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# VoIP

VoIP (Voice over IP) is a term used in IP telephony for a set of facilities for managing the delivery of voice information using the Internet Protocol (IP). In general, this means sending voice information in digital form in discrete packets rather than in the traditional circuit-committed protocols of the public switched telephone network (PSTN). This guide will help you understand how VoIP works and what to consider before migrating to VoIP.

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## Learning Guide

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## Section 1: Voice Over What??

by John Blake, December 20, 2005

### **The Surge in Interest in VoIP**

Voice over IP has been around for years, but it is only in the last 12 months that it has started to gain traction, moving beyond the early adopters to mainstream customers beginning to consider it as a viable alternative to traditional telephony. The impetus to upgrade to VoIP is coming from many different angles, not least from customer demand and manufacturer and service provider commitment to the new generation of voice services. The fact that it is now a 'hot issue', reflects the coming of the 'all IP' phenomenon in the corporate sector and the growth of broadband among smaller businesses and the residential market. So is this renewed enthusiasm for the technology proof that VoIP has thrown off its old reputation as an unreliable and second-rate voice service? The answer is that it is certainly starting to, but key to this lies in raising awareness that VoIP is not a stand-alone technology, but dependent upon the network that delivers it.

### **The Platform for VoIP**

Despite its name, VoIP can be run over a number of different network technologies. To illustrate this point, let's look in more detail at the different types of networks VoIP can be delivered over and why companies are opting to implement them.

### **The IP Era**

The trend towards an 'all IP' communications infrastructure is well underway. Established network providers around the world are investing in IP and looking to new revenue streams in the digital networked economy through new networked IT services. In the corporate sector, many large international companies already have an IP Virtual Private Network (VPN) in place. These networks enable swifter and more flexible communications and are designed to carry advanced IP applications. They also have quality of service (QoS) technology, which ensures that mission-critical traffic, such as IP voice packets, is not disrupted in the local area network (LAN) by other less important traffic like e-mail. Multi Protocol Label Switching (MPLS) based IP VPNs have class of service (CoS) technology, which categorizes traffic by importance in to separate channels and ensures that as traffic travels on to the wide area network (WAN) it continues to be prioritized. So, for companies with an IP platform in place, is the decision to switch to VoIP simply a 'no brainer'?

Running all traffic over a single converged network and thus having a single point of failure brings its own complexities. However, increasingly companies are considering the productivity, time and cost saving benefits to outweigh the risks of network downtime. To counter this risk, managed network services will include resilience planning, and other measures designed to eliminate single points of failure. One example of an organization already reaping the benefits of having a single network for all its communications streams is business information company, Datamonitor. It upgraded its network to a managed MPLS-based IP VPN for both its voice and data traffic between key sites in the U.K. and U.S. By implementing this managed service, Datamonitor was freed from the need to invest heavily in, or run, its own infrastructure and has cut the costs of calls between these sites by 50 percent.

But the switch to VoIP over a dedicated IP network is not just being made once the corporate data network has been upgraded. Having carried out a technology refresh at the end of 1999, to ensure their infrastructure would not be hit by potential 'millennium bugs', many companies are nearing the end of their five-year PBX lifecycles. They are therefore faced with the question of whether to renew their standard telephone exchange or to invest in an IP PBX, a more 'future proof' solution that supports next generation IP telephony. With pressure from manufacturers, who are increasingly announcing plans to phase out the production of traditional PBXs to focus on IP products, the impetus to move to VoIP is an important driver behind the decision by companies to migrate to a converged network.

### **Convergence Solutions, Pure IPT, and Hosted and Managed Options**

Once the decision to implement VoIP via a dedicated IP network has been made, the options open to companies are twofold: gradual migration via a convergence VoIP product or complete migration to a pure IP telephony environment:

- Convergence VoIP products connect traditional digital TDM phones via a PBX to a gateway, which turns TDM speech into IP for transport over the IP network. For many companies, gradual migration to VoIP via an IP gateway makes financial sense since it does not require huge upfront investment in expensive IP equipment. It also offers the company's employees the chance to adapt culturally to IP voice communications, a consideration that is not to be underestimated since for many the notion of picking up a phone and not hearing a ring tone can take time to come to terms with!
- Pure IPT is the most advanced and future-proof form of VoIP, where all elements are IP based. This involves overhauling a company's telephony infrastructure and installing new IP equipment – including the applications, PBXs and the phones themselves. Pure IPT supports advanced IP communications such as video conferencing, file sharing and white boarding (providing the right software is in place). It is also the most forward-looking form of the technology enabling companies that embrace it to derive competitive advantage. For organizations with a mainly office-based work force, including call centers, where the quality of voice and video communications are imperative to the success of their business, full migration to pure IPT is a sound choice.

Abbey, one of the largest U.K. high street banks, recently installed a pure hosted IPT service that incorporates Cisco technology, and expects to save millions of pounds over the five years of the contract by putting telephony and data over a single network. This is an example of an IPT product, which is hosted off-site by a carrier using IP Centrex. By using a hosted service, companies can avoid investing in an IP PBX since this is hosted on the service provider's site. For companies without a skilled IP networking department making large-scale IPT migrations, a hosted service is to be recommended.

### **Voice Over DSL Broadband: the Other Route to VoIP**

But IP is not the only technology that is luring customers to VoIP. DSL broadband in the form of symmetric digital subscriber line (SDSL) or asymmetric digital subscriber line (ADSL) also enables cost-effective transmission of voice over a data network. The U.K. alone already has six million broadband subscribers and the success of broadband voice providers like Skype and Vonage have helped to raise the profile of this flavor of the technology. In addition, we are seeing consumer market technology pushing functionality into the corporate space - many corporate workers are discovering new VoIP technologies at home and are expecting to see the same functionality at work. For many

small and medium sized businesses, particularly those whose workforce is not predominantly made up of office workers or whose business model does not rely on voice communications, broadband VoIP is a good choice. It is very easy to install, requires minimal investment, offers a converged environment that allows employees to use voice, instant messenger and Internet at the same time and can achieve impressive cost savings on voice calls. There are a number of enterprise specific broadband services available. Companies considering broadband VoIP, however, should be aware that it does not offer traffic prioritization capability and its voice quality can diminish if multiple users make calls at the same time.

### **The Choice of Voice**

So to summarize, transmission of voice over broadband and IP networks looks set to become prevalent in corporate communications. Ensuring the quality and efficiency of these communications is key to maintaining a competitive edge in today's digital networked economy. With all the variants of VoIP technology available, choosing the right solution is a critical task. Companies looking to upgrade to packetized voice must first consider what network they need to guarantee quality calls are delivered and then carefully choose between convergence or pure IPT solutions, DIY or fully managed options. Making an informed decision will ensure that their corporate communications requirements are not merely met, but superseded.

## Section 2: VoIP Analysis

Terry Slattery, December 23, 2004

VoIP is the new big thing in networking. Best-effort delivery of packets is not good enough for VoIP, so the network must be stable and predictable to provide quality comparable to circuit-switched telephone networks. The challenge is to provide the reliability and consistency that's needed.

### Call Quality Factors

VoIP call quality is primarily affected by three parameters: delay, jitter, and packet loss. Combinations of these factors affect the resulting call quality. Increasing delay affects the interaction of the parties participating in the call. One-way delay of up to 150 ms is generally considered to be acceptable. Of course, lower delay increases the level of interaction.

Jitter is the variation of the amount of delay. Let's say that a VoIP call is operating over a path that has a one-way delay of 30ms. Most packets will arrive at the destination about 30 ms after leaving the sender. But because other data packets are queued on the same interfaces as the voice packets, an occasional voice packet arrives after a delay of 35 ms. The jitter, or variation in delay, is 5 ms. Voice calls deteriorate as jitter approaches 30 ms. At some point, high jitter results in a packet arriving after the audio time slot in which it should have been played back. A burst of silence sounds a lot like a pop unless a whole stream of packets are significantly delayed, in which case it sounds like interrupted speech.

Dropped packets or damaged packets are just like very high jitter. The output audio stream is silent for the time slots of the lost packets. More than 1% of lost or damaged packets is unacceptable. Any regular volume of dropped or damaged packets should be examined and the cause determined. The most likely cause of dropped packets is due to output drops on an oversubscribed link. Damaged packets may occur on switch ports where a duplex mismatch has occurred.

Finally, network stability is a major factor in whether a network will support VoIP. Routing changes will typically result in jitter. Spanning tree topology changes will result in outages of 30 to 60 seconds if the default spanning tree timers are used. A network where QoS is not consistently implemented may exhibit intermittent symptoms of delay, jitter, and lost packets as data packets are queued in front of VoIP packets.

Identifying the source of each factor that affects voice call quality is important to a high-quality and smoothly running VoIP implementation. Performing end-to-end testing will tell you the characteristics of the path taken at the time of the testing. A network that is lightly loaded and not properly configured for VoIP may pass the tests and yet will fail after a period of time as data traffic builds and begins to compete with voice traffic. The dynamics of the interaction between voice and data require that the network infrastructure be checked for proper configuration as well as validating the basic characteristics of voice traffic.

**Analyzing Data**

Delay, jitter, and packet loss can be measured by conducting active tests or by collecting call detail records. A VoIP analysis tool can perform active tests without adding software or probes to the network. Often, it is a good idea to install a dedicated router to act as a centralized responder for the tests.

Another source of delay, jitter, and packet loss information is the call detail records (CDRs) of each call placed on the network. Cisco's Call Manager can be configured to collect these statistics from VoIP phones. An analysis tool sorts the values for delay, jitter, and lost packets to identify the worst calls. You can then isolate the paths of the calls and determine whether the poor characteristics are intermittent or constant. Once the call path is known, further analysis can be performed on the routers, switches, and subsystems in the path.

## Section 3: Duties of an IP Telephony Gateway when Integrating VoIP and the PSTN

Rich Parsons, October 21, 2004

In order to integrate a Voice over IP (VoIP) solution with the Public Switched Traditional Network (PSTN), an IP telephony gateway is used to convert the traditional voice traffic (analog) to digital data so that it can be transmitted over a data infrastructure. This tip explains the various features and responsibilities of an IP telephony gateway.

### Destination Mapping

Traditional voice calls are routed based on country code, area code, and exchange and line numbers through Local Exchange Carriers (LECs), Inter-exchange Carriers (IXCs), and exchange switches. However, the VoIP gateways that receive these dual-tone multifrequency (DTMF) tones via PSTN signaling are looking to map the 10-digit number to the IP address of a gateway that can terminate the call. Within each gateway area code/exchanges are mapped to other gateways' IP addresses. Based on the size of the network, these mappings can range from simple to complex.

### IP Connectivity

Once a gateway determines the remote gateway(s) necessary to complete a call, a "connection" must be made over the packet network. Today, there are two prominent standards-based protocols available to establish and maintain VoIP connections: the ITU-T H.323 specification and the Session Initiation Protocol (SIP). These protocols provide the suite of functions in order to make VoIP calls.

### Voice Digitization and Compression

The most prominent task required of the gateway is to convert the voice signal to a digital one in order to be transferred through a digital network. Usually, the digitization takes place in data rates of 64 kbps. IP Telephony gateways can additionally compress data rates for voice calls to a rate of 24 to 5.3 kbps per call. From a bandwidth utilization perspective, the gateways can provide up to 10 times the number of calls as compared to digital calls transmitted over a PSTN infrastructure.

### Decompression and Reconfiguration

If IP Telephony Gateway device functions as the IP connectivity endpoint to complete a call, the voice traffic must be decoded. Although the decoding is not necessary for the sender to transmit the call, it is necessary to decode the voice traffic so that the telephony equipment (fax or phones) can receive the correct signaling.

### Authorization, Access, and Accounting

Similar to the functionality of a PSTN switch, the VoIP gateway is responsible for assuring the proper security, system access and billing records for used services. From a security perspective, the gateways are responsible for not allowing specified phones to have control input. From an access control perspective, the gateway must only allow those services contractually agreed upon to the end user. From a billing perspective, the gateway must be able to provide the various billing models used in the PSTN networks today. Two examples of billing models are monthly fixed fee agreements or "pay per usage" plans.

## Section 4: VoIP: The Face of the New Network Police

Loki Jorgenson, May 26, 2005

In search of supreme network performance? Try deploying VoIP. Considering its reputation for creating serious network problems, it may not seem obvious that VoIP can actually show you how to make your network perform better. But it can and here's why:

Your network is a relatively complex form of ecology. There are devices of all sorts like routers, switches, client stations, servers and the like. Then there are the interconnecting media and interfaces such as cables, NICs and drivers. And there is the traffic and its various behaviors such as congestion and burstiness. Finally, there are network configurations, from interface settings to routing, that define the individual and aggregate behaviors.

Network performance is a function of all these elements and relies on very subtle interplay between them. For example, very low packet loss (< 0.1%) on a very high latency link can result in performance degradation for applications using standard TCP—an unintentional consequence of TCP's slow start mechanism over long network paths. No single element of the network is entirely responsible—the low packet loss would be considered reasonable under any other circumstances.

Even the devastating problem of duplex mismatch is defined by the not-so-subtle interaction between two interfaces — neither interface is actually malfunctioning (although it might be argued that auto-negotiation has a fault).

So if your network is a kind of ecology, then, Voice-over-IP plays the role of the amphibian. It is well known that frogs are highly sensitive to toxins in their environment and their disappearance can signal a serious ecological issue. If you can keep the frogs alive and well, chances are that most everything else will be healthy and happy, too. The same theory rings true for VoIP—to keep that VoIP frog alive, your network has to be clean. Very clean. And coincidentally, that is the key to high performance of all your applications.

A majority of performance problems are rooted in the degradation issues that also impact VoIP. Networks that cannot perform at 100% keep applications from operating at 100%. There is no way to tune an application to overcome a bad network.

Typical sources of network degradation include:

- Congestion
- Lossy media (wireless)
- Bad cables or poorly seated cards
- EM interference and poor shielding on copper lines
- Duplex mismatch
- Bad NICs/drivers
- MTU mismatch (RFC 1911 non-compliant)
- Mix of duplex domains
- Unplanned bottlenecks

While most typical data applications will work reasonably well in the presence of most of these—this is where the frog comes in—VoIP will not. And end users will clearly hear these problems. Even small amounts of loss and jitter are noticeable, forcing engineers to either resolve the degradation issues or compensate with QoS mechanisms. So, if you were wondering how VoIP was going to fix everything—it won't.

The good news is that the pressure to move to VoIP will force the development of tools and technologies capable of resolving hard-to-find network degradations. Allied Business Intelligence estimates that enterprises migrating to voice from traditional circuits will create a \$16.5 billion dollar IP-PBX market worldwide by 2006.

The bad news is the hard work has yet to be done—only 12% of corporations have deployed VoIP and another 29% are expected to join in next year (Harris Interactive)—and we don't have all the tools we need yet. However once we deal with the uniquely sensitive nature of VoIP, the worst will be behind us and we will have reaped the benefits for many other network applications.

So clean up your networks and keep the frog alive.

## Section 5: IP Telephony Implementation Planning

Ramesh Kaza and Salman Asadullah, June 9, 2005

### Getting Started

The first step in the planning phase is to understand the high-level business and technical expectations and requirements for the future IPT network, which include the following:

- Company vision, goals, and forecasted growth
- The plan for voice and data networks over the next three to five years
- Solution expectations
- Deployment and timing
- Financial expectations

To simplify the discussion for this case study, assume that XYZ expects its workforce to grow five to ten percent every year. XYZ requires that the new IPT system must emulate the functionality of the current PBX, voice-mail, and application systems, be scalable, and provide additional services and features that improve employee productivity. The new technology update project at XYZ received approval from the company's financial board to support the funding for the IPT project, and there are no major budget constraints.

After you understand the high-level business and technical expectations of the company, the next step is to conduct meetings with the engineers and architects in the LAN, WAN, IT, legacy PBX, legacy voice-mail, and applications network groups. During these meetings with the various groups, you should make sure that any high-level requirements are accurate. Most importantly, make sure that you understand how the existing network infrastructure is built so that you can identify the gaps in the infra-structure that need to be filled to support the converged traffic.

### Network Infrastructure Analysis

Another term commonly used for this analysis is IP Telephony Readiness Assessment. The purpose of this assessment is to check whether the company's network infrastructure is ready to carry the converged traffic. The assessment covers basic LAN switching design, IP routing including power and environmental analysis, and so forth. As a network engineer, you are required to identify the gaps in the infrastructure and make appropriate recommendations before you move forward with the IPT deployment.

The network infrastructure analysis of XYZ is divided into eight logical subsections:

- Campus network infrastructure
- QoS in campus network infrastructure
- Inline power for IP phones
- Wireless IP phone infrastructure
- WAN infrastructure
- QoS in WAN infrastructure
- Network services such as Domain Name Service (DNS) and Dynamic Host Configuration Protocol (DHCP)
- Power and environmental infrastructure

After reviewing the preceding list, you might be wondering why planning for the IPT network includes analyzing campus infrastructure (Layers 1, 2, and 3), WAN infrastructure, LAN and WAN QoS, and network services. The analysis of the aforementioned network infrastructure components is required during the planning phase of the IPT network deployment to identify the gaps in the current infrastructure to support the additional voice traffic on top of existing data traffic. After identifying the gaps, you need to make the appropriate changes in the network, such as implementing QoS in LAN/WAN, upgrading the closet switches to support QoS, and supporting the in-line power.

Legacy voice and data networks are migrating to new-generation multiservice networks. Multiservice networks require a set of technologies, features, and best practices to design a scalable and optimized infrastructure, which carries in parallel over the same IP infrastructure both real-time, delay-sensitive voice and video traffic and nonreal-time, delay-tolerable data traffic (i.e. FTP, e-mail, and so forth).

When you introduce real-time, delay-sensitive voice and video traffic into ensuring that your infrastructure is hierarchical, redundant, and QoS enabled, it becomes even more important to provide a scalable and redundant network infrastructure with fast convergence. Large network infrastructures use the access, distribution, and core layers at Layer 2 and Layer 3 for isolation, with redundant links and switches at these layers to provide the highest level of redundancy. This isolation helps you to summarize the IP addresses and traffic flows at different layers and troubleshoot the issues in a hierarchical manner when they occur.

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## Resources From CA



### [CA Voice Management](#)

CA's integrated, proactive management solutions enable you to count on both your data and voice networks to support mission-critical, revenue-generating operations.

### [UMASS Improves Voice Quality of Service with CA's eHealth for Voice](#)

The University of Massachusetts needed to optimize resources and assure quality of voice services supporting a burgeoning student and faculty base including revenue-generating departments such as Admissions. With over 18,000 students and faculty combined, UMASS needed menu-based applications to alleviate live calls to various departments throughout the university.

In order to improve reliability and performance, the solution would need to provide:

- A campus-wide view of potential security risks
- Performance metrics for monitoring trends
- A real-time view of the entire voice communications device population across the IT infrastructure.
- This case study reveals the voice management solution that UMASS implemented and how it provided them with a return on investment of over 600%.

### [Continental Improves Voice Messaging with CA's eHealth for Voice](#)

With over 41,000 employees worldwide and a business dependent on easy to use and broadly accessible voice messaging, Continental Airlines needed a management solution that would keep business moving, through smooth-running voice messaging.

This case study discusses Continental's decision to invest in a voice management software solution that could perform the following functions:

- Handle the large volume of data generated
- Carry out the bulk of the day-to-day analysis of voice mail performance
- Help the staff decide when to intervene using human intelligence to address specific performance issues uncovered

Read this case study to find out how Continental was able to use this solution to reduce time spent creating and analyzing data by 80%.

## About CA

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