INTRODUCTION

Using an ordinary phone for most people is a common daily occurrence as is listening to your favorite CD containing the digitally recorded music. It is only a small extension to these technologies in having your voice transmitted in data packets. The transmission of voice in the phone network was done originally using an analog signal but this has been replaced in much of the world by digital networks. Although many of our phones are still analog, the network that carries that voice has become digital.

In todays phone networks, the analog voice going into our analog phones is digitized as it enters the phone network. This digitization process, shown in Figure 1 below, records a sample of the loudness (voltage) of the signal at fixed intervals of time. These digital voice samples travel through the network one byte at a time.

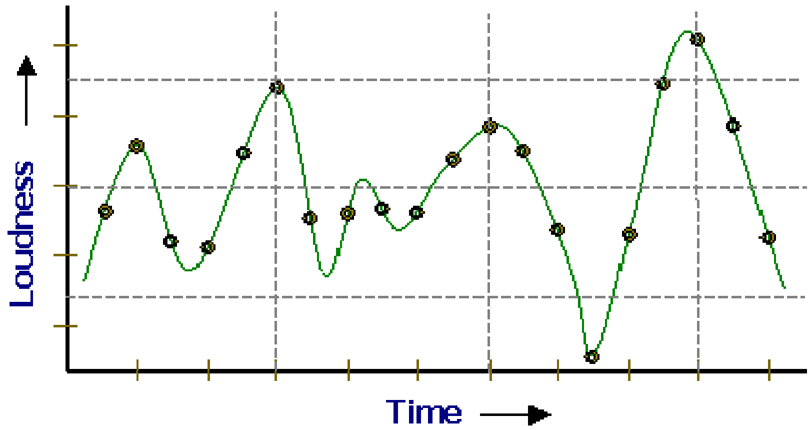


Figure 1. Digital Sampling of an analog voice signal

At the destination phone line, the byte is put into a device that takes the voltage number and produces that voltage for the destination phone. Since the output signal is the same as the input signal, we can understand what was originally spoken. The evolution of that technology is to take numbers that represent the voltage and group them together in a data packet similar to the way computers send and receive information to the Internet. Voice over IP is the technology of taking units of sampled speech data .

So at its most basic level, the concept of VoIP is straightforward. The complexity of VoIP comes in the many ways to represent the data, setting up the connection between the initiator of the call and the receiver of the call, and the types of networks that carry the call.

Using data packets to carry voice is not just done using IP packets. Although it won't be discussed, there is also voice over Frame Relay (VoFR) and Voice over ATM (VoATM) technologies. Many of the issues VoIP being discussed also apply to the other packetized voice technologies.

The increasing multimedia contents in Internet have reduced drastically the objections to putting voice on data networks. Basically, the Internet objections to putting voice on data networks. Basically, the Internet Telephony is to transmit multimedia information in discrete packets like voice or video over Internet or any other IP-based Local Area Network (LAN) or Wide Area Network (WAN). The commercial Voice Over IP (Internet Protocol) was introduced in early 1995 when VocalTec introduced its Internet telephone software. Because the technologies and the market have gradually reached their maturity, many industry leading companies have developed their products for Voice Over IP applications since 1995.

VOICE OVER INTERNET PROTOCOL

VoIP, or “Voice over Internet Protocol” refers to sending voice and fax phone calls over data networks, particularly the Internet. This technology offers cost savings by making more efficient use of the existing network.

Traditionally, voice and data were carried over separate networks optimized to suit the differing characteristics of voice and data traffic. With advances in technology, it is now possible to carry voice and data over the same networks whilst still catering for the different characteristics required by voice and data.

Voice-over-Internet-Protocol (VOIP) is an emerging technology that allows telephone calls or faxes to be transported over an IP data network. The IP network could be

- A local area network in an office
- A wide area network linking the sites of a large international organization
- A corporate intranet
- The internet
- Any combination of the above

There can be no doubt that IP is here to stay. The explosive growth of the Internet, making IP the predominate networking protocol globally, presents a huge opportunity to dispense with separate voice and data networks and use IP technology for voice traffic as well as data. As voice and data network technologies merge, massive infrastructure cost savings can be made as the need to provide separate networks for voice and data can be eliminated.

Most traditional phone networks use the Public Switched Telephone Network(PSTN), this system employs circuit-switched technology that requires a dedicated voice channel to be assigned to each particular conversation. Messages are sent in analog format over this network.

Today, phone networks are on a migration path to VoIP. A VoIP system employs a packet-switched network, where the voice signal is digitized, compressed and packetized. This compressed digital message no longer requires a voice channel. Instead, a message can be sent across the same data lines that are used for the Intranet or Internet and a dedicated channels is no longer needed. The message can now share bandwidth with other messages in the network.

Normal data traffic is carried between PC's, servers, printers, and other networked devices through a company's worldwide TCP/IP network. Each device on the network has an IP address, which is attached to every packet for routing. Voice-over-IP packets are no different.

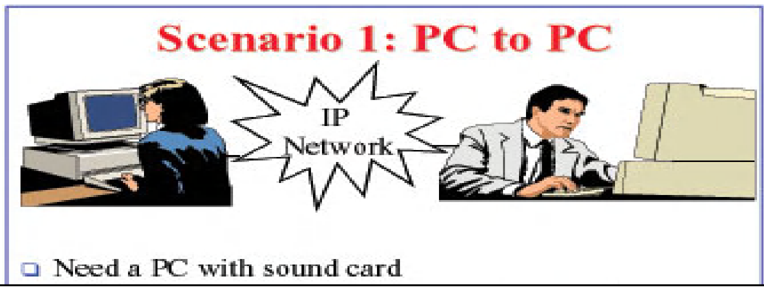
Users may use appliances such as Symbol's NetVision phone to talk to other IP phones or desktop PC-based phones located at company sites worldwide, provided that a voice-enabled network is installed at the site. Installation simply involves assigning an IP address to each wireless handset.

VOIP lets you make toll-free long distance voice and fax calls over existing IP data networks instead of the public switched telephone network (PSTN). Today business that implement their own VOIP solution can dramatically cut long distance costs between two or more locations.

SCENARIOS IN INTERNET TELEPHONY

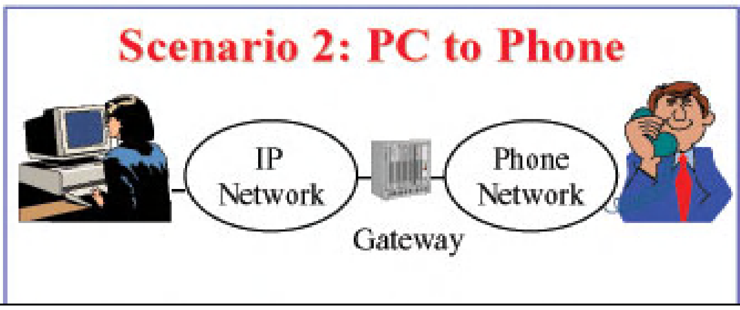
- PC to PC
- PC to Phone
- Phone to Phone

SCENARIO 1: PC TO PC



Need a PC with sound card.
IP Telephony software: Cuseeme, Internet Phone, …..
Video optional

SCENARIO 2 : PC TO PHONE

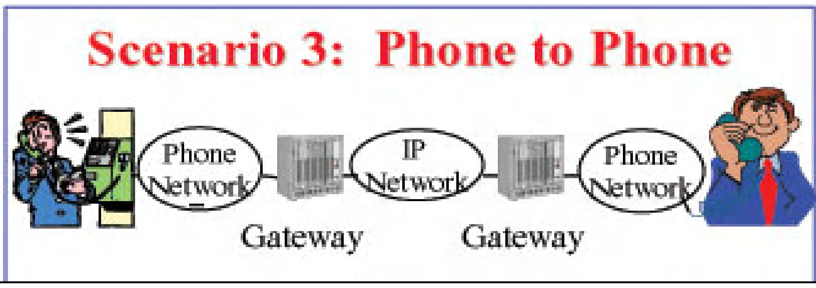


- Need a gateway that connects IP network to phone network (Router to PBX)

What are VoIP Gateways?

Many companies still have separate voice and data networks but would like to take advantage of the benefits of using Internet Telephony. A gateway therefore converts a telephone conversation into the correct format as data packets to enable it to travel across a data network as Internet Telephony. Gateways are required at both ends of a telephone conversation so that voice can be converted then reconverted back into intelligible language at the other end. VegaStream VoIP Gateways are available to cater for different sized companies according to the number of simultaneous conversations that would typical take place. For example, the Vega 50 is available with 8 ports to allow up to 8 simultaneous conversations.

SCENARIO 3: PHONE TO PHONE



- Need more gateways that connect IP network to phone networks.
- The IP network could be dedicated intra-net or the Internet.
- The phone networks could be intra-company PBXs or the carrier switches.

WORKING

In VoIP, analog voice signals is digitized using PCM. These digital voice samples are then buffered on an IP gateway. This device converts the PCM data stream into a compressed IP packet stream using DSP's (Digital Signal Processors). DSP's are responsible for converting from analog to digital as well as compression. The set of PCM samples are analysed as a discrete set of binary data. It checks the speech for all the moments of silence, which are a lot. Even when we speak, there are pauses in between that go unnoticed to the human ear, but are quiet discernible to the sampling device. The length and beginning of these pauses is noticed, while the remaining silence is removed from the data set. Similarly redundant data is also removed, making the data set more compact. Finally, an IP header is attached to this compressed data, which is then sent out on the network as discrete data packets. Once the voice packet is sent out, it finds its way to the destination just like any other data packet. It passes through various routers and switches to reach the destination gateway. Here, it gets decompressed, meaning all the periods of silence and redundant data are reinserted and is finally decoded to produce an approximation of the original sound. The compression algorithm used in the process can compress the voice signals and can even carry voice over a little as 5.3 kbps bandwidth.

APPLICATIONS OF VOICE OVER INTERNET PROTOCOL

Voice communications will certainly remain a basic form of interaction for all of us. The PSTN simply cannot be replaced, or even dramatically changed, in the short term (this may not apply to private voice networks, however). The immediate goal for VoIP service providers is to reproduce existing telephone capabilities at a significantly lower "total cost of operation" and to offer a technically competitive alternative to the PSTN. It is the combination of VoIP with point-of-service applications that shows great promise for the longer term. Some of the example are:

• **PSTN gateways**: Interconnection of the Internet to the PSTN can be accomplished using a gateway, either integrated into a PBX (the iPBX) or provided as a separate device. A PC-based telephone, for example, would have access to the public network by calling a gateway at a point close to the destination (thereby minimizing long distance charges).

• **Internet-aware telephones**: Ordinary telephones (wired or wireless) can be enhanced to serve as an Internet access device as well as providing normal telephony. Directory services, for example, could be accessed over the Internet by submitting a name and receiving a voice (or text) reply.

• **Inter-office trunking over the corporate intranet**: Replacement of tie trunks between company-owned PBXs using an Intranet link would provide economies of scale and help to consolidate network facilities.

• **Remote access from a branch (or home) office**: A small office (or a home office) could gain access to corporate voice, data, and facsimile services using the company's

Intranet (emulating a remote extension for a PBX, for example). This may be useful for home-based agents working in a call center, for example.

• **Voice calls from a mobile PC via the Internet**: Calls to the office can be achieved using a multimedia PC that is connected via the Internet. One example would be using the Internet to call from a hotel instead of using expensive hotel telephones. This could be ideal for submitting or retrieving voice messages.

• **Internet call center access**: Access to call center facilities via the Internet is emerging as a valuable adjunct to electronic commerce applications. Internet call center access would enable a customer who has questions about a product being offered over the Internet to access customer service agents online. Another VoIP application for call centers is the interconnection of multiple call centers.

H.323

The H.323 standard provides a foundation for audio, video, and data communications across IP-based networks, including the Internet. H.323 is an umbrella recommendation from the International Telecommunications Union (ITU) that sets standards for multimedia communications over Local Area Networks (LANs) that do not provide a guaranteed Quality of Service (QoS). These networks dominate today’s corporate desktops and include packet-switched TCP/IP and IPX over Ethernet, Fast Ethernet and Token Ring network technologies. Therefore, the H.323 standards are important building blocks for a broad new range of collaborative, LAN-based applications for multimedia communications. It includes parts of H.225.0 - RAS, Q.931, H.245 RTP/RTCP and audio/video codecs, such as the audio codecs (G.711, G.723.1, G.728, etc.) and video codecs (H.261, H.263) that compress and decompress media streams.

Media streams are transported on RTP/RTCP. RTP carries the actual media and RTCP carries status and control information. The signalling is transported reliably over TCP. The following protocols deal with signalling:

- RAS manages registration, admission, status.
- Q.931 manages call setup and termination.
- H.245 negotiates channel usage and capabilities.
- H.235 security and authentication.

VOICE CODING ALGORITHMS

There are several approaches to digitizing the voice samples. These approaches vary by the information that is transmitted, the complexity of the algorithm, and the assumptions of the sound being transmitted (e.g. voice, fax, music). Broadly classified, the various coding algorithms fall into two broad categories: coding of waveform and modeling of the vocal track. Pulse Code Modulation and Sub Band Coding are examples of waveform coding algorithms while Linear Predictive Coding is an example of an algorithm that models the vocal tract.

The Pulse Code Modulation (PCM) algorithm makes no assumptions about the sound that is being digitized and therefore does the best job on various types of sounds. It also produces the highest bit-rate for the data and has the shortest delay. The basics of the various PCM algorithms is that the voice is sampled at fixed time intervals (i.e. 8,000 times/second) and then a number is generated from the data based on each sample.

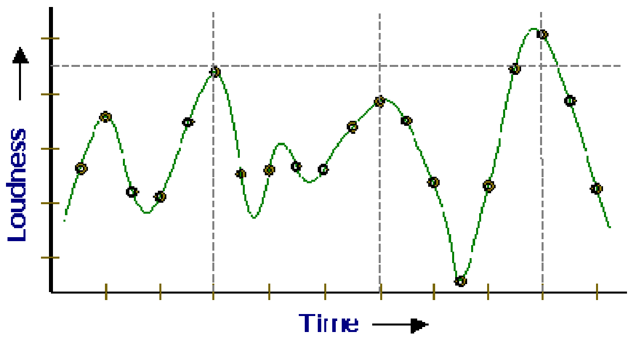


Figure 1. PCM algorithms sample the voice at fixed time intervals

ADPCM (Adaptive Differential Pulse Code Modulation), a variant of PCM, samples the voice at fixed time intervals and then calculates the change from the previous sample and sends that information. To save bandwidth, these step sizes are specially coded so that the step size at low volume is different than the step size at high volume. ADPCM provides about a 2:1 reduction in the data compared to PCM.

CODING STANDARDS

There are a number of voice coding standards and the ITU is the most active of the groups in this area. For information on the details of any of these standards, go to the ITU web site (http://www.itu.int/ITU-T).

Table 1 provides a summary of several of the major voice coding algorithms. As can be seen, there is a range of data rates available. The column labeled MOS (Mean Opinion Score), is a subjective score that listeners give to each of these algorithms. For point of reference, G.711 is what is used in the US phone system.

Table 1. Voice Coding Standards

Algorithm	**Bit Rate (Kbits/sec)**	**Complexity (Mips)**	**Delay (milliseconds)**	**MOS**
G.711 PCM	64	< 1	.25	4.4
G.723.1 MPMLQ	6.3	18	30	3.9
G.723.1 ACELP	5.3	18	30	3.6
G.726 ADPCM	32	1	.25	4.2
G.728 LD-CELP	16	30	3 - 5	4.2
G.729a CS-ACELP	8	20	10	4.2
GSM	13.2	4.5	40	3.7

Compelling VoIP Applications

The VoIP technology only becomes useful when compelling applications meet the needs of customers. Three such applications are corporations that replace PBX systems, cable operators offering telephony services using their plant, and video conferencing. These existing applications are the driving factors in allowing manufactures to make equipment, service providers to offer services, and customers to increase their productivity. Having an established market and a profitable business model allows VoIP to begin addressing the next generation of applications which then will allow the market to continue to grow.

Corporate

LAN-based telephony applications offer attractive business models to consumers today. The most important applications are the replacement of the traditional PBX, new client side applications using the PC, and the reduction in maintenance expenses for wiring changes.

LAN based PBX systems provide superior return on investment to traditional PBX systems. Although initial equipment costs are comparable, LAN PBXs typically cost much less than PBXs to install because they use the existing data infrastructure (Category 5 cabling) rather than separate voice wiring. Administration is also less burdensome because LAN and server administrators can manage the system without the need for dedicated telephony technicians.

The most compelling reason businesses consider IP telephony-type applications is for the integration of applications with voice. Over the years, a significant amount of work has gone into computer telephony integration (CTI) in traditional PBXs. These systems began to offer application-programming interfaces such as Telephony API (TAPI), Telephony Services API (TSAPI), and Java Telephony API (JTAPI). This work has resulted in advanced call center functions, including screen pops for agents and active call routing between call centers.

Client (end station) products offer LAN telephony services through the use of a software client on the user's PC, while others offer telephone instruments that plug into the

LAN. When there is a need to relocate the equipment, it needs only to be unplugged from one data port and plugged in at the new destination. This process avoids the wiring changes typically done for convention phones and thus VoIP clients reduce the cost of ownership. Over time, these savings add up to the point that a LAN-based telephony system can offer considerable savings over traditional PBXs.

PacketCable

PacketCable™ is the telephony architecture for using VoIP over the Cable TV system. With buried Cable TV plant passing hundreds of millions of homes worldwide, it is logical to assume that cable operators desire to offer new services that make use of their installed system.

The PacketCable standards (1.0, 1.1 and 1.2) were developed by CableLabs®, the research group of the cable operators and makes use of the same equipment used for the cable modem services. The cable modem architecture, known as DOCSIS (Data over Cable Service Interface Specification), is able to meet the requirements of transporting VoIP packets. DOCSIS version 1.1 provides Quality of Service (QoS), security features, and the prioritization of packet traffic that is necessary for voice communication.

The PacketCable specification also incorporates the Network-based Call Signaling (NCS) protocol for signaling voice calls over cable networks. NCS leverages the existing Media Gateway Control Protocol (MGCP) and the protocol is sometimes referred to as MGCP NCS. NCS uses network-based call agents to negotiate cable-based IP telephony calls.

Traditional telephones draw all the power they need from the phone lines. Part of the reason that the public phone system has evolved to such a reliable state, is that it is essentially immune from the effects of power outages. Electrical utilities in most areas do not offer this degree of unfailing reliability and this reality is an important issue for the cable television plant.

Normally, head-end and customer-premises cable equipment rely solely on the local electric company for their power and this puts users at risk of losing phone service should a power outage occur. There is not universal agreement among the cable operators about offering "lifeline" telephony service.

Some Time Warner systems focus their telephony service towards the "second" phone line in the home. They suggest that the customer keep the existing primary line and then add the Time Warner telephony service for a business line, fax line, or children's phone. AT&T has taken a different approach by enhancing many of their networks with alternate power sources thereby allowing lifeline telephony services to their customers.

Video Conferencing

Video conferencing includes the use of packetized voice and is an important application for the home and business. There are several video conferencing standards and one of the significant standards is known as H.32x where x can be 0, 1, 2, 3, or 4 representing video conferencing on different types of communications links. The H.32x model includes establishing connections between the source and destination devices and this method connection establishment has been adapted for use in VoIP gateways.

Table 1. The ITU-T H.32x Video Conferencing Standards

	H.320	**H.321**	**H.322**	**H.323**	**H.324**
Approval Date	1990	1995	1995	1996	1996
Network	Narrowband switched digital ISDN	Broadband ISDN ATM LAN	QoS packet switched networks	Non-QoS networks (Ethernet)	The analog phone system
Audio	G.711 G.722 G.728	G.711 G.722 G.728	G.711 G.722 G.728	G.711 G.722 G.728 G.723 G.729	G.723
Video	H.261	H.261/H.263	H.261/H.263	H.261/H.263	H.261/H.263

Control	H.230/H.242	H.242	H.230/H.242	H.245	H.245
Multiplexing	H.221	H.221	H.221	H.225	H.213
Comm. Interface	I.400	AAL I.363 AJM I.361 PHY I.400	I.400& TCP/IP	TCP/IP	V.34 Modem

A summary of the audio coding recommendations can be found in the section Voice Coding Algorithms.

The H.323 recommendation supports the largest number of audio coding standards. In fact, the audio codecs of the other recommendations are a subset of the audio codecs outlined in the H.323 recommendation. In addition, the H.323 is the recommendation that is made for a non-guaranteed bandwidth, packet switched networks such as the Internet.

H.323 Video Conferencing

The H.323 recommendation covers the technical requirements for audio and video communications services in LANs that do not provide a guaranteed Quality of Service (QoS). The scope of H.323 does not include the LAN itself or the transport layer that may be used to connect various LANs. Only elements needed for interaction with the Switched Circuit Network (SCN) are within the scope of H.323.

H.323 defines four major components for a network-based communications system: Terminals, Gateways, Gatekeepers, and Multipoint Control Units (MCUs). The terminal must support voice transmission with data and video transfer being an option.

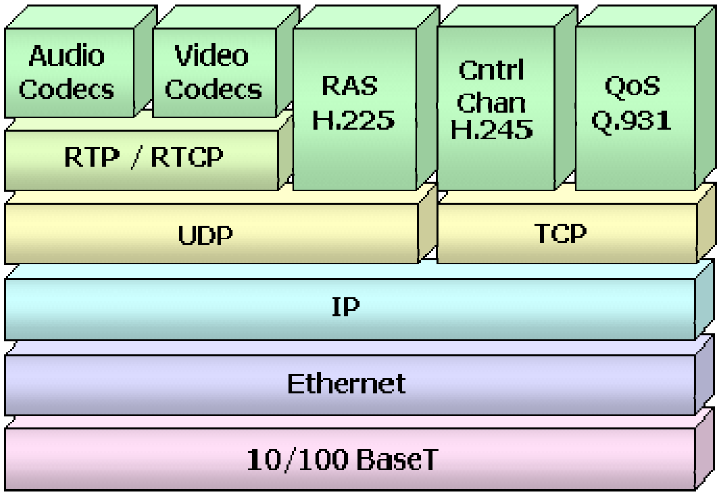


Figure 1. The H.323 Protocol Stack

H.323 uses the following standards as part of the video conferencing protocol stack (Figure 1):

- **Q.931 QoS -** Quality of Service. When the connection between devices is being established, the end stations and each data link in the route negotiate to determine what bandwidth is available, how much delay the application can tolerate, and how much jitter there will be in the packet arrival. Once the links agree to the QoS message, they must guarantee those parameters for the duration of the video conferencing session.
- **H.245 -** Control Channel Protocol. Provides capability negotiation between the two end-points such as voice compression algorithm to use, conferencing requests, etc. The H.245 channels transport must be reliable (e.g. TCP, SPX)
- **H.225 RAS -** Registration, Admission, and Status (RAS) Protocol. Used to convey the registration, admissions, bandwidth change and status messages between IP Telephone devices and servers called Gatekeepers that provide address translation and access control to devices.
- **RTCP** - Real-time Transport Control Protocol (RTCP). Provides statistics information for monitoring the quality of service of the voice call.

Control messages (Q.931 signaling, H.245 capability exchange and the RAS protocol) are carried over the reliable TCP layer. This ensures that important messages get retransmitted if necessary so they can make it to the other side. Media traffic is transported over the unreliable UDP layer and includes two protocols: RTP (Real-Time Protocol) that carries the actual media and RTCP (Real-Time Control

Protocol) that includes periodic status and control messages. Audio and Video information is carried over UDP because it need not be retransmitted because if a sound packet is lost and then transmitted, it would most probably arrive too late to be used for the real-time conferencing.

GATEWAY PROTOCOLS

H.323

The most widely embraced standard is the ITU-T’s H.323 umbrella standard. Originally designed as an end to end communications standard for video conferencing over packet networks. H.323 was adapted for VoIP applications. The result was a standard that defines far more functionality than is necessary for most VoIP environments. Its complexity is hard to implement efficiently and causes problems in interoperably since there are various ways to interpret the standard.

MGCP (MEDIA GATEWAY CONTROL PROTOCOL)

MGCP is a protocol that addresses control of media gateways, but it does not, as H.323 does, specify a complete end-to-end communication. MGCP uses simple endpoints called Media gateways (MGs). An intelligent media gateway controller (MGC) or call agent (CA) provides services. The endpoints provides user interactions and interfaces, while the MGC provides centralized call intelligence.

A master/slave relationship is presented at all times between the MGC and the MGs. Typically MGCP message are sent over IP/UDP between the MG and MGC. The media connection itself is usually over IP/RTP. For security, MGCP uses IP sec to protect the signaling information.

SIP (SESSION INITIATION PROTOCOL)

SIP is an application layer signaling protocol that specifies call control for multiparty sessions, IP phone calls or multimedia distribution. Unlike H.323, which is based on binary encoding, SIP is a text based protocol that is much easier to implement. Much like H.323, SIP is a peer-to-peer architecture (vs, master/slave for MGCP).

SIP depends on relatively intelligent endpoints, which require little or no interaction with servers. Each endpoints manages its own signaling both to the user and to other endpoints. SIP is more scalable than H.323 because it is inherently a distributed and stateless call model.

Perhaps the key advantage of SIP is that it is an Internet-model protocol from inception. It uses simple ASCII messaging based on HTTP/1.1. This means that SIP messaging is easy to decode and troubleshoot.

SIP ARCHITECTURE

- SIP teriminal: Supports real-time, 2 way communication with another SIP entity. Supports both signaling and media, similar to H.323 terminal.
- Proxy: Contacts one or more clients or next-hop servers and passes the call request further.
- Redirect Server: Accepts SIP requests, maps the address into zero or more new addresses and returns those addresses to the client. Does not initiate SIP requests or accept calls.
- Location Server: Provides information about a callers possible locations to redirect and proxy servers. May be co-located with a SIP server.

THE IMPORTANCE OF VOICE OVER IP

Of the key emerging technologies for data, voice, and video integration, voice over IP (Internet Protocol) is arguably very important. The most quality of service (QoS) sensitive of all traffic, voice is the true test of the engineering and quality of a network. Demand for Voice over IP is leading the movement for QoS in IP environments, and will ultimately lead to use of the Internet for fax, voice telephony, and video telephony services. Voice over IP will ultimately be a key component of the migration of telephony to the LAN infrastructure.

Significant advances in technology have been made over the past few years that enable the transmission of voice traffic over traditional public networks such as Frame Relay (Voice over Frame Relay) as well as Voice over the Internet through the efforts of the Voice over IP Forum and the Internet Engineering Task Force (IETF). Additionally, the support of Asynchronous Transfer Mode (ATM) for different traffic types and the ATM Forum's recent completion of the Voice and Telephony over ATM specification will quicken the availability of industry-standard solutions.

MGCP COMMANDS

•Endpoint Configuration (EPCF): Specify coding
•Notification Request (RQNT): Watch for event
•Notify (NTFY): Used by gateway to inform Call agent
•Create Connection (CRCX)
•Modify Connection (MDCX)
•Delete Connection (DLCX)
•Audit Endpoint (AUEP): Give me status
•Audit Connection (AUCX)
•Restart in Progress (RSIP): Used by gateway to indicate initialization/shutdown of endpoints/gateway

BENEFITS OF VOIP

Widespread deployment of a new technology seldom occurs without a clear and sustainable justification, and this is also the case with VoIP. Demonstrable benefits to end users are also needed if VoIP products (and services) are to be a long-term success. Generally, the benefits of technology can be divided into the following four categories:

• **Cost Reduction**: Although reducing long distance telephone costs is always a popular topic and would provide a good reason for introducing VoIP, the actual savings over the long term are still a subject of debate in the industry. Flat rate pricing is available with the

Internet and can result in considerable savings for both voice and facsimile. It has been estimated that up to 70% of all calls to Asia are to send faxes, most of which could be replaced by FoIP.

• **Simplification**: An integrated infrastructure that supports all forms of communication allows more standardization and reduces the total equipment complement. This combined infrastructure can support dynamic bandwidth optimization and a fault tolerant design.

• **Consolidation**: Since people are among the most significant cost elements in a network, any opportunity to combine operations, to eliminate points of failure, and to consolidate accounting systems would be beneficial. In the enterprise, SNMP-based management can be provided for both voice and data services using VoIP.

• **Advanced Applications**: Even though basic telephony and facsimile are the initial applications for VoIP, the longer term benefits are expected to be derived from multimedia and multiservice applications. For example, Internet commerce solutions can combine WWW access to information with a voice call button that allows immediate access to a call center agent from the PC.

ADVANTAGES OF IP TELEPHONY

Ubiquity

The IP standard is by far the world's most popular network protocol. It is developing fast and is accepted by every major vendor. Users can benefit from end-to-end connectivity to every data-networking device available, a tremendous amount of research from firms focused on IP, and a unique global addressing scheme which allows an IP device to address the entire network, regardless of size or location.

With more than 80 per cent of personal computers in business networks on a local area network (LAN), IP is an obvious transport of choice. It also seems to make sense to carry all forms of communication (data, voice, and video) over a common, ubiquitous medium.

Value Added Applications

Once an IP infrastructure is in place there is a seemingly unending number of add value applications that are available or being designed that are either far simpler than those enabled by regular telephony systems, or totally new applications like user registration that allows phone users to be identifiable and fully functional no matter which handset they're using.

Cost

For most large businesses, overhauling a PBX-based telecom infrastructure is a long-term investment decision. However, with telephony requirements becoming more demanding and new technology creating business advantage in shorter timeframes, IP telephony has become a more and more realistic alternative. And while cost projections will inevitably vary enormously from one company to another, most equipment vendors and IP telephony integration specialists would claim return on investment within one or two years.

Single Infrastructure

Putting the voice and data traffic on one set of wires instead of two also seems to make good commercial sense when compared to the cost of developing and supporting two separate infrastructures.

Overhead

A large part of that cost is a human one – and again there is a clear case for creating one pool of skills rather than two, so that voice calls essentially become just another application running on the network. In reality the network will only see information packets, some of this information originates and terminates within a data edge device whilst other packets of information on the network will originate and terminate in voice edge devices.

Call centre

Several advantages arise from a Call Centre environment based on an IP infrastructure. Web based forms, text chat sessions, email, and web collaboration are all examples of modes of communication, which are based on the IP standard. Voice is transformed to the IP standard in IP call centres. An IP infrastructure therefore facilitates integration of both web-based media and voice into the call centre.

IP also puts companies in a better position to take advantage of future developments in terms of new channels because once an IP 'skeleton' is in place, functionality is just a question of what software is developed.

MAIN JUSTIFICATIONS FOR DEVELOPMENT OF VOIP

- Cost Reduction.
- Simplification.
- Consolidation.
- Advanced Applications.

FUTURE TRENDS

Price is the key driver of the VoIP market today. End-user features such as multimedia conferencing, multicast, call centers, IP call waiting, and message unification are the benefits that will drive the VoIP market well into the future. The growing competition between ISPs is causing declining margins. ISPs are seeking value-added services to increase revenues per subscriber. Becoming an ITSP is the solution.The demand for convergent networks is evolving into a requirement for new network/telephone orders and upgrades.