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Selecting VoIP for Your Enterprise

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Selecting VoIP for Your Enterprise

Prepared for Global Knowledge by Technology Marketing Corporation

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Introduction

Voice over Internet Protocol (VoIP) is a process of digitizing and sending voice telephone signals over the Internet or other data network. Enterprises of all sizes can benefit from this technology, but they must do some research to decide if VoIP is right for them. Which vendor should I call? Would I rather deploy and manage my telecommunications in-house, or does it make sense to outsource to a hosted services provider? How much will this cost, and how much money can I save in the long run?

The first thing most people realize about VoIP technology is that it can save their business money by reducing or eliminating the toll charges for long-distance and even local calling. However VoIP is more than a plan to lower a company's phone bill. There are also so-called "soft" benefits enabled by VoIP, such as increased worker productivity, the ability to collaborate among multiple branch offices, and lower operational expenditures as a result of simplified management schemes.

According to the Probe Group, the market for enterprise VoIP will grow to $7.1 billion in 2008. This presents a tremendous opportunity for numerous vendors to supply products and services to this market. This paper serves as an introduction to VoIP (its benefits and potential pitfalls) and provides insight into the various options available to enterprise decision-makers who face the task of selecting a VoIP-based phone system.

Cost Savings and Other Benefits

The main benefit of VoIP technology is that it is a cost-saver. For many enterprises, that is enough of a reason to consider VoIP. For example, in the case of Mississauga, Ontario, the city spent $3.3 million dollars on a new VoIP network to serve over 2,000 phones. The city is now enjoying a $700,000 annual savings in operational expenditures over their previous system.

Of course, cost savings come in many ways. Tremendous cost savings come in the form of lower telephone bills. By converting voice into packets and transporting these packets over an IP network, be it a private WAN or the Public Internet, corporations are able to avoid the Public Switched Telephone Network (PSTN) and the associated tolls.

In the case of an enterprise with multiple branch offices, this is especially true. By using the company's data network, enterprises can eliminate all costs associated with calling between branches. Furthermore, they can have all locations served by a single IP PBX, thus enabling extension dialing between far-flung locations. By simply dialing a coworker's extension, you can speak to a distant colleague as if he or she were in the very next cubicle. VoIP enables seamless call transferring to experts across a connected enterprise, whether they are in the same building, across town, or across the globe.

If the data network reaches a remote location, so too do the telephony applications that employees at the main corporate location enjoy. Productivity enhancing applications such as conferencing, voice mail, unified communications, and click-to-dial are all enabled across the enterprise. But managing the system is simplified due to eliminating the need to monitor multiple networks.

By combining separate voice and data networks into one network, VoIP enables cost savings from a network infrastructure perspective as well. In so-called "greenfield" deployments, there is no need to run two separate network cables (one each for voice and data). Furthermore, network administrators need manage only the single converged network.

VoIP also reduces the cost and complexity associated with moves, adds, and changes. Many enterprise VoIP solutions allow administrators to manage the system via a web-based browser interface and enable managers to enact changes to an employee's phone and voice mail settings remotely. All without the need to call the phone system's manufacturer to send a representative to make those moves, adds, and changes. It may not take much work to physically move a single phone or to set up a new employee's voicemail system, but imagine an organization that employs thousands of people across multiple locations. The costs add up quickly. VoIP practically eliminates that expense.

Another benefit of VoIP is that many enterprise telecommunications systems are upgraded in a piecemeal fashion, allowing companies to migrate to VoIP slowly, department by department, in order to maintain a healthy measure of control over the transition. Any problems that pop up in early phases of deployment can be worked out before migrating the next group of users.

Potential Pitfalls and How to Overcome Them

Voice Quality

A serious reservation most people have is the belief that the quality of VoIP is inferior to that of traditional telecommunications. While that has been a problem in the past, it is becoming less of an issue. In fact, many VoIP deployments claim to have voice quality of service that is in some respects better quality than legacy communications.

Still the problem does not simply go away. There are still impediments to voice quality that need to be addressed. For the vast majority of enterprises considering running voice traffic over their data infrastructure, a network assessment is in order. In fact most businesses will need to spend some capital to upgrade their networks to be capable of delivering a consistent high-quality voice experience. A thorough understanding of the potential return on your investment is necessary to ensure that the migration to VoIP is a cost-effective endeavor.

Let's take a look at some of the more common problems experienced in a VoIP call: delay, jitter, and latency.

According to Tehrani's IP Telephony Dictionary, here are the three culprits defined:

Delay — The amount of time it takes for a signal to transfer or for the time that is required to establish a communication path or circuit.

Jitter — (1) Jitter is a small, rapid variation in arrival time of a substantially periodic pulse waveform resulting typically from fluctuations in the wave speed (or delay time) in the transmission medium such as wire, cable, or optical fiber. When the received pulse waveform is displayed on an oscilloscope screen, individual pulses appear to jitter or jump back and forth along the time axis. (2) The short-term variation of transmission delay time for data packets that usually results from varying time delays in transmission due to different paths or routing processes used in a packet communication network. (3) In IP telephony, the variance of inter-packet arrival times.

Latency — Latency is the amount of time delay between the initiation of a service request for data transmission or when data is initially received for retransmission to the time when the data transmission service request is granted or when the retransmission of data begins.

As you can see, all of these terms deal with the impediment to transmission of VoIP packets. And delayed or missing packets spell doom to quality VoIP communications. Consider that if you send an e-mail that arrives with a two-second delay. That delay is imperceptible to the end user. Now imagine that same delay in the middle of a conversation. Unacceptable!

According to the International Telecommunications Union, acceptable delay for VoIP calls can be described as follows:

- 0 to 150 milliseconds is acceptable for most any application.
- 150 to 400 milliseconds is acceptable for international connections.

Keep in mind that delay can creep in at many points throughout the network that are beyond your control. So it is imperative to employ a quality of service solution that keeps the delay to an absolute minimum on the originating side of the call.

Voice quality is calculated by a measurement called MOS, or mean opinion score. This measurement is determined by an actual group of human listeners who rate the quality of audio on samples played to them. The MOS rating system is judged on a five-point scale where a score of 1 implies poor quality, and a score of 5 is excellent. Toll quality telephone service is generally granted a MOS score of 4.0, and as such is the goal to strive for.

Security

Security is another critical issue facing enterprises interested in deploying VoIP. Certainly this is an area that needs to be addressed in any network deployment of voice. Some industry professionals believe that VoIP in the context of network security needs to be addressed as simply another data application on that network. Still others believe that VoIP has a number of peculiarities that demand more specific attention.

Most VoIP solutions are deployed on servers that run either proprietary operating systems or on commercially available servers such as those offered by Microsoft. These servers are as susceptible to hacking or denial-of-service attacks as any other servers deployed in an enterprise IT department. If troublemakers were to break into a company’s voice system, vast amounts of potentially sensitive data would be at risk. As such, it’s important to separate voice equipment behind firewalls—frequently updated with the latest patches and monitored to keep an eye-out for intruders.

Other concerns include the "hijacking" of a voice gateway to place unlimited unauthorized free calls across the globe. Also, eavesdropping is of great concern to executives who would prefer their phone calls to be as secure as possible.

The use of computer-based softphones, and even IP handsets, also leaves an enterprise open to a security breach. These devices should be encrypted. User access should be controlled through password protection. And phones might even be limited as to what type of calls they can place (international, 900 number restrictions, etc.) to make sure no unauthorized users can place calls if they are able to breach the firewall and get inside the network.

Other general suggestions for protecting your voice traffic from security threats include: making sure to encrypt the voice stream; keeping a close eye on your network through the use of an intrusion protection solution; making sure your firewalls are properly configured; and taking precautions to secure your servers as you would for other critical business infrastructure and applications.

Emergency Services/911

One of the more pressing concerns facing VoIP is the issue of emergency services such as 911. As VoIP enjoys rapid deployment, developers are tasked with the challenge of making sure that the speed of innovation does not outstrip the ability to supply emergency services to users of VoIP. While many consider this to be a service provider or residential customer concern, there are a number of concerns for enterprises that should be addressed.

One current limitation of VoIP is that the caller's physical location is not broadcast to the emergency services' PSAP (public safety answering point). Enterprises can work around this shortcoming by assigning a physical location to each phone, and having that information transmitted to the PSAP. This way emergency services workers can respond to the exact location of an injured employee. Even if that employee switched offices and took his or her IP phone, chances are the physical location would be in the same building. But what happens when that employee, who uses a softphone, travels to another city, or simply chooses to work from home? If an emergency call is placed to 911 from the caller's new location, chances are emergency service personnel will respond to the main office location—wasting time and valuable resources deployed to the wrong location.

Another issue involves funding of emergency 911 infrastructure. VoIP users can avoid paying appropriate 911 surcharges while still using the service provided by their VoIP service provider. There is concern among public agencies that system funding will come under attack as enterprises drop legacy phone services and deploy VoIP.

Network Outages

Traditional telephone systems are powered over the lines that are drawn from the local phone company's central office, and it is extremely rare for these phone systems to suffer outages due to problems with supplying electricity. This is not the case for VoIP phone systems. VoIP phones demand a power source within the enterprise, which is not a problem. The problem is that when power is interrupted, the phone systems will not work. When selecting a VoIP phone system for your enterprise, it is necessary to factor this into your selection criteria.

A business' phone system is critical. Customers have an expectation that they can reach you whenever they need to. Likewise, companies that are switching to VoIP have the expectation that their phone system will perform at least as well (if not better) than their legacy systems. If a new phone system with all the bells and whistles promised by VoIP sits idle for lack of a well thought out power plan, it amounts to nothing.

Some of the solutions a business needs to look into are an uninterruptible power supply (UPS), high-quality power that will not degrade the telecom equipment, redundant systems, and remote management and monitoring.

A UPS will continue to supply power to the phone system for a limited time until the main power is restored. These devices come with a varying amount of battery backup power. If multiple branch offices are served from a central server, it is important to make sure that there are no disturbances in the power supply or else the quality may be affected at the branches. Furthermore, you should consider outfitting your branches with UPSs in case localized power outages occur. Lastly, a remote monitoring system will allow you to see if there are any power issues within your network. This will help response time, and increase the ability to nip any problems in the bud.

Other power issues revolve around supplying power to the individual phones that reside on people's desks. Power over Ethernet solutions abound, and it is imperative that you ask your provider to explain how they work and why they are important. Essentially, Power over Ethernet is able to supply power and data to IP phones, as well as other devices over existing LAN wiring, without the need to modify the existing infrastructure. Among the benefits provided by this technology is that it obviates the need to run power and data cabling to each desktop phone location. Furthermore, when used in conjunction with a UPS, this solution maintains power to the telephones for as long as the battery backup remains active.

Deployment: In-house vs. Service Provider

While there are many choices when it comes to enterprise VoIP, in the end the basic decision boils down to deploying the solution in-house versus having a service provider supply the solution in a hosted fashion (IP Centrex). Both approaches are viable. The decision depends on what's right for your enterprise.

Gartner, an industry analyst firm based in Stamford, CT, predicts that business lines supplied as part of local VoIP carrier services in the United States (which includes IP Centrex) will grow at an 83 percent compound annual growth rate (CAGR) out to 2008. Gartner also predicts strong growth in the enterprise IP telephony space with a 34 percent CAGR of Pure IP-PBX line shipments out to 2008.

As with buying or leasing a car, with either system, you still get phone services and equipment with similar voice quality and reliability. With IP Centrex, it is like leasing. Users do not own the equipment, and in fact most of the equipment, with the exception of the telephones, resides at the remote IP Centrex site, not at the user site. With IP PBX, users buy the equipment and maintain it at their site. With either choice, the features reach across the network, regardless of geography or type of telephone, whether IP telephone, softphone, or wired phone, giving users access to the office telephone system remotely or at the company. Individual users probably will not even know or care if their company has IP Centrex or an IP PBX. They just want to know that their phones work.

With IP Centrex, the user pays on a periodic basis. There is not a steep upfront cost—but the cost is ongoing. With an IP PBX, the upfront cost can be significant or it can be financed to make it more affordable. Today, IP Centrex providers charge much like a wireless provider, billing a contracted monthly rate plus a variable rate for additional services.

Choosing a vendor plays a key role. Geography does not matter, because the provider of IP Centrex services can be based anywhere and still provide services. What does matter is the vendor's reputation and track record for quality, reliability, and overall feature set. In fact, security features have become a major consideration in today's telecom decision.

As with leasing cars, if a company plans to keep its system for a while, say more than five years, buying an IP PBX may make more sense, especially if it comes with the ability to migrate to future systems and/or grow to accommodate the company's growth.

There is no doubt that ownership equates control, but it also means that there is a learning curve. Owning an IP PBX means there is a significant up-front commitment to install, maintain, and learn how to use the system. All these considerations should be factored into the total cost of ownership. The upside is that once the system is running smoothly, it should stay that way for a very long time. For companies that do not have the staff or bandwidth to implement an IP PBX, IP Centrex may be the best way for them to get the benefits of IP telephony.

With either IP Centrex or IP PBX, users can move the phones wherever there is Internet access without the traditional service charges for moves, adds, or changes. For IP PBX systems, it is typically less expensive to add new users than with IP Centrex, because all the data and infrastructure is already built into the telephone system. As well, both IP Centrex and IP PBX are very reliable systems.

The jury is still out on what the best solution is for a company with multiple sites. Since IP phones or softphones can be plugged in anywhere there is Internet access, they can basically function as nodes off the home office's IP PBX or from the IP Centrex. With either system, companies can have IP telephony in as many offices or home offices as they'd like, without any geographic limitations.

For users concerned with disaster recovery and survivability, there isn't a clear winner. It all depends on where the disaster takes place. For maximum survivability, the system, whether it's an IP PBX or IP Centrex, needs to be redundant and not physically co-located. Calls need to be automatically rerouted when the system goes down. The one advantage IP PBX systems have is that because users can set it up themselves, they have the control to establish the level of potential disaster recovery that is a priority for their business.

IP Centrex systems are also very reliable during a disaster, if it occurs at the client site. IP Centrex can usually reroute calls very swiftly, and because the IP phones can be plugged in anywhere, users can be connected very quickly in the event of a disaster. However, if the disaster happens at the IP Centrex Switch, it becomes a major issue for the enterprise.

What it really comes down to is preference:

- If you want to own it a long time, the IP PBX might be cheaper in the long run.

- If you want to keep the system a shorter time, IP Centrex might not cost you as much up front and you can contract for less time.
- If you want special services or customization, buy the IP PBX.
- If you want basic services (calls in, calls out) use IP Centrex.

Consider the trade-offs and be clear on what you really must have in your system today and into the future.

Summary

VoIP is proving to be a viable new alternative to traditional telecommunications solutions in the enterprise, as businesses who are shopping for phone systems are increasingly turning to VoIP as their first choice. But telecommunications is the lifeblood of an enterprise, and as with any major decision, one needs to do some research before deciding on how to proceed.

VoIP offers many benefits, from lower ongoing telecommunications bills to soft benefits such as increased worker productivity and the ability to conduct ad hoc collaborative sessions across multiple locations. But there are many elements to consider when selecting an enterprise VoIP phone system, and many potential pitfalls that you need to be aware of before making the final move to VoIP.

Enterprises need to figure out if VoIP is right for them. And if so, which approach works best: hosted or in-house. Both approaches are viable, yet both have their drawbacks. The decision depends on what’s right for your situation.

VoIP is the future of enterprise telecommunications. If you arm yourself with the right questions and a thorough understanding of your needs, your migration to VoIP should be a positive experience.

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Glossary

CAGR — Compound annual growth rate. The year over year growth rate of an investment over a specified period of time.

CLEC — Competitive Local Exchange Carrier. A telephone service company that provides local telephone service that competes with the incumbent local exchange carrier (ILEC).

Codec — Coder/Decoder A technique for compressing information to a fewer number of bits for more efficient transmission and storage (coding), and subsequently recovering the original data (decoding). Normally the term codec applies only to compression of human-perceived signals such as speech, audio, images, or video; and it usually refers to lossy compression.

CTI — Computer Telephony Integration.

Delay — The amount of time it takes for a signal to transfer or for the time that is required to establish a communication path or circuit.

FCC — Federal Communications Commission

Firewall — A firewall is a data filtering device that is installed between a computer server or data communication device and a public network (e.g. the Internet). A firewall continuously looks for data patterns that indicate unauthorized use or unwanted communications to the server. Firewalls vary in the amount of buffering and filtering they are capable of providing.

FTTC — Fiber to the curb. A distribution system that uses fiber optic cable to connect telephone networks to nodes that are located near homes or any business environment (near the curb). The fiber optic transmission is used to provide broadband services beyond the central office, all the way to the last 50-100 feet from the subscriber. The service pedestal is said to be "at the subscriber's curb."

FTTH — Fiber to the home. A distribution system that uses fiber optic cable to connect telephone networks to nodes that are located in the homes of customers. The fiber optic transmission is used to provide broadband services beyond the central office, all the way through the drop wire to the optical node that is located in the customers home.

G.711 — A standard analog to digital coding system (coded) that converts analog audio signals into pulse code modulated (PCM) 64 kbps digital signals. The G.711 is an International Telecommunications Union (ITU) standard for audio codecs. The G.711 standard allows for different weighting processes of digital bits using mu-law and A-law coding. The G.711 standard was approved in 1965.

G.723 — An International Telecommunication Union (ITU) standard for audio codecs that provides for compressed digital audio over standard analog telephone lines.

G.729 — A low bit rate speech coder that was developed in 1995. It has low delay due to a small frame size of 10 msec and look ahead of 5 msec. It has a relatively high voice quality level for the low 8 kbps data transmission rate. There are two versions of G.729: G.729 and G.729 A.

H.323 — H.323 is an umbrella recommendation from the International Telecommunications Union (ITU) that sets standards for multimedia communications over Local Area Networks (LANs) that may not provide a guaranteed Quality of Service (QoS). H.323 specifies techniques for compressing and transmitting real-time voice, video, and data between a pair of videoconferencing workstations. It also describes signaling protocols for managing audio and video streams, as well as procedures for breaking data into packets and synchronizing transmissions across communications channels.

ILEC — Incumbent local exchange carrier. A telephone carrier (service provider) that was operating a local telephone system prior to the divestiture of the AT&T bell system.

IP Centrex — IP Centrex is the providing of Centrex services to customers via Internet protocol (IP) connections. IP Centrex allows customer to have and use features that are typically associated with a private branch exchange (PBX) without the purchase of PBX switching systems. These features include 3 or 4 digit dialing, intercom features, distinctive line ringing for inside and outside lines, voice mail waiting indication and others.

IP PBX — A private local telephone system that uses Internet protocol (IP) to provide telephone service within a building or group of buildings in a small geographic area. IPBX systems are often local area network (LAN) systems that interconnect IP telephones. IPBX systems use a IP telephone server to provide for call processing functions and to control gateways access that allows the IPBX to communicate with the public switched telephone network and other IPBX's that are part of its network. IPBX systems can provide advanced call processing features such as speed dialing, call transfer, and voice mail along with integrating computer telephony applications. Some of the IPBX standards include H.323, MGCP, MEGACO, and SIP. IP PBX represents the evolution of enterprise telephony from circuit to packet. Traditional PBX systems are voice-based, whereas their successor is designed for converged applications. IP PBX supports both voice and data, and potentially a richer feature set. Current IP PBX offerings vary in their range of features and network configurations, but offer clear advantages over TDM-based PBX, mainly in terms of reduce Opex (operating expenses).

IP Phone — An Internet protocol phone (IP phone) is a device (a telephone set) that converts audio signals and telephony control signals into Internet protocol packets. These stand alone devices plug into (connect to) data networks (such as the Ethernet) and operate like traditional telephone sets. Some IP Telephones create a dialtone that allows the user to know that IP telephone service is available.

ISP — Internet service provider

Jitter — (1-general) Jitter is a small, rapid variation in arrival time of a substantially periodic pulse waveform resulting typically from fluctuations in the wave speed (or delay time) in the transmission medium such as wire, cable or optical fiber. When the received pulse waveform is displayed on an oscilloscope screen, individual pulses appear to jitter or jump back and forth along the time axis. (2-packet) The short-term variation of transmission delay time for data packets that usually results from varying time delays in transmission due to different paths or routing processes used in a packet communication network. (3-IP Telephony) The variance of interpacket arrival times.

LAN — Local-area network

Latency — Latency is the amount of time delay between the initiation of a service request for data transmission or when data is initially received for retransmission to the to the time when the data transmission service request is granted or when the retransmission of data begins.

MOS — Mean opinion score (MOS) is a measurement of the level of audio quality. The MOS is number that is determined by a panel of listeners who subjectively rate the quality of audio on various samples. The rating level varies from 1 (bad) to 5 (excellent). Good quality telephone service (called "toll quality") has a MOS level of 4.0.

PBX — Private Branch eXchange. A private telephone network used within an enterprise.

PSAP — Public safety answering point. An agency that receives and processes emergency calls. The PSAP usually receives the calling number identification information that can be used to determine the location of the caller.

PSTN — Public switched telephone networks are communication systems that are available for public to allow users to interconnect communication devices. Public telephone networks within countries and regions are standard integrated systems of transmission and switching facilities, signaling processors, and associated operations support systems that allow communication devices to communicate with each other when they operate.

QoS — Quality of service (QoS) is one or more measurements of desired performance and priorities of a communications system. QoS measures may include service availability, maximum bit error rate (BER), minimum committed bit rate (CBR) and other measurements that are used to ensure quality communications service.

RBOC — Regional Bell Operating Company. A United States telephone company that is one of the seven telephone companies that were created as a result from the division of AT&T in 1983. RBOCs are also known as the Baby Bells. The RBOCs were Ameritech, Bell Atlantic, BellSouth, Nynex, Southwestern Bell Corporation, Pacific Telesis, and US West.

ROI — Return on Investment is a financial measurement that compares the profit with the original investment. ROI evaluates the impact of an investment on the telephone company's profitability or operational efficiency: dollars spent compared to benefits gained.

SIP — SIP is an application layer protocol that uses text format messages to setup, manage, and terminate multimedia communication sessions. SIP is a simplified version of the ITU H.323 packet multimedia system. SIP is defined in RFC 2543.

SMB — Small and medium businesses

SOHO — Small office, home office

UPS — A battery backup system designed to provide continuous power in the event of a commercial power failure or fluctuation. A UPS system is particularly important for network servers, bridges, and gateways.

VoIP — A process of sending voice telephone signals over the Internet or other data network. If the telephone signal is in analog form (voice or fax), the signal is first converted to a digital form. Packet routing information is then added to the digital voice signal so it can be routed through the Internet or data network.

WAN — Wide-area network