed dial name. Together with a pluggable telephony headset, it enables voice communications and other productivity features, including:

- Clear voice communications possible from alternate work locations including home, hotels, Internet hotspots, and customer locations
- Integration with Outlook contact lists for autodial support
- LDAP directory access
- Phone numbers displayed
- Incoming calls synchronized with directory look-ups for simple screen pop applications
- Instant messaging and presence tracking
- Point-and-click dialing
- IP-enabled desktop telephone features accessible from computer
- Multiple call appearances (as opposed to just the one call you are currently on)
- Point-to-point video calls application (available with selected versions)

Saving money with IP SoftPhone

Larry is a Human Resources Specialist who works for a kitchenware manufacturer. His home office is located at the company's headquarters, but Larry frequently travels among the company's 23 locations, which include plants and sales offices spread out across the country. He conducts interviews with new employee candidates, including screening and second-round interviews.

Larry's company used to pay enormous toll charges. The largest toll services billing came from their regional toll services. VoIP eliminated an enormous 92 percent of the company's toll charges. Every time Larry traveled to a different company location, the IT staff would set him up with a computer network connection and a telephone. Because the company went to VoIP, Larry merely has to plug in his computer, which runs IP Softphone at any available port. He does not need to have anyone from IT make special configuration changes for him. He can even reroute his extension using IP SoftPhone to any phone in the office. The phone does not even have to be an IP phone.

Moreover, with the new version of IP Softphone, Larry can make use of videoconferencing from his laptop computer. He no longer needs to travel to other sites to conduct screening interviews. The candidates report to the company location nearest them, and the sponsoring location allows the interview candidate to use one of their stations that similarly runs videoconferencing. Through VoIP, Larry's company saves in toll charges and travel costs.

IP SoftPhone for Pocket PC

The Avaya IP SoftPhone for Pocket PC is designed to connect a Pocket PC device to the company's IP Telephony environment — and the best part is that it does so wirelessly. The IP SoftPhone for Pocket PC is essentially a piece of software that you load on the Pocket PC device. The software connects the Pocket PC (which must be Windows-based) via a Wi-Fi or Wi-MAX network interface to the company's LAN. It enables voice communications and other productivity features, including:

- Multiple call appearances
- Call transfer
- Conference calls
- Point-and-click dialing from directories
- Outlook contact lists

IP SoftPhone on the go

Whenever Larry goes to lunch, he carries his IP-enabled Pocket PC. The Pocket PC is a wireless telephone that gives him full mobile desktop capability with standard off-the-shelf Pocket devices and standard Wi-Fi or Wi-MAX Ethernet interfaces. It provides Larry with increased mobility options and value — with no specialized hardware required.

Larry can receive calls virtually anywhere. The Avaya IP SoftPhone for Pocket PC is perfect for Larry's combined need to be mobile and accessible by phone.

Whether Larry is on the road, at one of their warehouse locations, at a sales office, or just in a meeting room down the hall, IP SoftPhone for Pocket PC helps him stay connected, responsive, and productive.

Part II: Simplified Management 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 (VoIP does work over the Internet, but the quality of service for a mid- to large-size company is not the same or even close to what can be obtained from a dedicated private network.)

Even though VoIP in 1995 was not ready for primetime, 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 call features include all of the features offered in POTS-related telephony systems, plus many additional features which have never been seen before in telephony systems. VoIP features enhance the collaboration of employees across the enterprise and ultimately reduces the operating expenses of the company.

Saving Money with VoIP

One of the major benefits of VoIP is that companies can enjoy an immediate cost-benefit with their regional toll and long distance voice and videoconferencing charges. Prior to 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 to 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 a small amount, but when you add up all the minutes from every line in operation the cost each month can frequently add up to hundreds of thousands of dollars. Especially for companies with multiple locations that cross intralata boundaries within the same state.

Some say that VoIP does not really save a company much on toll charges. Usually they are not considering the hidden recurring cost factor: the intralata, regional toll (also known as *local toll*) monthly charges. 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 extra regulatory fees

There are other POTS-PSTN related monthly regulatory fees. These are charges that go to various government entities versus to the Local Exchange Carrier (LEC). These fees are based on a percentage of each line's monthly access cost. They include:

- The Federal Line Surcharge
- 911 fee
- Other charges depending on the location of the POTS line

These charges are based on a percentage of the monthly per line access cost, but before you draw any conclusions about these costs being nominal, add up the number of lines and the total cost. Depending on where all of your locations are located (that is, which LATAs), these regulated fees vary somewhat. But if you add up the total line access costs and take about 4 to 7 percent of your total monthly access costs, you can get a close estimate.

With VoIP, you pay regulatory fees for your dedicated network transports, but you already pay these in support of your computer data network. VoIP runs over your packetized computer network, so you have no more added regulatory costs for VoIP telephony or videoconferencing.

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 are added uses of the POTS line beyond simply making telephone calls. These features include options such as voicemail, call transfer, and call forwarding. Sometimes these features are priced out individually 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 out based on the individual line. If your company has hundreds or thousands of lines, the overall cost for all features for all lines can be astronomical.

Traditional telephony system models

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 reduces greatly 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. Here's how they work:

- **POTS:** Companies that have fewer than 15 phone stations and are not bothered by high regional and long-distance toll charges can stay with the POTS line model. With the POTS 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 *wire line 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. Prior to 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 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:

  – The PBX can use dedicated high-bandwidth lines out to the carrier or to other locations on the company's network.

  – Interfaces can be used on the PBX to provide full motion video conferencing.

  – The PBX has extensive Call Management capabilities and the capacity for setting up and controlling multiple call centers.

By using their own system, companies 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 could be integrated. Although companies could 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 multiple millions of dollars per month.

See Table 1 for a summary of the four traditional telephony systems models.

Table 1 The Four Traditional Telephony System Models

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's company KTS switch.	Monthly recurring charges (MRC) per line, start up costs of KTS and phones. Regulatory fees apply to access line costs.	Most features 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 either 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, monthly recurring charges (MRC) per line. Regulatory fees apply to access line costs.	Little or no maintenance costs, higher priced lines compared to POTS, suitable for VoIP if company has substantial MRC's for either regional, intrastate, or interstate toll carrier services.
PBX	Dedicated carrier lines to customer's PBX	Dedicated access lines. Highest MRC line charges. Dedicated amount of bandwidth. Regulatory fees apply to access line costs.	All POTS call features 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 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, the company 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 the time it takes to satisfy customers is reduced. 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. This better service translates into more revenue for the supplying company. 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.

Another major benefit is that 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 may hear the term for VoIP known as *toll bypass.* The PSTN is not involved at all in any such calls.

This capability 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. At that LAN, to a gateway device that is attached to the LAN on the inside and attached to the PSTN on the outside. Through the 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.

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 technology 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-savings move and a productivity enhancing strategy. If you cannot consider it for the huge 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.

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 considering it. No doubt some companies will have to go into Chapter 11 before they realize the error of their strategic planning.

Most companies are reluctant to change systems simply because they are comfortable with what they have. 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 drive is required to consider, understand, and set 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. Often 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 can speak to upper-level managers is 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.

A seamless transition

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 while still having 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 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 positioning 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 if used strengthens the company's infrastructure. And, unification of the respective support staffs ensures that the company can succeed in operating in the forthcoming converged marketplace.

Meeting 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 also 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 toward 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 are needed for a specific application that is seeking 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 system can make changes to support growth and increase access to/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.

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-locations, multi-LAN Wide Area Network (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 rather than a frustrating, time-consuming hassle.

Moves, Adds, and Changes (MAC) costs

As a result of the high degree of scalability and the intuitive intelligence of VoIP networks, MAC changes are a thing of the past. In companies where they still operate under one or more of the traditional telephony systems, many companies still must pay for expensive MAC changes every time an employee needs to move, 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 at $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.

Part III: Three Phases to VoIP Migration

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. The Avaya three-phase approach to VoIP converged communications is your solution.

Migrating to Converged Communication

Avaya 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

[2] Results are highly dependant on individual operating environments. Different implementation methodologies, assumptions, processes, and objectives may contribute to lower or higher results.

Where everyone starts: The Traditional Phase

Enterprises 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 enterprise as a whole has a private, dedicated WAN running IP protocols for computer data. If an enterprise 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.

Telephone system needs are typically met through one or some combination of one 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, are about as good as it can get.

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 an enterprise 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.

Grant Thornton teams up with Avaya

With 50 U.S. offices and $459 million in annual revenue, Grant Thornton LLP is the largest domestic accounting firm serving public and private middle-market clients. With the major geographic expansion for many Grant Thornton clients in the 1990s, it became clear that if Grant Thornton was to continue to provide premium service to its customers while remaining responsible to its bottom line, it needed to adjust its communications systems to be in step with the increasingly collaborative and mobile nature of the business.

Grant Thornton assigned a team to research the best way to modernize the communications system. After several months of testing different solutions and assessing vendors' support capabilities, the team recommended a multi-phase program to completely transform Grant Thornton's approach to communications.

The first step was to establish uniform voice and data infrastructures, and help ensure all Grant Thornton employees had access to basic applications like full-featured voicemail. As their network standards, the team chose Avaya as the sole provider for their voice infrastructure, and Cisco for the backbone routers and data switches. Each office had its own IP-capable Avaya DEFINITY® voice server. Full-featured voicemail was provided by 26 Avaya Octel® Messaging Systems. Because the team intended to consolidate the voicemail platform in a later phase, they elected to lease the systems to provide migration flexibility.

Here is a brief summary of the phases of the conversion that followed:

Implement an efficient architecture. With the help of Avaya, Grant Thornton converted their network transport infrastructure from inefficient and costly dependence on point-to-point circuits to a star configuration which provides fault-isolation and maximizes bandwidth allocation, resulting in considerable savings.

Introduce high-value applications. The team incorporated applications such as Avaya Phonetic Operator, which implements a toll-free number from which clients can easily and immediately be transferred to any office or team member through spoken commands.

Deploy IP Telephony. The team introduced IP trunking between all Grant Thornton's U.S. sites. First they IP enabled all 50 of the Avaya PBXs so that the traffic between GT offices would go over the network and completely avoid long distance or toll charges. Then they opened a new state-of-the-art data center in Oakbrook, Illinois, and installed an Avaya S8700 IP Media Server to provide the IP Telephony application. They deployed an Avaya Data Switch for the interface to the LAN, which interoperates perfectly with the Cisco backbone. Because the data center was a completely new location, all Oakbrook employees received IP phones that plug right into the LAN.

This scenario underscores one of the economic benefits of having an Avaya IP Telephony solution. Had the data center been an existing site with a mix of digital and analog sets already in place, the Grant Thornton team could have easily re-used them, or supplemented them with as many IP phones as were needed. Kevin Lopez, National Telecommunications Manager for Grant Thornton, says, "The fact that Avaya engineers their systems for maximum reusability means solid investment protection for us." "The financial paybacks extend even past the relatively low cost of implementation. Grant Thornton experienced big reductions in operating expense. All told, Grant Thornton is saving nearly $170,000 a year in lease and support costs. And the savings in this area will only increase as the system is further modernized.

Because all traffic between the offices goes through the new hubs, Grant Thornton was also able to take down the majority of dedicated point-to-point circuits that had interconnected most of the offices. The savings across 50 offices was a tremendous $300,000 per year! And with long-distance expenses eliminated from interoffice calls, Grant Thornton saves another $30,000 per month.

All in all, the investment reduced direct costs to the firm, provided measurably improved efficiencies, and improved the delivery of professional services to Grant Thornton's clients. The Avaya solution was an across-the-boards winner.

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 leads 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.

Avaya is taking the lead in modularization of its software and systems into open communications architecture to help organizations smoothly transition to converged communications for a more adaptive enterprise.

Mindpearl partners with Avaya

In the airline business, efficiency is everything, and top-notch customer service is the ticket to competitive advantage. In order to deliver both on a global scale, Mindpearl enlisted the help of Avaya Global Services.

Mindpearl operates five global contact centers in 22 languages to support Europe's biggest airline alliance, The Qualiflyer Group, serving 300 million passengers a year. It's a transcontinental challenge that requires top-flight technology and world-class network management. Avaya offered the perfect solution: a comprehensive outsource solution that includes around-the-clock remote network monitoring, fault and performance management, on-site technical support, and an Avaya engagement manager that provides a single point of accountability for all technical support issues.

The contact centers are powered by Avaya MultiVantage™ Communications Applications and an Avaya DEFINITY® Server, all designed, implemented, managed, and maintained by Avaya Global Services.

Now Mindpearl has throttled back on network management costs and rerouted internal IT staff to focus on the more profitable business of serving client airlines and giving airline travelers an upgrade to first-class customer care.

Session Initiation Protocol (SIP)

For Avaya, Session Initiation Protocol (SIP) is a catalyst for the next phase of open communications using not only IP Telephony and VoIP, but the full suite of IP-related protocols. SIP is an interoperable protocol in a multi-vendor environment that enables mobility and systems flexibility in multi-service networks.

A user with multiple endpoint devices such as a cell phone, desk phone, PC client, and PDA can rely on SIP to permit such devices to operate as a single system to meet changing needs for real-time communications. SIP brings about increased efficiency and productivity. SIP provides a practical means of multi-vendor integration at the highest and most diverse communication levels.

In a VoIP converged network with SIP, organizations can pick the best of breed from a variety of vendors to create a seamless converged communication network.

Avaya implements SIP through its Communication Manager product. SIP "trunking" functionality will be available on any of the Avaya media servers (S8300, S8500, or S8700). *Trunking* is making a network line support a specific protocol. A POTS trunk, for example, supports Plain Old Telephone Services. By means of having SIP-enabled endpoints controlled by Communication Manager, many features can be extended to these endpoints. The media servers can function as POTS gateways and support analog; H.323 stations; and analog, digital, or IP trunks.

SIP integrates with traditional circuit-switched interfaces and IP-switched interfaces. This integration allows the user to evolve easily from the traditional circuit-switched telephony infrastructures to next generation IP infrastructures. As a result, you don't have to use a "light switch" approach to migrate to VoIP. A reasonable migration plan can be implemented that optimizes support for the organization's business needs.

Part IV: Top Ten Reasons to Switch to VoIP

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.

Strategic Direction of VoIP Carriers and Vendors

Over the next few years, much of the $300 billion per year telecommunications industry will be migrating its equipment and carrier services to support IP Telephony on the LAN side and packetized VoIP services on the WAN. It will not be long before the current conventional 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 will become non-competitive. As a result, IP Telephony and VoIP networking technology has become today the strongest influencer in the telecommunications provider marketplace.

Avaya has emerged as a worldwide leader in secure and reliable IP telephony systems, communications software applications, and full life-cycle services. The Avaya leading role relates to many innovations and industry differentiators. These include:

- VoIP migration strategy that leverages existing and new networks to protect traditional systems investments and avoids "forklift upgrades."
- Communication network market share penetration that includes over 90 percent of today's FORTUNE 500 companies.
- Expertise and core product line that migrates and builds converged communications networks versus computer networks that happen to run voice applications.
- Industry leader in the product and features roll-out of Session Initiation Protocol (SIP) related applications.

Over one million businesses worldwide rely on Avaya solutions and services to enhance value, improve productivity, and gain competitive advantage.

Feature-Rich Cost-Effective Alternatives

Most traditional POTS-PSTN telephony calling features have made their mark on the industry. They have become familiar to all of us. Leading the charge are features such as voicemail, call transfer, call forwarding, and three-way calling. 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 or as part of a bundle based on the individual line to the carrier company. IP telephony and VoIP clearly make the "wire line service" related features out-of-date. All the traditional telephony features as well as many new features and communications applications are available in the IP-enabled world of converged communications. The number and type of IP Telephony and VoIP calling features are overwhelming and compelling. And they all come with no additional cost because they are IP-based and are carried over the computer network. They are more like computer applications that operate and run like well, computer applications.

Simple effective 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, go far and above any feature that a POTS-related system can deliver. Think how many times you wasted time over POTS telephony calling someone only to get a busy signal or their voicemail not knowing whether they were at their desk or not. The Presence feature is just one of many features available in the IP-converged communications world.

Or how about the ability to run IP SoftPhone software on your computer and doing telephony using a point-and-click process with a headset — talk about integrating the telephone with your computer! This capability could never be contemplated in the POTS world because it could not support computer-related applications in a seamless manner.

In a POTS world, separate systems had to be maintained to manage who was in the system and what their profiles were, known as the *translation.* The POTS telephony expert, depending on the system used by the company, has to gather this data on the user and key it into the POTS telephony systems of choice. The same information for the most part needs to be duplicated on the computer data network.

In the IP-converged communications world, most if not all of this type of information can be entered once and maintained in a uniform manner. Whether the company uses an Enterprise Resource Planning (ERP) software based approach or some other variation of Database Management Systems (DBMS), it can be integrated through Application Programming Interfaces (APIs) with all related application systems in the company as needed, including now all IP Telephony and VoIP systems.

VoIP Investment Protection

Most organizations today have one or more traditional telephony system models in place, or they have entirely or partially migrated to IP Telephony and VoIP to support their enterprise. If you have existing digital equipment (such as PBX with digital telephone stations), you can protect your investment by reusing most if not all of your equipment with VoIP.

Maintaining VoIP Seamless

Because of a foundation that eradicates duplication and redundant information systems, the major tasks of installing and managing IP telephony and VoIP become more cohesive. Managers have more effective and direct applications to support their many challenges. They can manage not only computer data applications, but IP-based telephony and videoconferencing systems, as well. Unified database applications running over the network provide real-time, seamless access to all information needed to maintain the VoIP network.

Moves, adds, and changes formerly requiring highly complex and costly resources and changes do not require the manager to do anything. The VoIP network automatically adjusts itself to accommodate the user's new location. Usage, accounting, and other metrics data are available to the manager through any computer device attached to the network. With IP-based converged communications, managing and maintaining the network become cost-effective and seamless. Staff do not get caught up in problems and stay focused on business deliverables.

Flexibility and Portability

IP SoftPhone is an IP telephone client for Windows-based PCs. It provides transparent access to real-time communications and productivity-enhancing features. It offers simple point and-click dialing. Through wireless extension to cellular, users have never had more telephone options for mobility available to them. For example, wireless extension to cellular enables the Follow Me feature. Employees can have calls ring at both their office and cellular telephones so they never miss a call. With IP SoftPhone for Pocket PC, employees can make and receive calls via the IP network with no recurring charges.

In an IP-converged communications network, any employee in the company can travel to any of the company's locations, plug in his or her IP-enabled laptop, begin work, and make and receive telephone calls. Employees have at these distant, temporary locations all of the rich features available to them at their home office locations. The network automatically identifies the user and applies that user's profile information in the company controlling database. Employees can even direct their calls to any digital desktop telephone at the temporary locations (the telephone does not even have to be IP-enabled).Absolutely no one needs to be called or notified that this user is connecting at this remote location. Managers no longer have to make costly and time-consuming accommodations for computer data and telephony connections for a coworker who is visiting their location.

Compelling Applications

If, as a manager, you remain unconvinced about VoIP with reduced overall operating expenses, increased and enhanced productivity, seamless integration of data, voice and video systems, unified controlling database that converges the need to maintain multiple databases into just one, increased mobility calling features that save time and money, then perhaps nothing will convince you to start your company's move toward VoIP.

But if you are on the fence and not sure which way to go, consider the emergence of the industry-leading Session Initiation Protocol (SIP) which is enabling many new features and applications all designed to make the employee more agile and mobile with information technology. Avaya once again has emerged as the industry leader with SIP technology applications. For example, the *Presence Detection, Follow Me,* and seamless *Moves, Adds, and Changes (MAC)* application features of the IP converged communications network are just a sample of the many new SIP applications. These features make IP Telephony and VoIP a truly valuable time-saving network service and increase employee productivity. SIP is one of the many cool IP developments that come with the IP converged communications network.

A user on an Avaya IP communications network with multiple devices like a cell phone, desk phone, PC Client, and PDA can rely on SIP to seamlessly integrate these entities. With SIP, the days of having to remember multiple voice-mail access codes or hardware addresses are over. SIP can logically integrate all of these codes and addresses through the IP communications network as if they are one device. SIP redefines productivity.

Increased Network Management

IP-based communication networks provide a foundation for comprehensive network management. As a result, the ability for you to manage every bit and byte that runs over your IP telephony LAN and your VoIP WAN has never been more enabled.

Likewise, you have at your disposal tools such as Avaya EXPERT Systems℠ Diagnostics Tools that find and fix network issues so quickly that managers rarely know anything at all happened. These types of tools can support local and remote network monitoring. In dedicated networks, 99.999% quality is provided. That's not to say that problems *never* occur, but in an IP-converged network environment, your ability to detect symptoms and make changes to your setup in advance of any problems that might befall your IP communications network is greatly enhanced.

Real-Time Collaboration

If you are still unconvinced, consider the fact that VoIP is IP based and many of the Web applications that previously ran exclusively over the Internet will now run over your private IP-based network. Many Web-based HTML applications are portable to your company's IP-based communication network. Your users can have their favorite Web page riding on their IP telephone. Or they can post special Web links on their telephone-based Web page. Many of the Web-HTML based applications are candidates for running with your IP telephones.

Users can add 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 this video and audio that runs on the company's network versus the Internet is free from the latency and jitter you see when running video and audio over the Internet.

Better Use of Available Bandwidth

Many people wrongly assume that when you add IP Telephony and VoIP to an enterprise computer network, you won't have enough bandwidth available to support the change. The fact is that dedicated network transports supporting computer data on traditional telephony systems are generally about 30 percent utilized. Even though converged networks that add IP Telephony and VoIP increase overall network traffic volume, you must look at how the IP based traffic operates.

On the LAN side, fault isolation provided through the 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 even 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 the WAN transports are higher and recur monthly when compared to LAN side Ethernet that is usually a one-time cost investment. Also, the bandwidth capacity of dedicated transports is usually measured on the basis of how many Digital Service Channels (DS0s) are possible.

A T-1 line, for example, has 24 DS0s and is among the most popular dedicated transports in the corporate world. If you run circuit-switched POTS-PSTN calls over the T-1, you can keep up 24 simultaneous telephony calls. (The LEC will still charge you for 24 POTS line equivalents.) However, the beauty of VoIP is that it is packetized and streams the packets in through the T-1 line over one fraction of one DS0 channel's bandwidth. As a result, you gain multiple times the bandwidth equivalent with VoIP when compared to POTS-PSTN on the WAN side.

Reduced Telephony and Video conferencing 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 local toll, intralata, intrastate, and/or interstate toll charges. All of these toll billing areas have recurring minute charges and regulatory fees. The big showdown area of toll service charges today is no longer in the realm of interstate carrier services. These costs have gotten down to as low as 2 cents per minute, and even this rate can be leveraged against overall minute volumes to define a lower per minute rate — below 1 cent per minute.

IP Telephony and VoIP can reduce local charges. But VoIP also reduces or eliminates regional or local toll carrier services charges. Depending on the number of locations your company may have and over how many intralata boundaries your current calling plans cover, you can save millions of dollars per month by converting to VoIP across the enterprise. This savings is mainly because if you put all your locations on VoIP, all their intralata (local or regional toll) *on-net* calls travel over your company's computer network. In this way, all these calls bypass the regulated, wire line carrier services of the conventional telecommunications carrier companies.

If your organization has significant international calling, the same argument applies except that your company can save even more toll and regulatory costs. International toll charges are the most heavily regulated.

Case Study: How Avaya Helped AGL Resources

With corporate roots extending back nearly 150 years, AGL Resources (AGLR) has grown from a pioneering gas light provider in downtown Atlanta, Georgia to become one of the largest natural gas distributors in the United States. AGL Resources has earned a reputation as one of the nation's fastest growing and most efficiently run utility companies.

High Performance Communications: Essential at AGL Resources

A highly disciplined firm, AGLR sees innovation technology and cost-effective business processes as fundamental enablers for business growth. Using these tools, the management team at AGL Resources has maintained a tight focus on growing shareholder value.

With growth of net income being a key measure of corporate success, AGL Resources certainly has reason to celebrate — in the face of a less than optimum economy, most recent year core earnings increased more than 22 percent on sales exceeding $860 million.

If you asked AGL Resources to identify the key corporate assets that directly impact the bottom line, business communications would be prominent on their short list. As manager of AGLR's voice communications, Louis Acuna is responsible for making sure the impact is a positive one.

"AGL Resources is very focused on the processes that are critical to the success of the business. Our communications infrastructure is viewed as one of AGLR's strategic assets and essential to revenue realization."

For AGLR, keeping a tight focus on satisfying customers is a fundamental business imperative, and communications capabilities are central to fulfilling that mandate. From an IT perspective, AGLR has a broad view of their customers; they range from internal employees to AGLR's external stakeholders —residential customers, commercial accounts, state regulators, and business partners. Their ability to conduct business directly depends on the integrity of AGLR's communications network.

As an energy utility, AGLR also has serious public safety responsibilities. With natural gas distribution being one of AGL Resources' core businesses, there is an absolute need for an always available communications network. When a gas line is damaged by a contractor or homeowner, their communications network plays a vital role in their ability to rapidly respond. In a business like this, there is no room for network downtime.

AGLR's Business Requirements

AGL Resources has 34 locations throughout the Southeast U.S., with more than 2,000 end users. Mr. Acuna explains that "As is often the case with businesses that engage in acquisition, the expansion of our business brought with it some site-to-site variability in our communications networks."

The lack of standardization presented some network management challenges, but far more apparent to the end users was the lack of consistent feature functionality across the different locations. Architecturally, their communications capabilities were still based on separate voice and data networks, and they weren't sure that was the best forward-looking approach.

AGLR had been following the evolution of IP converged platforms and saw that there appeared to be some substantive benefits to adopting an integrated infrastructure. Since one of their corporate thrusts is to leverage new technology to improve the efficiency of the business, Mr. Acuna and his team knew that AGLR leadership would be very interested in their findings.

The business needs were actually quite straightforward and all rotated around the importance of communications in supporting collaboration and rapid decision-making. Fundamentally, it is all about empowerment — using the communications network to empower all associates with information and reinforce a common sense of purpose.

Uniform information access

To ensure that the entire organization receives key information in a real-time fashion, leadership wanted messaging capabilities that would allow for simultaneous voicemail broadcasting to all locations. Because only 6 of the 34 sites shared a common voicemail system, this would put all associates on a fully integrated platform across all locations.

Fostering collaboration

To support a one-team approach, AGLR needed to provide associates with a common set of communications features to replace the local variations. They wanted the network functionality and interface to be the same regardless of which facility an employee happened to be on. They also needed a common dialing plan to foster real-time collaboration —allowing all associates to easily reach each other using an abbreviated internal numbering scheme.

Reducing operating costs

Leadership also wanted to reduce ongoing operational costs. This requirement was one of the main factors that really supported a move to IP convergence. Having voice ride on one set of facilities and data on another is an inefficient and unnecessarily costly network design.

Straightforward scalability

Platform scalability was another key requirement. Because AGLR's business is built on a growth strategy, the communications infrastructure needed to easily and cost-effectively scale to meet that growth.

Application interoperability

Network interoperability was a must. If AGLR went with a converged platform, the new voice hardware and software would need to seamlessly interact with its Cisco Systems backbone routers and Extreme Networks Ethernet switches.

AGLR also had a sophisticated 300 agent call center running on an Avaya DEFINITY Server R at its Riverdale location in Georgia. They had a variety of applications operating smoothly together — an Avaya Call Management System (CMS), Witness recording, Aspect workforce management, Avaya Interactive Voice Response (IVR) units, skills based routing — and there was no room for any performance backsliding.

Going with Convergence

In keeping with AGLR's focus on tight business processes, Mr. Acuna and the IT team made sure the project recommendations were well supported.

Before making a final decision on staying with traditional voice architecture or making the shift to convergence, Mr. Acuna and his team did considerable research. They talked with enterprises using a VoIP solution, and they issued a competitive bid that resulted in several submissions.

Armed with their research and the insights from other enterprises, they methodically evaluated the competitive options. When all was said and done, they decided on a total VoIP solution from Avaya — hardware, software, and full life-cycle services support.

At the heart of their new network is an S8700 Media Server running Avaya MultiVantage Communications Applications. The remote locations are running IP-enabled DEFINITY SI or Prologix voice servers. Each server is fully networked, enabling all employees to reach each other via 5-digit dialing.

A general theme of this new technology is doing more for less. The new platform allows AGLR to put all interoffice voice and data traffic on the wide area network (WAN) over common facilities. Running IP trunking to support the network connection needs of their 34 sites — including the new headquarters —significantly reduces facility and transport costs. As AGLR management drives these expenses out of their IT budget, they're able to reinvest those savings on other business technology initiatives.

For voice messaging, AGLR has three networked Avaya INTUITY™ AUDIX® systems that provide seamless voicemail across all locations. They also have a multimedia application — Avaya Message Manager — so that all associates can manage their voicemail and e-mail in the same mailbox. For AGLR's call centers, they're using an Avaya IP Agent application. This allows AGLR to deploy virtual call center agents in any of its locations, but with the same full functionality as the main center. The external customers still dial the same reach number, and the actual location of the agent is completely transparent to the caller.

The AGLR IT team also provided their executives with the Avaya Extension to Cellular (formerly EC500) feature, which enables "follow me" functionality so they can give out a single number that rings on their office and cell phones for maximum reach ability. On the internal IT side of things, they'll be deploying the Avaya VisAbility network management suite, which enables them to bring their network planning and reporting efforts toad new level.

Mr. Acuna adds, "If we choose to, the Avaya S8700 gives us the flexibility to host a new site directly from headquarters, speeding up the turning-up of the new location and eliminating the cost of a local server." He concludes, "Capacity for growth, rich features and applications, cost-effectiveness — this Avaya solution really positions us well for the future."

Decision Factors

In the final analysis, there were several key factors behind AGLR's decision to go Avaya, as described in the following sections.

Investment protection
AGLR had a considerable amount of Avaya digital and analog station equipment in the network already, and the VoIP design they chose allowed them to re-use all of it. Since terminal gear is typically one of the largest outlays involved in a network upgrade, AGLR was able to drive that expense to nearly nothing.

Ease of migration
AGLR's ability to move into convergence without disrupting the business was a must-have. This meant that the new system had to be easy for the employees to use, and that the actual deployment be transparent to the business. In terms of interacting with the system, the Avaya features are extremely intuitive and the required end-user training was minimal.

Confidence with Avaya
After several meetings with Avaya engineers, AGLR was convinced that their system could be smoothly deployed with no disruption to the business and end-users would remain productive. The Avaya converged solution was clearly the best combination of well-engineered technology coupled with the ability to provide expert support throughout the life cycle.

Convergence technology & service
The AGLR team and the Avaya implementation engineers worked closely throughout the entire implementation phase — from the upfront needs assessment and design, right through the physical cutover.

Collaborative approach to maintenance

When it came time to decide who to use as a maintenance partner it was easy to choose. A collaborative approach was essential. Mr. Acuna says, *"When you purchase an Avaya Global Services IP Maintenance agreement, you are purchasing the convergence expertise and experience that you would be hard-pressed to recreate internally. Basically, you are getting unlimited access to the experts. The simple fact is that maintaining a converged network at peak Quality of Service requires a variety of highly specialized troubleshooting and performance optimization skills. Avaya has the right people for the job."*

Learn More

For more information on how Avaya can take your enterprise from where it is to where it needs to be, contact your Avaya Client Executive or Authorized Avaya BusinessPartner, or visit us at **www.avaya.com**

About Avaya

Avaya enables businesses to achieve superior results by designing, building and managing their communications infrastructure and solutions. For over one million businesses worldwide, including more than 90 percent of the FORTUNE 500®, Avaya's embedded solutions help businesses enhance value, improve productivity and create competitive advantage by allowing people to be more productive and create more intelligent processes that satisfy customers.

For businesses large and small, Avaya is a world leader in secure, reliable IP telephony systems, communications applications and full life-cycle services. Driving the convergence of embedded voice and data communications with business applications, Avaya is distinguished by its combination of comprehensive, world-class products and services. Avaya helps customers across the globe leverage existing and new networks to achieve superior business results.

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Printed in the U.S.A.
10/05 • EF-LB2881