VoIP without Hype

What businesses need to know.

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VoIP: On the Rise Despite Shortcomings

"With voice over IP, you can have long distance for less – even for free!" That is the Siren's song being played by VoIP, the Internet's newest promise. Millions of consumers with broadband connections are responding. And there's no end in sight; it's expected that 32 million Internet phone lines will be in use by 2009 (Source: Gartner).

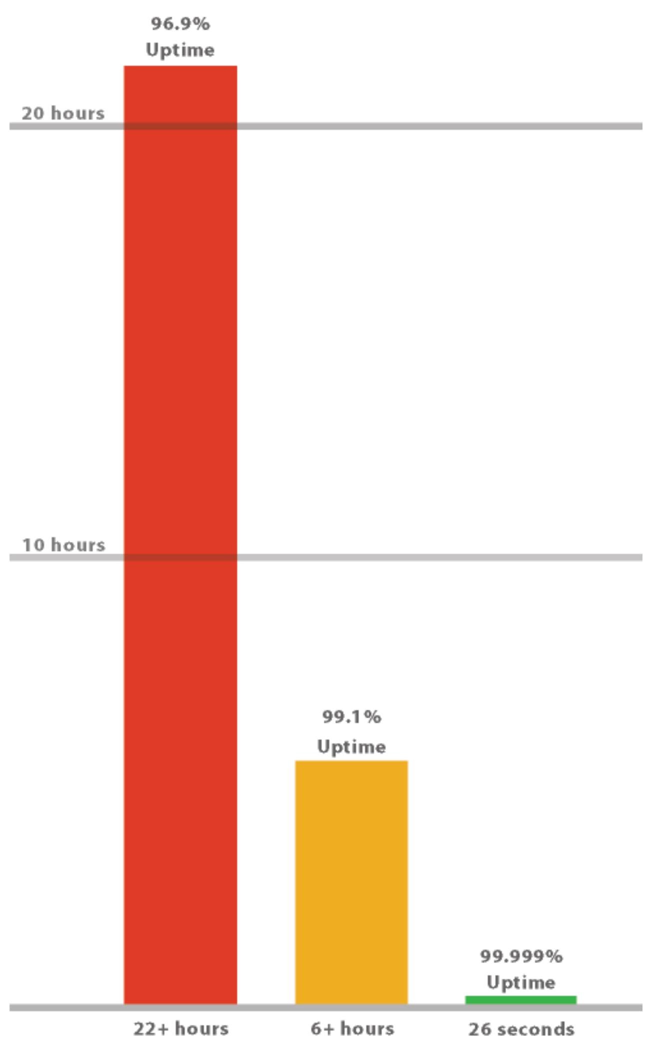
Akin to the cellular phenomenon, consumers are rushing to Voice over Internet Protocol (VoIP) despite the fact that audio quality and reliability are not yet up to traditional landline telephony standards (*see "I want my V-O-I-P" section for quality analysis*). Downtime and quality aside, the value proposition of VoIP has clearly resonated with consumers. During 2005, U.S. subscriptions to VoIP calling plans, which cost as little as $20 to $25 per month for unlimited domestic long distance (LD), more than tripled, from 1.3 million to 4.5 million (Source: TeleGeography).

Seductive as the savings message may be to consumers, the lure of VoIP is even stronger for businesses – whose monthly LD bills are often in the hundreds or even thousands of dollars. Many of these businesses have exacerbated toll charges because they either pay for calls between branch offices or have call centers, which typically incur heavy line usage. For such firms, slashing LD expenses could literally drop thousands of dollars a month straight to the bottom line. By the same token, the more dependent a business is on telephone communications, the less it can afford to compromise on the quality of its phone connections. Poor audio quality can undermine productivity and customer satisfaction; dropped calls cost sales and money; a total telecom service outage might well be disastrous.

I Want My V-O-I-P

It's important to approach VoIP with your eyes wide open – to understand the technology involved, the investment required, and an affordable migration path to VoIP that will deliver value to your business, both short- and long-term, without compromising the quality of your communications. But first, let's talk quality. A study by Internet performance monitor Keynote Systems found that consumer VoIP service reliability improved from 96.9 percent to 99.1 percent between June 2005 and January 2006. 99.1 may sound good, but actually it is significantly lower than the 99.999 percent reliability rating that people have grown used to with the plain old telephone system (POTS), also referred to as the public-switched telephone network (PSTN). To put these figures into perspective, 96.9 percent uptime actually equates to over 22 hours of downtime per month and 99.1 percent uptime still equates to more than 6.5 hours of downtime a month. Compare those numbers to 99.999 percent uptime, which equates to just 26 seconds of downtime per month, and you can see the impact of a few decimal points!

VoIP Downtime Analysis



Packets 101: What Makes VoIP Vulnerable

As most people understand, VoIP is telephone calling over the Internet. So, it may appear that switching from the POTS to VoIP is a fairly straightforward proposition for the average small or medium size business (SMB). In a typical SMB, all employees have PCs, which are connected together via a local area network (LAN). This LAN is then connected to a router/firewall which talks to the Internet via what is called a wide-area network, or WAN. So, if you put a VoIP-capable phone on each desk and plug it into your LAN, the call goes out your WAN and voilà – IP telephony, right? Wrong.

Many people don't realize that the Internet was *not* originally built for telephone calling. Neither were most LANs – yes, this means you! Even if you have a great LAN, it is unlikely that your WAN, or the WAN of your Internet Service Provider (ISP) is ready to go. In fact, the Internet Protocol itself was not really designed for real-time communication of any sort – especially the streaming nature inherent in audio, video, or online gaming.

Let's get geeky. IP-based networks divide information – for example, an email message or Web page – into thousands of small chunks of data called "packets". Each packet has a "header" which contains both the sender's and the receiver's Internet address; this enables the packets, which are transmitted separately, to be reassembled at the receiving end. Think of packets like ants, each of which knows how to get back to their anthill, but doesn't necessarily need to get back at the same time or by the same route. This ability for a packet to "choose its own route and time" is why data transmission over the Internet is so efficient.

See, when data packets carrying email or web pages arrive slowly, or out of order, it is usually not a big deal. So what if it takes you an extra second to download your email? Space out for a minute, watch the paperclip, you know the drill. But this method of "out of order" packet delivery spells disaster for real-time protocols that need packets to arrive at the right time, in the right order, all the time. Welcome to the stringent demands of transmitting real-time audio.

OK, let's adjust our pocket protectors and really talk about this; during a VoIP call, speech is captured as analog information by a phone, then converted into digital information, compressed, and divided into packets for transmission. This whole process is relatively fast, easy and reliable. The potential problems happen at the receiving end, when the packets must be reassembled in the correct order, absolutely error-free, and reconverted from their digital form into a seamless audio stream – all *in real time*.

If there are any substantial glitches in transmission, a VoIP call "breaks up" like the reception from a distant radio station. As engineers say, the audio stream *stutters*. As the people on the call will tell you, it sounds like gibberish. Houston...we.. av...a..ro..bl...em. In the worst case, which is not all that rare, the entire call is dropped – that is, cut off – because the transmission becomes so overwhelmed by problems that the connection simply fails. Dropped calls can mean lost revenue and/or dissatisfied customers!

VoIP vs. IP Telephony: A moment of distinction

Recently, VoIP has become a catch-all buzzword. Yet, it is important to distinguish between VoIP, which is a digital transport vehicle for *phone calls*, and IP telephony, which is a digital *phone system* based on Internet standards. This is important, because businesses stand to benefit from both VoIP and IP telephony – in substantially different ways.

VoIP is a method of digitizing your voice so that it can be transmitted across the Internet to save money on toll charges. Whereas IP telephony is a way of digitizing your phone system so that it can leverage the Internet, your computer, and your other business software applications (CRM, CTI, Outlook) to increase productivity within your business. VoIP is actually a subset of IP telephony – look at it this way: VoIP is an "arrow in the quiver" of IP telephony.

Any IP telephony system will use VoIP as a way of transmitting voice, in some manner or another (SIP, Skinny, MGCP, H.323). But, an IP telephony system, such as an IP-PBX, goes far beyond cheap phone calls; it enhances business productivity by providing additional features that weren't available or affordable with legacy phone systems.

Is Your Network Ready for VoIP? Probably Not

Here's a fact: the LANs and Internet connections (WANs) used by most SMBs are simply not ready to handle VoIP. The basic firewalls commonly used for security and virus protection often cause VoIP calls to break up. The low cost routers from the local computer store often don't have the horsepower to drive quality VoIP calls. LANs can also become congested, especially when users are transferring large files across the internal network, such as when sending or receiving emails with large files attached, downloading documents, doing file backups or copying media files.

Of course, your VoIP network includes not only your LAN but also your WAN. Your WAN begins with your broadband modem and ends with your broadband Internet provider. Most people don't understand that just having a broadband connection is not enough. You actually need a *high quality* connection to deliver the call quality you need to run a real business. Many SMBs connect to the Internet via DSL or cable, most often with inexpensive modems. While such connections work fine for web browsing and email, they are not designed to handle VoIP transmissions, much less the combination of voice *and* data.

Similarly, many of the WANs now in use by ISPs were designed and built before the advent of IP telephony. They weren't originally designed to meet the demanding requirements of error-free, reliable VoIP transmission. In fact, most of them actually run on a business model designed for oversubscription, which results in frequent latency and jitter (*see sidebar: "Into the Weeds: VoIP explained"*). Even the ISPs that aren't oversubscribed have not yet deployed true quality-of-service (QoS) technology to ensure that voice packets get priority over data packets across their networks.

As if ensuring you have a great LAN and WAN was not enough, you also have to choose a high quality VoIP provider (VSP or ITSP). [Is your acronym meter at full capacity yet?] Much like LD providers, which deliver long distance services over the PSTN, VSPs provide you with VoIP calling over the Internet. Like ISPs, not all VSPs are created equal in terms of network strength, proximity to the PSTN backbone and, of course, good old fashion customer service. The VSP industry is a new one, so be sure to choose carefully. Remember a great LAN and WAN don't mean anything if you have a weak VSP!

Into the Weeds: VoIP explained

Let's talk protocol! This will help us understand why the real-time demands of VoIP put such a strain on the Internet and your LAN.

As mentioned previously, the Internet was originally designed so that packets could arrive "out of order" and be reassembled by the client. The protocol most commonly used to transmit these packets is called TCP/IP.

Yet, TCP/IP is rarely used for VoIP packets because this protocol was not designed for real-time communications. Instead, VoIP often uses the UDP protocol. UDP tends to carry low overhead, making it a good choice for voice calls. But, the low overhead of UDP also makes it sensitive to network conditions. Therefore, VoIP-over-UDP can sound poor when it encounters any of following conditions:

Latency is the time it takes for a data packet to make a round trip between the sending and receiving location. When the average latency of a VoIP connection is above 200ms, call quality suffers. The best VoIP connections have latency under 80ms. Email and web access can gracefully handle latency of 400ms.

Jitter is the result of *variance* in latency between subsequent packets. For example, if you ping a network and get results such as 90ms, 92ms, 89ms, the network is jitter-free or, to be exact, it has jitter of 3ms (variance between 92ms and 89ms). But if your pings looked like: 50ms, 70ms, 190ms, then your network has jitter of 120ms. Jitter that exceeds 100ms degrades the quality of a VoIP call. Jitter under 50ms is gracefully handled by most IP phone systems.

Packet loss occurs most commonly when an Internet network is congested. Under such conditions, packets are often simply discarded. TCP/IP automatically retransmits these lost packets, but VoIP-over-UDP will not. Packet loss will create "stuttering", or in extreme cases, "silent gaps" in your phone call.

Upgrading for VoIP: Four Approaches

For most SMBs, the migration to VoIP will involve more than choosing a VSP to replace their current LD provider. It will involve upgrading on two fronts: the LAN – that is, the networking equipment on your premises – and the WAN, including your connection to the Internet and the equipment and capabilities of your service provider. There are four distinct approaches you can take to go VoIP, depending on your quality requirements and budget.

Approach #1: Best and Most Expensive

For the best voice quality you should have two entirely separate LANs at your place of business and two entirely separate WANs (connections to the Internet). Dedicate your first WAN+LAN to data and your second WAN+LAN to voice – thus ensuring no physical possibility that your data packets can "stomp" your voice packets. Then, when planning your voice WAN, try to find budget room for a T1 connection instead of a lower quality DSL/cable connection.

Now, when planning for your two LANs, you will need to purchase two separate routers, each with their own physical wiring, which will terminate as two separate Ethernet jacks at each employee location. This means that your employees will plug their computer into one jack and their IP phone into another. Expect to pay $80-$100 per employee location for each extra "drop", but much less if you have your wiring company run both drops at the same time.

Finally, when you have built your dedicated WAN+LAN combos, you must carefully choose a VSP!

Approach #2: Second Best, Less Expensive

If you don't want to pay for two separate LANs, you can still get pretty darn good quality if you do two things. First, you must still get two separate broadband providers for your WANs, one for voice and one for data – again, a T1 if you can afford it. Second, you should upgrade to a QoS Ethernet switch on your LAN. A QoS-capable switch (with IEEE 802.1P support, for example) will ensure that voice packets and data packets are prioritized properly on your LAN – thus when large files are moving across your LAN, your switch will make them pause momentarily to let voice traffic go first. Your users won't even notice the pause, but your voice quality will be significantly improved.

Approach #3: Cheaper but Harder to Pull Off

A more economical approach, but often the hardest to do right, is to add QoS capabilities to both your LAN and WAN, therefore allowing you to get away with only having one of each. To do this properly, you'll first need to upgrade your LAN with a QoS-capable switch/router. You'll also need to ensure that your broadband provider has QoS capabilities and that your VSP uses the same type of QoS as you use on your LAN. Yes, QoS comes in many flavors. In fact, the easiest way to pull all of this off is to get a T1 that provides you both Internet access and VoIP-based LD on the same circuit, all with QoS, and all from the same broadband provider!

Approach #4: I Want Really Cheap!

OK, OK, we get it. You love your wallet and you didn't pay for this white paper. So, at the absolute minimum, you must upgrade your broadband connection to ensure sufficient bandwidth for voice traffic from your premises to the Internet. A typical VoIP call will use about 64KB of upstream and downstream, and you should factor in at least 90KB for your first call because of what is called "IP overhead." So, do your math assuming peak concurrent line usage. If your peak usage will be 5 concurrent calls, then you add 90KB + (4 x 64KB) = 346KB. Be sure to remember that this is 346KB up *and* down. Many broadband providers will give you much more downstream than they will upstream, so be sure to ask! Finally, if you are going to go cheap, and cannot afford a T1, try to at least get "business-grade" DSL (*see sidebar, "What It Will Cost"*).

In case we haven't provided you too much information already, there are two final things to note about making your leap to VoIP. First, ensure that your VSP has a local PSTN media gateway in your area; this will shorten the path your VoIP calls have to take over the Internet before they are converted to travel across the PSTN. Second, try to use the same broadband provider at your main office and all remote offices; VoIP performance is usually better when calls travel over a single provider's backbone vs. having to "hop" across multiple backbones.

Hybrid IP-PBX: The First Step to VoIP

Whatever approach you decide on for migrating to VoIP, a hybrid IP-PBX is an excellent first step. Hybrids operate in three modes – PSTN, VoIP, and what's called PSTN-fallback – a mode which ensures that you'll always have phone service, even during Internet outages. With a hybrid IP-PBX, you can also connect and use analog phones (including cordless sets), IP phones, or a combination. So you can convert select employees to IP telephony according to their needs and the capacity of your Internet connections.

A hybrid IP-PBX enables you to start saving money right away, even if you choose to use the PSTN connections to the outside world. With a hybrid IP-PBX at your business, you get free VoIP calls between offices and with all your telecommuters, but you can selectively choose to pay more for calls across the PSTN where the quality matters most. Think of it this way, your employees get free VoIP calling between themselves, but your customers are guaranteed perfect POTS quality when they call you, or you call them.

But hybrid IP-PBX systems aren't just about VoIP and cheap LD, they also enable SMBs to improve their "communications image" – presenting callers with a professional (and time-saving) auto-attendant, allowing employees to work from home or the road, 4-digit dialing from anywhere in the world, blending Outlook into your phone system and more.

Hybrid IP-PBX capabilities can include:

- unlimited extensions and voicemail
- multiple auto-attendants (IVR)
- unlimited call queues (ACD)
- telecommuters – even for call queues
- uploading of professional voice prompts
- scheduling – time and date settings
- music on hold with upsell messages
- parking, paging and call forwarding
- integration with Outlook and CRM software
- customizable caller ID
- click-to-call from your website
- extensive reporting for productivity analysis

With the advanced functions and features offered by a hybrid IP-PBX system, your SMB can gain the efficiencies of IP telephony and cost savings of VoIP today without having to sacrifice the quality we have all come to expect from the POTS!

What Will it Cost?

As you upgrade towards VoIP, your costs will vary based on which networking approach you choose and the relative cost of broadband Internet access in your area.

Equipment: If you choose to upgrade your LAN with a QoS switch, plan on spending about $400 for a minimum quality device and up to $2,000 for a very good one. If you decide to take the premium approach by adding a second LAN for IP telephony, you'll probably spend $80-$100 per employee for an additional Ethernet "drop".

Broadband services: DSL may be sufficient for a low volume of VoIP calls; with costs ranging from $40 for consumer DSL to $150 for business-grade DSL, depending on your location. T1 lines, which can handle up to 20 VoIP calls simultaneously, cost from $400-$800 per month, again depending on your location. For a lower VoIP calling volume, you may be able to save some money with a fractional T1 line, if that level of service is offered in your area.

If you have remote offices or workers who telecommute, remember that you may have to upgrade their routers and/or broadband connectivity. You can get a reasonable VoIP-capable router that can adequately handle a few concurrent calls for around $50-$100.

About the Author

Chris Lyman is the CEO and founder of Fonality. Chris combines his extensive telephony and VoIP knowledge with his data-centric roots as founder and CEO of Virtualis, a top ten web hosting company, which he sold in 2000. He often presents at industry forums and conferences on topics such as IP telephony, data architecture and open source business models.

About Fonality

Fonality is the leading provider of affordable hybrid IP-PBX systems for SMBs. Fonality's PBXtra product line comes VoIP-ready but also supports calling via the PSTN and PSTN-fallback. PBXtra Standard and Call Center editions provide enterprise-class features at a fraction of the cost of traditional industry offerings. Deployed to tens of thousands of business users in the U.S. and other countries, Fonality's products can be purchased direct or through a network of more than 1,300 resellers. For more information, visit www.fonality.com.