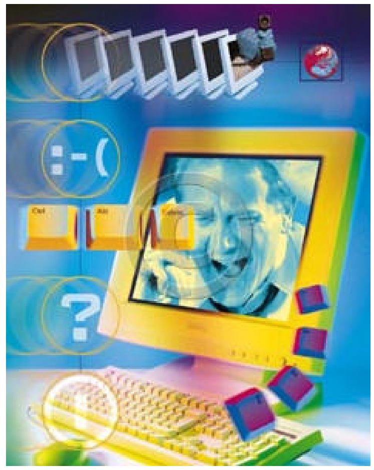
VOICE OVER INTERNET PROTOCOL OVERVIEW



INTRODUCTION

Everyone is talking about it, and you might even be doing it from time to time. What is it? **Voice Over Internet Protocol (VoIP).** Using the telephone over the Internet has become more popular and more important than ever. Whether, you are using it for an important meeting conference or just talking to you friends, VoIP has become the communications media of the future.

Recent improvements in the quality of voice traffic transmission over Internet Protocol (IP) promise a cost-effective alternative for businesses selecting an efficient communication system. Voice over Internet Protocol allows users to make free telephone and fax calls over IP connections (i.e. corporate data networks, Intranets and the Internet). VoIP technology works by creating an overlay voice/fax network on top of any existing IP connection. The voice or fax traffic is enclosed in an IP packet and sent over the IP network. The products themselves range in complexity from software packages that pass digitized voice to a specified destination in a network, to combined hardware/software products providing directory services and complex voice quality assurance capabilities. Businesses are increasingly choosing the latter due to this superior flexibility and greater functionality.

HOW DOES IT WORK

While IP has historically been used for data transfer and is optimized for data traffic, there are several ways of transmitting voice over an IP network.

Internet Telephony

Much attention has been given to Internet telephone by using a microphone and speakers connected to a computer to communicate over the Internet without paying long-distance charges. Those who have used Internet telephony say that the audio quality is poor. The problems with voice-over-the-Internet are problems with two-way communications. One-way voice transmission, voice messaging and broadcast audio can be sent over the Internet well.

What will it take to make Internet Telephony work?

Assigning a higher priority to voice connection will make VoIP better. The network protocol used by the Internet, intranets, and many local area networks was not designed to provide a way to give voice traffic higher priority than other types of traffic, such as e-mail. Internet congestion causes slow voice transmission, which causes poor audio quality.

Several methods have been proposed to solve this problem. Resource Reservation Protocol (RSVP) provides a way to reserve IP bandwidth and give delay sensitive traffic such as voice and video, priority over data traffic. RSVP requires that every router along the data path provide special support for RSVP. Since RSVP is relatively new, most routers on the Internet need to be upgraded or replaced. Considering that there are over one million register domains on the Internet, the number of routers that would need to be upgraded is very large.

Increasing Bandwidth

Another method for improving audio quality would be to increase the bandwidth of the Internet to prevent congestion. Unfortunately, Internet traffic is growing faster than the capacity to handle it. Internet service providers are struggling to provide enough bandwidth for e-mail and Web traffic. The excess bandwidth requires improving Internet audio quality is presently beyond their reach.

VoIP uses the real-time protocol (RTP) to help ensure that packets get delivered in a timely way. Using public networks, it is currently difficult to guarantee Quality of Service. Better service is possible with private networks managed by an enterprise or by an Internet telephony service provider (ITSP).

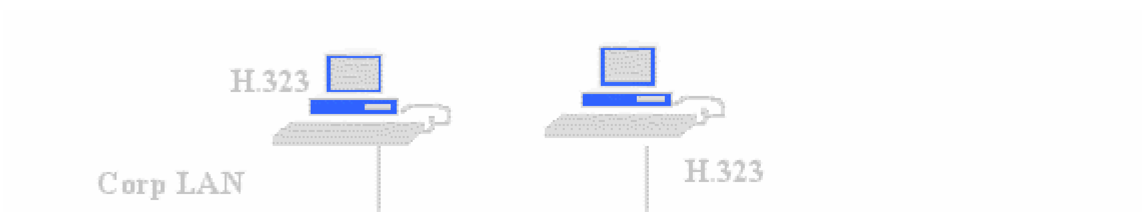
<u>H.323 Protocol</u>

The main issue for computer communication centers on standard ways of providing connectivity from call control (finding other parties, ringing, etc.) to video and audio encoding. Intel believes that real-time, multimedia communications standards such as H.323 provide an unparalleled ability for compatibility and subsequent expansion.

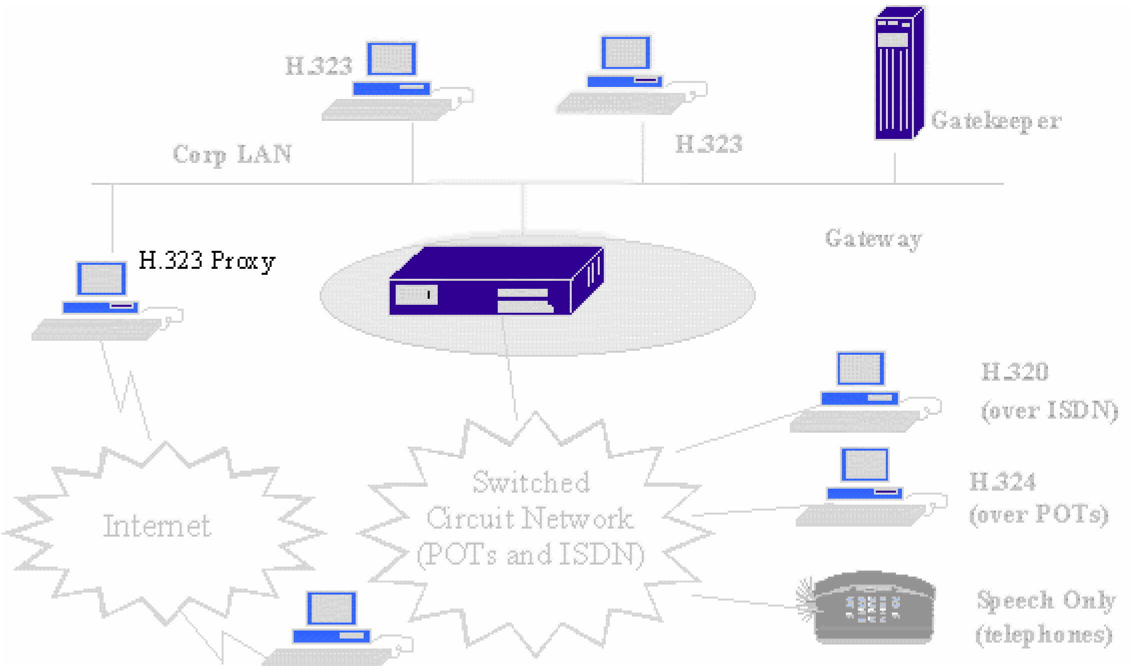
The ITU-T (International Telecommunications Union) H.323 standard defines how a flexible, real-time, interactive set of multimedia communications can be exchanged on packet-based networks. Personal computers can inter-operate sharing a variety of audio, video and data across all forms of packet-based Inter/Intranets and circuit-switched networks. This standard is the first standard that was provided through the collective input of both traditional telephony communications experts and those from the computer communications arena. In addition to fully interactive media communications such as conferencing, H.323 also has provisions for other forms of communication, such as multimedia streaming.

H.323 Network

The diagram below shows the H.323 configuration on a network. It covers two or more clients that run a multimedia application using the H.323 protocol. Any combination of audio, video, or data may be exchanged between the computers. Implementations on IP networks need nothing more from the network than to provide IP addresses of each computer.



Although H.323 requires little for the simplest implementation model, the standard does define a number of H.323 protocol interactions. The protocols or entities such as Gatekeepers, MCs, MCUs, Gateways, and Proxies each provide some additional functionality to the H.323 standard. The picture below illustrates these protocols or entities and their relationship.

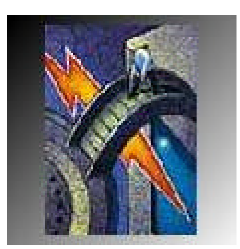


H.323 Standards

H.323 standard was developed from the H.320 standard. This standard was the original ISDN videoconferencing standard. The H.323 is an extension to this standard. It is for videoconferencing over local area networks. The H.323 standard is made up from of the following standards:

- H.225 - call control protocol
- H.245 - media control protocol
- H.261 - video codec for >= 64kbps
- H.263 - video codec for < 64kbps
- G.711 - PCM audio codec 56/64 kbps
- G.722 - audio codec for 7 KHz at 48/56/64 kbps
- G.723 - speech codec for 5.3 and 6.4 kbps
- G.728 - speech codec for 16 kbps
- G.729 - speech codec for 8/13 kbps

The H.323 standard was created mainly by the International Multimedia Teleconferencing Consortium (IMTC) and are distributed by the International Telecommunications Union (ITU). The project to develop this standard started in May 1995. In June 1996, version 1 was approved by ITU-T. In January 1998, version 2 was added with functions. In 1999, version 3 was introduced.



Gateway

Gateways allow interoperation of H.323 systems with other audio/video conferencing systems on integrated services digital networks, plain old telephone systems, asynchronous transfer mode and other transports. They provide worldwide connectivity and interoperability from LAN using H.320, H.324, regular plain old telephone system (POTS) telephones. It can track call signaling (Q.931 to H.225.0), map control (H.242/H.243 to H.245), and media mapping.

A gateway provides the ability for H.323 devices to operate in network environments. These environments can be different in both the communications protocols that they use, and the fundamental transports that carry those communications. Gateways provide interoperability with other standards-based media devices. The H.323 standard pre-defines a number of gateway devices. Gateways are defined for H.320 (ISDN-based Video Conferencing terminals), H.324 (Telephony-based Video Conferencing terminals), and POTS (Plain Old Telephone System) devices. Gateways also provide the ability to match up different devices (for example, a plain voice-only telephone with a multimedia terminal).

The gateway is an element to the H.323 Protocol. The main operation uses of gateways are 1) connection set-up to telephones, 2) connection set-up to video conferencing systems, and 3) connection set-up to video conferencing systems over modem lines. They also provide a function between H.323 terminals and non-H.323 terminals. The gateway also transforms the coding formats, mainly audio, and performs the connection control for the LAN side, as well as the WAN side.

A VoIP gateway bridges calls between the PSTN and an IP network. Gateways support two different types of interfaces, analog and digital. An analog interface connects to the PSTN with the same interface as a regular phone. Analog interfaces are almost identical worldwide, and are probably the easiest way to connect a gateway to the PSTN. On the other hand, they are not a good solution, and do not have good support for some calling features, especially when it comes to call progress indications such as disconnect signals and caller ID.

Gateways are only used if connections are established in different networks. There is no need for them if you are communicating within the same LAN.

Gatekeepers

Gatekeepers provide admission control and address translation to H.323 endpoints. A gatekeeper acts as a monitor of all H.323 calls within the LAN. It provides two main services: call permissions and address resolution. An H.323 client that wants to place a call can do so with the assistance of the gatekeeper. The gatekeeper provides the address resolution to the destination client (this division of work is due to alias name registration procedures). During this address resolution phase, the gatekeeper may also make permission decisions based upon available bandwidth. The gatekeeper can act as an administration point on the network for IT/IS managers to control H.323 traffic on and off the network. The gatekeepers work with the gateways. While gateway transfer calls between the PSTN and IP networks, gatekeepers provide the “brains” for a VoIP network. Among other things, gatekeepers provide network security by preventing unauthorized usage. As with gateways, gatekeeper redundancy can prevent downtime. As a first step, component “duplication,” such as hard disk raid array and backup power supplies within a single gatekeeper, are a good start for realizing sufficient redundancy.

With the gatekeeper in address translation “email like” names are possible, and “phone number like” names are possible. The gatekeeper gives permission to complete a call and can also set limits for bandwidth. In addition, it sets up methods to control the LAN traffic.

Multipoint Control Unit (MCU)

A Multipoint Control Unit provides the ability to have multiparty, multimedia conferences. It can emerge more than two H.323 computers in a centralized multipoint conference. It coordinates all of the media capabilities of the participants and can provide features such as audio mixing and video selection for computers that cannot accomplish this locally. The MCU can provide chair control capabilities in multipoint conferences. It also makes it easy for entrance and exit of conference participants.

Developers use enabling software to build MCUs, gateways and terminals. The H.323 standard provides a framework and does not define specific implementations; therefore, developers can use H.323 software to develop H.323 compliant products.

A MCU supports simultaneous conferences between three or more locations. In the H.323 standard, multipoint session dynamics are very flexible. The standard allows for a variety of ad hoc conferencing scenarios in addition to the traditional method of scheduled resource usage found commonly in H.320 based systems. The MCU also controls conference resources by determining how the audio and video streams will be multicast.

VoIP Signaling

VoIP signaling has three distinct areas, signaling from the PBX to the router, signaling between routers, and signaling from the router to the PBX. The corporate intranet appears as a trunk line to the PBX, which will signal to corporate intranet to seize a trunk. Signaling from the PBX to the intranet may be any of the common signaling method used to seize a trunk line, such as FXS (Foreign Exchange Station) signaling.

Quality and Characteristics

Not only is the H.323 protocol used to telephone over the Internet, but it is also used for audio, video, and data transfer. For audio, the latest mixed media call servers limit end to tend voice delay to less than 400 milliseconds, or below the normal range. A silence suppression features transmits only data grams containing speech, and doesn't transmit when there is silence. For video, an advanced mixed media call servers use lip-synchronization features to coordinate audio and video. Last, for data transfer, it is important to consider in data conferencing to synchronize updates that the user controlling the file or application is making at the same time. Data can also be split routing some traffic over the data WAN and other traffic (mainly audio and video) goes over the PRI WAN.

Benefits and Drawbacks

Costs

The business of helping people connect their phone calls over the Internet is still relatively small, ranging in the tens of millions of dollars annually. In a few years it could rake in billions of dollars in sales of hardware and alternative phone services. The big money has attracted some attention. Long distance service providers such as MCI and AT&T Corp., which were concerned early that Internet telephony could wear away their profits, have begun investing in it. Some nations that profit from their government owned phone monopolies have moved to block Internet telephony.

They are concerned they will lose major revenue gained from fees known as settlement rates, which many governments add to regular long distance phone calls. These fees inflate the prices of many international calls over the true costs of providing such services. These governments stand to lose money if overseas phone calls over the Internet become popular.

When a traditional phone call is placed, it is routed over what is known as the public switch telephone network, a system of dedicated phone lines, antennas and satellites. Charges for using the system are generally distance based with long distance calls costing more than local calls and international calls the most expensive.

The Internet could care less about distance. When someone makes an Internet call, he or she uses a computer or a phone that accesses the Internet through a service provider. From here, special software converts the voice into digital information packets that are bounced around the global computer network like other Internet data, such as World Wide Web pages, until they reach their destination.

VoIP promises to reduce cost by providing voice access over the existing data communications network. This cost reduction is seen through a reduction in required equipment. For example, a company is only required to provide data communications equipment, rather than both voice and data, and by a decrease in per minute usage charges (e.g. Internet access by a common ISP is generally cheaper than long distance charges. Recent legislation before the FCC may change that, as long haul carriers lobby to enable per-minute charges for Internet access).

IP telephony is efficient. An IP voice conversation requires less than 10% of the bandwidth of plain old telephone service. A compression telephone technique such as G.723 compress the 64 kbps plain old telephone service takes to 6.23 kbps. Its true that the 6.23 kbps grows when adding the IP overhead of about 40 bytes per packet, but overall reduction of 6-to1 is realistic. POTS requires full duplex equivalent to 64 kbps in both directions-to support a telephone conversation. That feature is wasteful because in conversations, the only one person is speaking most of the time. VoIP products sense the silence to cease transmission when one party is quiet. This technique almost halves the required bandwidth. IP telephony commonly takes as little as one-twelfth the bandwidth of POTS to transmit conversations.

Another cost savings for corporations is using IP Telephony to reduce maintenance. Most companies having IP base intranet and PBX telephone network running side by side. Merging the two networks can ease system monitoring and software upgrades and can save costs in the long run. At corporate level, IP telephony lets corporations integrate e-mail, fax, pager, and voice mail allowing telecommuters and business travelers to collect and forward multiple message types during one connection.

Enterprise Benefits

VoIP calling provides clear benefits for the enterprise. It lowers end user training costs because of the easy to use integration of calling and messaging tasks. It also consolidates all traffic onto a single s TCP/IP network. This keeps total cost for ownership down. In addition to these benefits, there will be less network administration involvement because there is less complex network configuration. As mention before, there will be a significant amount of savings from long distances calling if the enterprise uses the backbone network for voice communications.

Advantages Over Freeware

There are many things that need to be done in order to make an Internet call with freeware. The following are some steps for this process:

1. 1. Look up team members number
2. 2. Call to find out the team members IP address (or learn which directory server has the listing)
3. 3. Make another phone call to check if the person is there
4. 4. Enter the IP address or directory server
5. 5. Try talking
6. 6. Contact the person over the regular telephone network to actually speak to him/her.

As you can see it is very tedious and repetitive, just to talk over the Internet. Many people are dropping their use of freeware. They are giving it up for a specialized task like data conferencing.

Voice Delay and Jitter

Voice delay is a major challenge confronting VoIP technology. Voice delay also called latency is the time delay from the moment a speaker utters word until the listener actually hears the word when using IP telephony system. Delay is caused by several factors such as distance, Gateway Incurred Interference Latency, and network incurred latency.

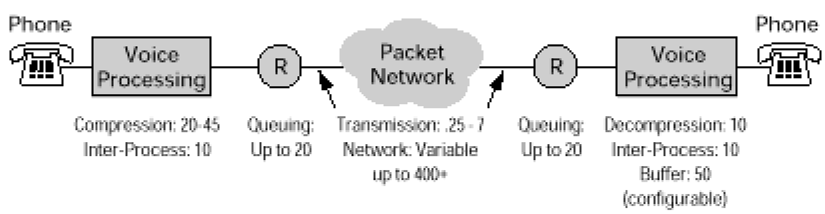
A major cause of delay in voice networks is distance. Propagation delay is the time required for a signal carrying voice to travel from the physical networks medium. In short distances, propagation delay is not noticeable. The greater the distance, the greater the delay will be.

Gateway Incurred Latency

From telephone to another, a voice conversation travels through a network interface in a gateway to the telephone networks. The network interface frames and connects the digitized audio PCM data streams into the internal PCM bus for transport to the Digital Signal Processor (DSP). The DSP performs a process called voice coding or "vocoding". Voice coding includes compressing or decompressing speech deleting silence, generating tones and canceling echo. For vocoding to perform efficiently, DSP implementation depends on entire frame data processing at one time. But processing data in frames has a negative side because none of the data can be processed until the frame is full. So the size of the frame used to process data will affect the amount of latency. From the DSP and passing the data to the Wide Area Network (WAN), packet-handling processes will occur that affect the system latency. After the voice coding processes, the system further buffer the compressed voice data frame before passing them to the network system. Buffering is intended to reduce the number of times the DSP needs to interface with the main gateway CPU. After buffering, the coded voice is assembled into packets, is prepared for transport over the WAN. This process is done by TCP/IP protocol stack using User Datagram Protocol (UDP) and Real Time Protocol (RTP). A typical IP Telephony packet starts with an IP, UDP, RTP header that totals 40 bytes. After the IP header, one or more frame of coded voice data would follow. Decision has to be made whether to pack more than one frame of data on a single packet because putting more than one coded voice frame per IP packet reduces the inefficiency of the IP packet over head. Because data arrive at a very inconsistent rate, IP networks cannot guarantee the delivery times of data packets. The variability in the arrival rates is called "jitter". All systems need to compensate for the jitter in the data arriving from a network by at least on packet of data before passing it to the DSP. These "jitter buffer" reduce the occurrences gaps in the data and ensure the correct timing when sending the data to the DSP. But the negative side is increased latency.

Network – Incurred Latency

After the gateway has compressed and packetized the voice dat a, it is now passed to the Wide Area Networks for transport to the far-end gateway. It passes through many different physical media to inter connect gateways, routers and other net routing devices. The queuing algorithms used by most routers were designed before the concept of IP. Many existing routers use "Best-Effort" routings that is not latency sensitive voice traffic. It lacks a priority attribute that results in the router delaying all data during congestion situations. Resource Reservation Protocol (RSVP) is IETF's one measure of managing resources with in routers and gateways. It is a new development and majority of current public network cannot yet support RSVP.



The success or failure in deploying VoIP depends in large measure on the performance of the network elements that carry and route the voice packets. Gateways are required to perform the conversion of voice to IP packets fro applications that cross between the PSTN (Public Switched Telephone Network) and VoIP networks. In addition to network element performance, these gateways must process voice reliably under extreme loads.

A methodology is needed to resolve the quality of voice transmission. The H.323 standard covers Internet Telephony but only describes the protocols or handshaking between voice and data and the data packet formats (RTP/RTCP). Equipment manufacturers have proposed using tones for stress testing but it primarily tests the continuity of connections. Tones are useful for testing latency. For a viable VoIP testing methodology should have the following elements:

1. 1. It must exercise or provide stress on all the components of the VoIP gateway such as the codec and other system components.
2. 2. It must provide the basis for audio quality testing
3. 3. It must conform to the quality standards such as Mean Opinion Scores (MOS), used for voice carried by PSTN.

Providing stress on all of the components of the VoIP gateway is required to see if voice connections are degraded or lost when traffic over the IP network increases. Fully stressing the base signal converter algorithms of the codec and DSP and other critical applications such as Voice Activity Detection (VAD) assesses the effect on speech by echo cancellation, latency, background noise etc.

Echo Control

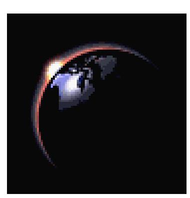
Echoing is annoying and disruptive. Control is key to a successful VoIP implementation; suppression must be present. The Gateway provides full-duplex echo canceling algorithms and a network management system that optimizes audio quality over different types of connections.

Mean Opinion Score

Quality standard, Mean Opinion Score (MOS) is required because VoIP, except in intranet settings does not replace the PSTN network. On the other hand, VoIP is a convergence technology integrating two networks – the PSTN and the data network. Since voice traffic carried over the IP network may originate or terminate over a conventional PSTN local loop, the voice testing methods should conform to PSTN telephony standards. MOS is a widely used subjective measure of voice quality. Scores of 4 to 5 is good, 3 to 4 are acceptable, and less than 3 is poor.

ITU Delay Recommendation
One-Way

Delay	Description
0-150	Acceptable for most user applications.
150-400	Acceptable provided that administrators are aware of the transmission time impact on the transmission of the quality of user applications.
400+	Unacceptable for general network planning purposes although in some exceptional cases this limit will be exceeded.



Management Issues

VoIP networking brings many advantages, it also introduces several management challenges for network administrators. Effective management of any service is essential to its success. Packetized voice networks require a higher level on management because of the stability of the existing voice networks. Transient network delays and errors, which may be acceptable for simple data traffic, are often unacceptable when the network is supporting voice traffic. Also, for any migration to be successful the new implementation must replicate the services available in the existing environment. For VoIP networks, this means that they must successfully replicate not only the high standards for availability, but also the full functionality of existing voice systems including call logging and auditing, monitoring and diagnostic technique, and security.

Availability and Uptime

Most VoIP networks today offer solid availability. The phone systems are simply expected to work or be present all the time. Failures within most phone systems are extremely rare, and failures within the PSTN are even more rare. These services are always expected to function.

Data networks outages are more common. Server and networks related issues still plague enterprise networks and the Internet has instilled in users an acceptance of a best effort service where an individual sites and/or ISP are expected to experience temporary outages from time to time. While users may not expect to have any problems reaching a company via the phone, they are tolerant of slow access to a web site or even a message indicating that the site is temporary unavailable. The data network must be engineered to provide consistent performance with reliability expectations similar to those of existing voice networks. Managing network availability and uptime is critical for VoIP deployments. Existing management tools must be modified to support of VoIP centric view of application network availability.

Configuration

Configuration management is not really a new concern for either data or voice networking environments; however, the specific requirements have changed. The actual configuration of VoIP networking equipment requires attention to details from both environments. Voice specific concepts and implementation must be thoroughly understood, as should the advance data networking concepts.

Most networks today are best-effort environments where bandwidth is often applied as cure-all for networks quality of service (QoS) issues. Until recently, very few QoS techniques were available for IP networks. Engineers familiar with basic IP routing may not be familiar with VoIP QoS requirements or how to configure the IP networking equipment. Queuing and signaling techniques such as WRED and RSVP are new concepts to most network administrators, and must be understood for successful deployment and maintenance of VoIP networks. Managing these services poses a challenge for network administrators, as well as existing network management applications that do not address the issue.

Monitoring and Diagnostics

Monitoring voice activity and diagnostics problems in a VoIP environment requires a diverse set of skills. Three categories of monitoring and diagnostic skills are required: basic telephony connectivity, IP and data network connectivity, and VoIP specific connectivity. Management application tools and procedures need to be developed and deployed to cover all of these areas.

Performance Management

Network performance is an important element in supporting voice calls over a data infrastructure. Oversubscribed circuits and congested WAN and LAN links lead to network delay and inconsistencies which result in poor voice quality and/or missed calls. Proactive network performance monitoring must be performed in order to keep a handle on network conditions and engineers must continually design around potential trouble spots. In addition to network performance monitoring, voice call performance mush also be monitored. Statistics such as call completion rates and voice quality parameters must be collected, stored, and regularly reviewed to ensure that the voice network is receiving proper service levels for the data network.

Hardware

Advanced digital signal processors (DSP) perform computer intensive tasks required for voice and date integration. This processor is low cost and performs well when carrying signal processing functions in VoIP systems. This makes it unreasonable for a single processor system to effectively process multiple voice streams and still perform routing and systems management functions. The use of multiple DSP's offloads this functions from the central CPU and allows the system to process additional calls based upon DSP availability and echo cancellation because they process voice traffic in real-time and have quick access to onboard memory.

Application specific integrated circuit (ASIC) development has fast, more complex, and therefore more capable Asics. ASIC are chips that perform a single application, or small set of functionally similar applications. They are highly optimized for their specific functions and generally perform their tasks one or more orders of magnitude faster than a general purpose processor would.

ASIC are preprogrammed to perform an even more limited number of functions even more quickly. ASIC are inexpensive to produce and are used in networking devices including routers and switches to perform such processes as route lookup, packet forwarding, pack classification and inspection, and queuing functions. Their use in these devises results in high performance and low cost. They benefit the emerging VoIP market in that they provide increased bandwidth and better QoS support within networks.

Security Issues

As the Internet has grown extensively, privacy and security service to audio or data users have been lacking. In the past few years, security measures have been taken to improve the situation. The IPsec is being developed by the Internet Engineering Task Force (IETF), IPsec Work Group. The concept of an IP tunnel in terms of how it is used is as follows:

In a broad sense, a tunnel conveys the idea of the secure transport of traffic between two systems across a non-trusted network, or a single link. The actual passing of data traffic from one end of a network to another end of a network, involves simultaneous passing of the security policies existing between the sending and receiving systems. An IP server tunnel refers to all procedures, protocols and encryption methods that ensure the safe passage of the traffic between two systems. This capability is called a security association (SA). SA defines the meta-characteristics shared between two connecting entities and protects the communication process between the two systems. SA defines the destination IP address, security protocol, secret keys, encapsulation mode and Security Parameter Index (SPI). The operations that are to be performed on user traffic are defined by the security protocol cryptographic operators and encapsulation mode. The SA can be unidirectional or bi-directional. SA can also be grouped together. One SA could be used to ensure the integrity of the traffic. The rule regarding the use of SA bundles is that the destination address of SA must always be the same.

Firewall

A firewall is a system, a collection of hardware and software that controls access policy within organizations. It intercepts the unauthorized traffic to pass from one side to the other of the firewall. It protects trusted networks from untrusted networks. In some situations, within an organization, trusted and untrusted networks could depend on the need to and the need to protect resources. That is where the concept of internal firewall and external firewall can be an important consideration in building security mechanisms.

Radius

Radius, (RFC 2138) is the Remote Authentication Dial In-User Server published by Internal Network Working Group. This specification defines the procedures to implement authentication server, with a central database that identifies dial-in users and the information to authenticate the user. RADIUS permits the server to consult with other server. It serves as a proxy to the other server. RADIUS does not allow a server to send an unsolicited message to a client. It requires the user to present authentication information.

Diameter

Diameter, used in VoIP Gatekeepers and Call Agents has been created to define common format for security messages. The Diameter Protocol consists of a base protocol and the Resource Management and Accounting Management extensions. It is the base protocol that performs authentication and privacy services, version negotiation feature discovery, coding and decoding of the messages. The base protocol permits to send unsolicited message to a client.

The feature discovery is used to increase the efficiency of configuring clients and servers. This is an easy means to locate the policy server in a network and use Diameter message to retrieve information from the policy server.

The Resource Management extension allows two nodes to share session information about Internet users. If a user needs reserved bandwidth, a designated server keeps track of the user's request. It also keeps track of a user session to ensure that users are not giving their account numbers and domain names.

Accounting extension allows records to be shared and exchanged between users. In a roaming user, two ISPs can use the accounting extension to aid in their telecommunication needs.

<u>Modems</u>

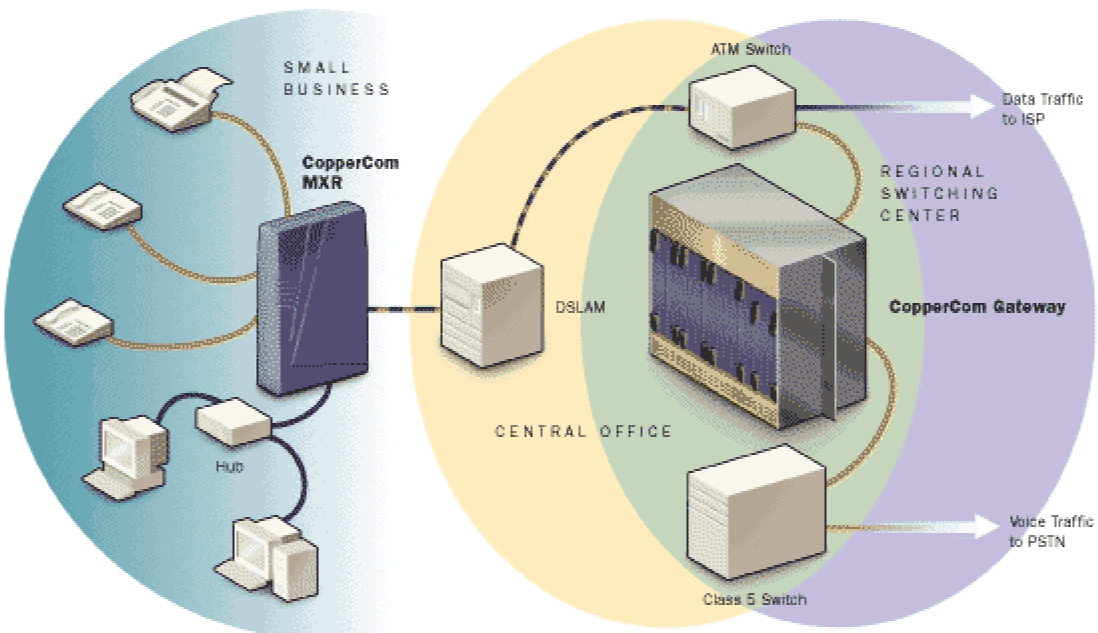
There is a new standard of high speed Internet Connections being developed by Lucent Technology that will further enhance the quality VoIP. DSL is short for Digital Subscriber Line, an asymmetric, broadband modem technology (also known as G.DMT or full-rate DSL) that utilizes standard phone lines for high-speed digital data transfer. DSL allows ultra-fast Internet access and maximizes telecommuter access to corporate networks. It also improves performance of interactive multimedia applications such as multiplayer gaming, video on demand, video catalogs, home shopping, multimedia access and specialized PC services.

Through the use of digital coding techniques, DSL increases an existing phone line's capacity without affecting your standard phone services. With DSL, you could surf the World

Wide Web or retrieve your e-mail while you talk on the phone or send a fax.
The G.Lite modem is a global standardized version of DSL. G.Lite DSL is capable of target data transfer speed is up to 1.5 Mbps downstream and up to 384 Kbps upstream. G.Lite DSL uses a dedicated line, and as such, other users in the neighborhood do not affect its performance. G.Lite is capable of Simultaneous Internet and voice/fax capabilities over a single phone line.

Minimum Requirements for DSL

Processor 200Mhz Pentium Memory (RAM) 16 MB Operating system Microsoft Windows 95 Other requirements NIC[1], with necessary drivers DSL terminal adapter or 1.5 Mb Max Digital DSL/56K Modem.



VoIP is not limited to PC users. E-tel offers a family of open Voice over IP telephones that are fully standards compliant. They allow the OEM to implement their own applications for enhanced features, product differentiation, customer retention and phone top advertising. They are designed for the large OEM or carrier who wants to offer complete VoIP solutions with IP endpoints. Basic telephony features, when the phone is used with OEM gatekeepers, include call waiting, call forwarding, pick up, hold, mute, multiple lines, speed dial, last number redial, programmable keys and call center integration for web based or traditional call centers.

Lucent technology and Texas Instruments have made big and bold announcement regarding (DSP) digital signal processor. Lucent's solution draws upon Bell Labs extensive digital signal processor (DSP) resources and system-on-a-chip experience, much of which was gained in the design of systems-on-a-chip for the cellular phone industry.

In the early days of development, Internet telephony engineers had to "borrow" DSPs that were developed for other telephony applications, such as voice messaging, and through sheer strength of programming extend the DSP's functions and whatever embedded code and APIs were in use in order to build a solution. Since DSPs are specialized chips that are optimized to do a few things exceedingly fast and well, making any kind of changes above and a little beyond the initial design spec took a lot of time and energy.

Migrating Voice Services from Circuit-Switched Networks to Packet-Switched Networks

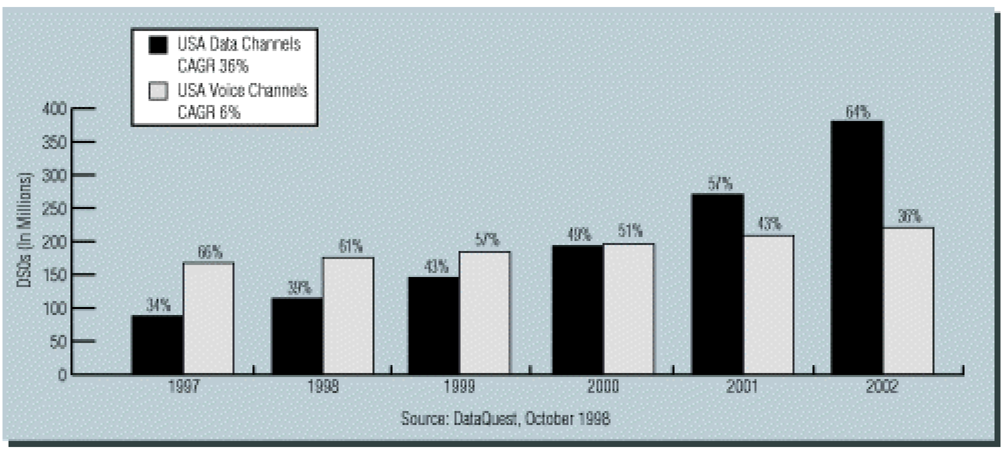
The global communications transformation is in full swing. Packet-switched technology has moved from data-only applications into the heart of the network to take up the functions of

traditional circuit-switched equipment. While the lower cost of packet-switched networks initially drove this change, the improving quality and reliability of voice over these networks is speeding integration of voice and data services. Consequently, the overwhelming majority of voice networks in service today will be replaced by packet infrastructure within the next decade. Service providers, therefore, must develop a plan to migrate their voice services from circuit-switched networks to packet-switched networks to ensure their future success---and survival.
The transition from the old world of circuit-switched networks to the New World of packet-switched networks doesn't just lower costs. It opens the way to a rapid deployment of new voice and data services on a scale that was simply impossible in the past.

Marketing and Future of VoIP

Data Traffic on Service Provider Networks

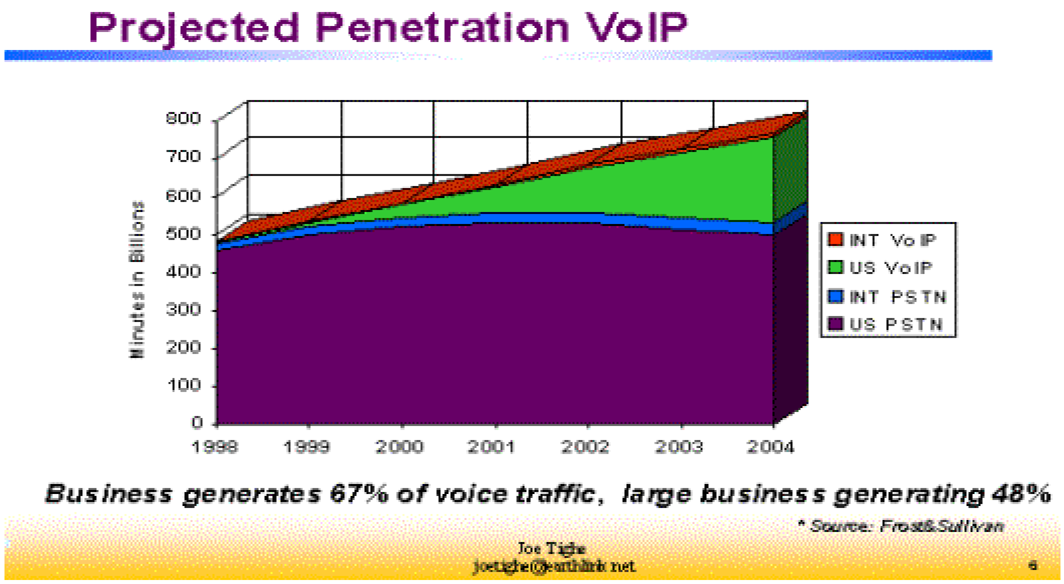
The volume of data traffic is growing faster than that of voice. While growth rates vary by country and carrier, it is certain that data transport will dominate telephony networks. In the United States, data traffic will surpass voice traffic by the year 2000 (DataQuest, October 1998). Data has already surpassed voice on some U.S. carriers' networks, including Ameritech in 1998. It is the driving force behind global network growth.



Cost Savings on Communications

Enterprises seek the savings of toll bypass and the equipment and operations savings of a single, integrated voice-data network. Consumers want low-cost voice, similar to the toll-free cost of their data communications through their PCs. Service providers must therefore drive down costs to compete in a deregulated environment---and remain profitable.

Expected growth rate for IP-enabled telephone equipment is at 132% between 1997 and 2002 with an expected market of some $3.6B in 2002. Annual revenues for the IP fax gateway market are expected to increase to over $100M by the year 2000 (from less than $20M in 1996.



Open Standards

Telecommunications equipment providers have traditionally relied on proprietary protocols and closed development environments to lock service providers into monolithic solutions. This hindered service providers' ability to develop and deploy new services rapidly---and drastically limited service differentiation. By contrast, innovation is extremely fast in the open, standards-based Internet. Deregulation will compel the same pace of change and service differentiation in traditional voice networks. Today's new packet telephony voice services are built from open standards such as H.323, and Simple Gateway Control Protocol/Media Gateway Control Protocol (SGCP/MGCP).

The logic of these market forces is clear: the distinctions between voice and data networks are disappearing. In fact, the vast majority of voice networks deployed today either will evolve to, or be replaced by, a data infrastructure. Therefore, Cisco terms this trend conversion rather than convergence. As they develop strategies for this integrated New World, service providers are asking:

- How can our network best carry data and voice?
- How can we integrate our services?
- How can we retain our subscribers and get new ones?
- How can our network be more profitable?
- How will conversion to packet telephony offer new revenue-based services?

OPT holds the answers to these questions---and much more. However, to fully understand its implications, we must begin with an overview of the traditional network.

Circuit Switched Telephony

The public-switched telephone network (PSTN) is a closed, proprietary, inflexible circuit-switch network. Based on time-division multiplexing (TDM), the network was built for just one technology, voice calls, and one access drive, the telephone.

The circuit-switched network remained essentially unchanged amid enormous technological changes---the explosion in personal computer use, the rapid growth of the Internet, and the development of virtual private networks (VPNs). The growth of intelligent network technology and Signaling System 7 (SS7) did allow---to a limited degree---the ability to develop services independently of the switch manufacturer. However, actual call control remained embedded in the switch as the service switching points (SSPs). This severely constrained service providers' flexibility to offer new services and made it expensive to do so.

BIBLIOGRAPHY

Bates, Regis. *Voice and Data Communications Handbook.* McGraw Hill, San Francisco, 1998.

Bezar, David. *LAN Times Guide to Telephony.* McGraw Hill, San Francisco, 1995.

Black, Uyless. *Voice Over IP*. Prentice Hall PTR, New Jersey, 2000.

Caputo, Robert. *Cisco Packetized Voice & Data Integration*. McGraw Hill, San Francisco, 2000.

Edgar, Bob. *PC Telephony.* Flatiron Publishing, Inc., New York, 1997.

http://internettelephonyexpo.com

http://www.audiocodes.com

http://www.databeam.com

http://www.ednmag.com

http://www.halcyon.com

http://www.networkcomputing.com

http://www.okint.com

http://www.symmetrycomm.com

http://www.tmcnet.com

http://www3.nortelnetworks.com