VOIP

SARAT ETHAKOTA

999 10 9519

Background

The past

In the past ,20-30 years ago Internet didn't exist and Interactive communications were only made by telephone at PSTN line cost. Data exchange was expansive (for a long distance) and no one had been thinking to video interactions. There was only television that is not interactive, as known.

Yesterday

Few years ago we saw appearing some interesting things starts to happen like: PCs to large masses, new technologies to communicate like cellular phones and finally the great net: Internet has evolved. People begun to communicate with new services like email, chat, etc. and business reformed with the web allowing people buy with a "click".

Today

Today we can see a real revolution in communication world. Everybody begins to use PCs and Internet for job and free time to communicate each other, to exchange data (like images, sounds, documents) and, sometimes, to talk each other using applications like Netmeeting or Internet Phone. Particularly starts to diffusing a common idea that could be the future and that can allow real-time vocal communication: VoIP.

The future

We cannot know what is the future, but we can try to imagine it with many computers, Internet almost everywhere at high speed and people talking (audio and video) in a real time fashion.

What is VoIP?

VoIP stands for 'V'oice 'o'ver 'I'nternet 'P'rotocol. As the term says VoIP tries to let go voice (mainly human) through IP packets and, in definitive through Internet. VoIP is the ability to make telephone calls and send faxes over IP-based data networks with a suitable quality of service (QoS) and superior cost benefit . 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 . Voice over IP (VoIP) allows the human voice and fax information to travel over a packet data network concurrently with traditional data packets.

A major advantage of VoIP and Internet telephony is that it avoids the tolls charged by ordinary telephone service.

VoIP, now used somewhat generally, derives from the VoIP Forum, an effort by major equipment providers, including Cisco, VocalTec, 3Com, and Netspeak to promote the use of ITU-T H.323, the standard for sending voice (audio) and video using IP on the public Internet and within an intranet. The Forum also promotes the user of directory service standards so that users can locate other users and the use of touch-tone signals for automatic call distribution and voice mail. In addition to IP, 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). VoIP can use accelerating hardware to achieve this purpose and can also be used in a PC environment. VoIP provides rich benefits for all levels of users, from networking equipment manufacturers who are providing next-generation equipment to service providers, who can now market new services to business and home users.

At the very minimum, these benefits include more cost-effective traditional voice and fax service. However, users will also benefit from revolutionary, new VoIP-based services. VoIP will fundamentally change the way communication occurs.

Overview:

The possibility of voice communications traveling over the Internet, rather than the PSTN, first became a reality in February 1995 when Vocaltec, Inc. introduced its Internet Phone software. Designed to run on a 486/33-MHz (or higher) personal computer equipped with a sound card, speakers, microphone, and modem, the software compressed the voice signal and translated it into IP packets for transmission over the Internet. This PC-to-PC Internet telephony worked, however, only if both parties were using Internet Phone software.

In the relatively short period of time since then, Internet telephony has advanced rapidly. Many software developers now offer PC telephony software but, more importantly, gateway servers are emerging to act as an interface between the Internet and the PSTN. Equipped with voice-processing cards, these gateway servers enable users to communicate via standard telephones over great distances without going over the "Long Distance" telephone network.

A call goes over the local PSTN to the nearest gateway server, which digitizes the analog voice signal, compresses it into IP packets, and moves it onto the Internet for transport to a gateway server at the receiving end. This server converts the digital IP signal back to analog and completes the call locally. With its support for computer-to-telephone calls, telephone-to-computer calls and telephone-to-telephone calls, VoIP represents a significant step toward the integration of voice and data networks.

Originally regarded as a novelty, Internet telephony is attracting more and more users because it offers tremendous cost savings relative to the PSTN. Users can bypass long-distance carriers and their per-minute usage rates and run their voice traffic over the Internet for a flat monthly Internet-access fee. VoIP provides a competitive threat to the providers of traditional telephone services that, at the very least, will stimulate improvements in cost and function throughout the industry.

VoIP could be applied to almost any voice communications requirement, ranging from a simple inter-office intercom to complex multi-point teleconferencing/shared screen environments.

What is the advantages using VoIP rather PSTN?

When you are using PSTN line, you typically pay for time used to a PSTN line manager company: more time you stay at phone and more you'll pay. In addition you couldn't talk with other that one person at a time.
In opposite with VoIP mechanism you can talk all the time with every person you want (the needed is that other person is also connected to Internet at the same time), as far as you want (money independent) and, in addition, you can talk with many people at the same time.

If you're still not persuaded you can consider that, at the same time, you can exchange data with people are you talking with, sending images, graphs and videos.

WORKING OF VOIP

To setup a VoIP communication we need:

1. First the ADC to convert analog voice to digital signals (bits)
2. Now the bits have to be compressed in a good format for transmission: there is a number of protocols we'll see after.
3. Here we have to insert our voice packets in data packets using a real-time protocol (typically RTP over UDP over IP)
4. We need a signaling protocol to call users: ITU-T H323 does that.
5. At RX we have to disassemble packets, extract datas, then convert them to analog voice signals and send them to sound card (or phone)
6. All that must be done in a real time fashion cause we cannot waiting for too long for a vocal answer! (see QoS section)

BENEFITS OF THE TECHNOLOGY

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. Reducing long distance telephone costs is always a popular topic and provides a good reason for introducing VoIP. Today flat rate long distance pricing is available with the Internet and can result in considerable savings for both voice and facsimile (at least currently). The sharing of equipment and operations costs across both data and voice users can also improve network efficiency since excess bandwidth on one network can be used by the other, thereby creating economies of scale for voice (especially given the rapid growth in data traffic).
- 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. The differences between the traffic patterns of voice and data offer further opportunities for significant efficiency improvements.
- Consolidation. Since people are 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. Universal use of the IP protocols for all applications holds out the promise of both reduced complexity and more flexibility. Related facilities such as directory services and security services may be more easily shared.
- 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. Needless to say, voice is an integral part of conferencing systems that may also include shared screens, whiteboarding, etc. Combining voice and data features into new applications will provide the greatest returns over the longer term.

QOS ISSUES

The advantages of reduced cost and bandwidth savings of carrying voice-over-packet networks are associated with some quality-of-service (QoS) issues unique to packet networks.

Delay

Delay causes two problems: echo and talker overlap. Echo is caused by the signal reflections of the speaker's voice from the far-end telephone equipment back into the speaker's ear. Echo becomes a significant problem when the round-trip delay becomes greater than 50 milliseconds. As echo is perceived as a significant quality problem, voice-over-packet systems must address the need for echo control and implement some means of echo cancellation.
Talker overlap (or the problem of one talker stepping on the other talker's speech) becomes significant if the one-way delay becomes greater than 250 milliseconds. The end-to-end delay budget is therefore the major constraint and driving requirement for reducing delay through a packet network.

Accumulation Delay (Sometimes Called Algorithmic Delay)

This delay is caused by the need to collect a frame of voice samples to be processed by the voice coder. It is related to the type of voice coder used and varies from a single sample time (.125 microseconds) to many milliseconds. A representative list of standard voice coders and their frame times follows:

- G.726 adaptive differential pulse-code modulation (ADPCM) (16, 24, 32, 40 kbps)—0.125 microseconds
- G.728 LD–code excited linear prediction (CELP)(16 kbps)—2.5 milliseconds
- G.729 CS–ACELP (8 kbps)—10 milliseconds
- G.723.1 Multirate Coder (5.3, 6.3 kbps)—30 milliseconds

Processing Delay

This delay is caused by the actual process of encoding and collecting the encoded samples into a packet for transmission over the packet network. The encoding delay is a function of both the processor execution time and the type of algorithm used. Often, multiple voice-coder frames will be collected in a single packet to reduce the packet network overhead. For example, three frames of G.729 code words, equaling 30 milliseconds of speech, may be collected and packed into a single packet.

Network Delay

This delay is caused by the physical medium and protocols used to transmit the voice data and by the buffers used to remove packet jitter on the receive side. Network delay is a function of the capacity of the links in the network and the processing that occurs as the packets transit the network. The jitter buffers add delay, which is used to remove the packet-delay variation to which each packet is subjected as it transits the packet network. This delay can be a significant part of the overall delay, as packet-delay variations can be as high as 70 to 100 milliseconds in some frame-relay and IP networks.

Jitter

The delay problem is compounded by the need to remove jitter, a variable interpacket timing caused by the network a packet traverses. Removing jitter requires collecting packets and holding them long enough to allow the slowest packets to arrive in time to be played in the correct sequence. This causes additional delay.

The two conflicting goals of minimizing delay and removing jitter have engendered various schemes to adapt the jitter buffer size to match the time-varying requirements of network jitter removal. This adaptation has the explicit goal of minimizing the size and delay of the jitter buffer, while at the same time preventing buffer underflow caused by jitter.

Two approaches to adapting the jitter buffer size are detailed below. The approach selected will depend on the type of network the packets are traversing.

The first approach is to measure the variation of packet level in the jitter buffer over a period of time and incrementally adapt the buffer size to match the calculated jitter. This approach works best with networks that provide a consistent jitter performance over time, such as ATM networks.

The second approach is to count the number of packets that arrive late and create a ratio of these packets to the number of packets that are successfully processed. This ratio is then used to adjust the jitter buffer to target a predetermined, allowable late-packet ratio. This approach works best with the networks with highly variable packet-interarrival intervals—such as IP networks.

In addition to the techniques described, the network must be configured and managed to provide minimal delay and jitter, enabling a consistent QoS

VOIP STANDARDS

VoIP technology is rapidly evolving, but still there are ongoing debates about a de facto standard that will bridge traditional circuit-switched networks and packet-switched networks. Four protocol standards govern VoIP technology: H.323, SIP, MGCP and Megaco/H.248.

H.323

H.323 is part of a broader family of standards developed by ITU describing how audio, video and data communications occur between terminals, network equipment and services on IP networks that do not provide guaranteed QoS--most notably, the Internet. H.323 defines four major components for networked communications:

- Terminals, which are LAN client end points that provide two-way communications.
- Gateways, which provide for real-time, two-way communications between H.323 terminals on an IP network and other ITU terminals on a switched-based network or to another H.323 gateway, functioning as a translator.
- Gatekeepers, which act as central points for calls in their zones and provide services to registered end points.
- MCUs (multipoint control units), which are end points that provide the capability for three or more terminals and gateways to participate in a multipoint conference.

Because it was initially designed to support video packets, H.323 has considerable overhead, which is a disadvantage for IP telephony applications. As an early VoIP protocol, however, H.323 has been promoted as the standard for interoperability by Internet phone and VoIP vendors.

Session Initiation Protocol

SIP, an application-layer control protocol proposed by the IETF (RFC 2543), overcomes H.323's shortcomings. It can establish, modify and terminate multimedia sessions or calls, such as conferences, distance learning, Internet telephony and similar applications. SIP enables VoIP gateways, client end points, PBXes, and other communications systems and devices to communicate with each other. It provides a lightweight protocol that enables scalable call control and a platform for applications.

SIP handles user location, user capabilities, user availability, call setup and handling. It supports name mapping and redirection services that facilitate the implementation of ISDN and Intelligent Network telephony services that allow for mobility. It can also initiate multiparty calls using an MCU or fully meshed interconnection instead of multicast. SIP works in conjunction with RSVP (Resource Reservation Protocol), RTP (Real-Time Transport Protocol), RTSP (Real-Time Streaming Protocol), SAP (Session Announcement Protocol) and SDP (Session Description Protocol). Unlike H.323, SIP has little overhead, as this protocol reuses most of the header fields, encoding rules, error codes and authentication mechanisms of HTTP.

Media Gateway Control Protocol

MGCP is a standard proposed by the IETF (RFC 2705) for the conversion of audio signals carried on PSTN to data packets that travel over the Internet. MGCP enables MGCs (media gateway controllers) and media gateways to communicate. It combines IPDC (IP Device Control) and SGCP (Simple Gateway Control Protocol). It assumes an architecture in which the call-control "intelligence" is outside the gateways and handled by external call-control elements, or call agents. It is a master/slave protocol, wherein gateways execute commands sent to them by the call agent. MGCP is losing momentum as a true VoIP standard with the emergence of the Megaco/H.248 standard.

Megaco/H.248

This new protocol, jointly proposed by the ITU-T Study Group 16 and Megaco work group of IETF, lets an MGC control media gateways. As a successor to MGCP, it adds peer-to-peer interoperability capabilities and provides a means of control appropriate for IP telephone devices operating in a master/slave relationship, similar to MGCP. Megaco/H.248 decomposes the H.323 gateway function into subcomponents and specifies the protocols used by each component for communication. Megaco/H.248 will allow low-cost gateway devices to interface with signaling systems found in circuit-switched networks. H.248 leverages on the existing PSTN, making its implementation cheaper and faster for network operators

Although the use of voice over packet networks is relatively limited at present, there is considerable user interest and trials are beginning. End user demand is expected to grow rapidly over the next five years. Frost & Sullivan and other research firms have estimated that the compound annual growth rate for IP-enabled telephone equipment will be 132% over the period from 1997 to 2002 (from $47.3M in 1997 to $3.16B by 2002). It is expected that VoIP will be deployed by 70% of the Fortune 1000 companies by the year 2002. Industry analysts have also estimated that the annual revenues for the IP fax gateway market will increase from less than $20M in 1996 to over $100M by the year 2002. It is clear that a market has already been established and there exists a window of opportunity for developers to bring their products to market, and for consumers to realize significant savings.

New applications such as web-enabled call centers, collaborative white boarding, remote telecommuting, and personal productivity applications such as "follow-me" services and unified message handling, are all part of the VoIP sea change.

VoIP Summary

A VoIP software architecture has been described for the interworking of legacy telephony systems and packet networks. Some of the key features enabling this application to function successfully are as follows:

- an approach that minimizes the effects of delay on voice quality
- an adaptive playout to minimize the effect of jitter
- features that address lost-packet compensation, clock synchronization, and echo cancellation
- a flexible DSP system architecture that manages multiple channels per single DSP

Carrying VoIP networks provides the most bandwidth-efficient method of integrating these divergent technologies. While the challenges to this integration are substantial, the potential savings make the investment in a quality implementation compelling.

REFERENCES

http:// www.iec.org

http:// www.protocols.com

http://www.fatamorgana.com