Internet Telephony: Services, Technical Challenges, and Products

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ABSTRACT

The rapid proliferation of the Internet in the last few years has given rise to a strong interest in carrying telephony over the Internet. Because the Internet supports data communications, a range of other services can be bundled together with Internet telephony. The Internet, however, was designed for non-real-time data communications, and hence it poses several technical challenges that must be overcome before the Internet can be successfully used for carrying telephone services. This article discusses new services we can expect from Internet telephony, the technical challenges and solutions, and the emerging products that promise to support Internet telephony.

INTRODUCTION

Circuit switching and packet switching are the two main technologies for computer and telecommunications networks. The current telephone system is based on circuit switching, which offers a guaranteed quality of service (QoS) to customers. A circuit has to be set up between two endpoints before the start of communication. On the contrary, data communications have so far been considered non-real-time, and hence have been carried over packet-switched networks such as the Internet, which is based on the TCP/IP protocol suite.

The proliferation of the Internet in the last few years has given rise to a strong interest in the possibility of carrying real-time voice traffic over the Internet. Since the Internet was not initially designed for real-time communications, carrying voice over the Internet presents a number of challenges and technical issues that need to be solved prior to successful deployment of telephony over the Internet. Some of these technical challenges include the lack of guarantee in terms of bandwidth, packet loss, delay, and jitter which affect the quality of voice over the Internet.

However, telephony over the Internet brings in a number of services and their integration which are not possible using traditional circuit-switched telephone networks. It allows the integration of voice, fax, and data over the same network, transport of intracompany voice communications between remote sites of an enterprise, and others. Services such as Web-based call centers, real-time billing, remote teleworking, and enhanced teleconferencing using shared whiteboard and shared applications are some of the services possible using IP telephony.

A number of products have been developed by commercial vendors which allow voice communications over the Internet. These products can be divided into three groups according to the end user: carrier, enterprise, and individual users. The objective of this article is to bring together a survey of the new services possible with IP telephony, the technical challenges of IP telephony, and products that facilitate IP telephony. The above topics will be described in detail.

APPLICATIONS AND SERVICES

A major difference between the conventional public switched telephone network (PSTN) and IP telephony is that IP telephony is a voice service built on top of existing data communications services. As a result, IP telephony can go beyond the services offered by simple voice communications. Data services can easily be combined with voice services to create new applications and services which are not possible with conventional telephony. A few of the emerging applications enabled by this new technology are described below, but these applications should not be construed as constituting the entire IP telephony market. Almost every month new applications are created; any of these applications may become a killer application in the next millennium.

INTEGRATION OF DATA, VOICE, AND FAX

Because IP telephony can be supported by data communication networks, multisite enterprises can consolidate their existing telephone networks with data communications networks to achieve large-scale cost savings. The cost saving comes from the fact that enterprises now need to maintain only one network to support voice, data, and fax. Since IP telephony also supports video communications, further network consolidation is possible by combining the video network (if the enterprise has a video network for conducting videoconferencing among different sites) into the same network.

SOUND GRADING

PSTN supports only one grade of sound, 4 kHz toll-quality sound. PSTN is therefore not suitable for high-fidelity stereo and surround sound. IP telephony can support higher grades of sound if there is enough bandwidth in the IP network.

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VIDEO TELEPHONY

Since IP telephony also supports video transmission, it is easier to support video telephony over IP telephony.

UNIFIED MESSAGING

Most employees have a number of communications services on which they rely to keep in touch with customers or colleagues in the course of their duties. Most people have an e-mail address, a mobile phone number, a telephone number, and a fax number at which they can be reached during working hours, and perhaps another phone and fax number pair for home use. This proliferation of contact points results in the user being increasingly out of touch as s/he can only be in one place at a time.

The messaging services provided by traditional phone companies are restricted to voice mail only; they do not allow one to access faxes or e-mail. With the use of packet-switching systems such as IP networks, a unified messaging system has become a reality. The user can get all the messages sent to one location from which s/he can access it at his/her convenience. Voice mail from the home or work telephone can be forwarded to the same location as e-mail. This feature can even be extended further to using a single telephone number for all telecommunication services.

A VIRTUAL SECOND LINE

Many home Internet users subscribe to two telephone lines; one line is used to make and receive voice calls, the other for Internet surfing. With IP telephony, home users can use the same telephone line for voice calls even when they are using it for Internet surfing. IP telephony therefore provides a virtual second line at no extra cost.

WEB-BASED CALL CENTERS

Web-based call centers allow users browsing the Internet to initiate a voice-over-IP (VoIP) call from an organization's Web site to its call center. The Internet surfer does not need to stop browsing; instead, the Internet call will just be an extension of his/her activities. The advantage of this service is twofold. First, it helps to capture a potential client's attention while still at its peak. Many people tend to lose their keenness to do something as time goes on. This is also true with potential customers; they may like a product advertised on a Web site, but might not follow on because it is inconvenient to terminate their browsing immediately in order to place an order over the phone. The second advantage arises in cases where the Web surfer may require further information, which may even be available on the same site. With Web-based call centers, the surfer can speak to the call center staff, and be directed to further information.

LOW-COST VOICE CALLS

Unlike PSTN, which uses circuit-switching technology, IP telephony uses packet-switching technology. With packet switching, no communication link is dedicated to voice calls. All calls share the network resources. Such sharing significantly brings down the cost of a phone call.

REAL-TIME BILLING

Although the core network of the PSTN has used intelligent devices for decades, this has remained transparent to the user, especially in terms of the billing information, since the user has to wait until the bill comes in the mail to know the exact amount due. This has been mainly due to the limited functionality of conventional telephone sets, which by and large still have 12 standard keys for operation. With the use of VoIP, computers are increasingly being used by the end user, allowing them to gain access to the gateway for billing information in real time. The fact that VoIP and the Internet are both IP-based may also be the reason why some service providers allow their clients to access billing information from the providers' Web sites. This enables those callers who may be calling from a conventional telephone to also know the status of their account.

REMOTE TELEWORKING

The use of VoIP also enhances the resources available to teleworkers. Teleworkers need access to the company private branch exchange (PBX) to receive and send calls just like other workers physically located at the company. They may also require access to the LAN at the office from time to time. These services can be made available to remote users through the company VoIP gateway.

ENHANCED TELECONFERENCING

Teleconferencing started well before the idea of IP telephony began. However, IP telephony has revolutionized the way teleconferencing is conducted. Conventional teleconferencing requires expensive equipment in specially prepared rooms for a reasonable picture quality. Teleconferencing over IP is more flexible, allowing users in more than two locations to hold a conference with modest equipment such as a desktop camera and a multimedia PC. Improvements in video encoding techniques have helped to improve the quality of the moving picture. Teleconferencing over IP networks has also brought a richness of services that were not present in traditional teleconferencing systems. Users can share documents through electronic whiteboarding; they can share an application installed on only one conference participant's computer, and transfer files among the conference members in real time.

TECHNICAL CHALLENGES

IP telephony faces many technical challenges such as loss, delay, and jitter. This section looks at the technical challenges and solutions for IP telephony.

PACKET LOSS

Packet loss is a common phenomenon in all packet-switching networks, including IP networks. Unlike PSTN, which is a circuit-switched network, no end-to-end physical circuits are established in IP networks. IP packets from many sources are queued for transmission over an outgoing link in a router; packets are transmitted one by one from the head of the queue.

A major difference between conventional PSTN and IP telephony is that IP telephony is a voice service built on top of existing data communication services. As a result, IP Telephony can go beyond the services offered by simple voice communications.

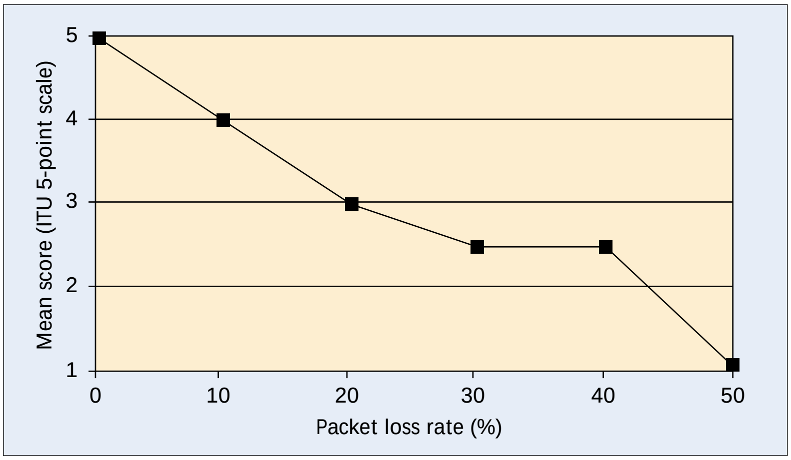

Figure 1. *Voice quality as a function of packet loss rate.*

An arriving packet is lost in the network if there is no space in the queue. As more and more people use the Internet, routers often become congested, resulting in packet loss.

Packet loss can cause severe damage to voice quality for IP telephony. Each IP packet contains 40–80 ms of speech information, matching the duration of critical units of speech called *phonemes*. When a packet is lost, a phoneme is lost in the continuous speech. While the human brain is capable of reconstructing a few lost phonemes in speech, too much packet loss makes a voice unintelligible. Figure 1 shows how the quality of voice degrades as the loss rate increases [1].

A number of techniques used to address the problem of packet loss in IP telephony are discussed below. While some of these techniques focus on reducing packet loss, others concentrate on repairing the damage caused by packet loss.

Network Upgrade — Since packet loss in the IP router is a direct result of insufficient link bandwidth and/or router packet processing speed, upgrading the IP network infrastructure, the links and the routers, is a direct engineering solution to the packet loss problem. In the last few years several promising technologies have emerged which can significantly upgrade the transmission capacity of IP backbone links and routers. High-speed transmission technologies include asynchronous transfer mode (ATM) for megabit-per-second, synchronous optical network (SONET) for gigabit-per-second, and wavelength-division multiplexing (WDM) for terabit-per-second line speeds. To complement the phenomenal increase in link bandwidth, high-speed switching-based router technologies have emerged which can process packets in the order of millions of packets per second.

Although network upgrade provides a network engineering solution to the packet loss problem, it is a very expensive and long-term solution. While network upgrade attempts to reduce packet loss, there are other techniques that focus on repairing the damage done by packet loss to voice quality. Several such techniques are described in the following sections.

Silence Substitution — As packets arrive at the destination, the contents of the packets are played back to reconstruct the original voice. When a packet is lost in the network, the content of the packet cannot be played out. Some VoIP systems, such as Internet MBone, substitute silence in place of a missing packet. While it allows the destination to continue to play the voice without any disruption, experience showed that silence substitution caused voice clipping, which deteriorated the quality of the voice significantly. This is particularly true for large packets and high loss rates. Studies on silence substitution revealed that it achieves adequate performance only for packet sizes smaller than 16 ms at loss rates up to 1 percent [2].

Noise Substitution — Substitution for lost packets with white (background) noise has been shown to perform better than silence substitution. This has been attributed to the ability of the human brain to repair the received message if there is some background noise (known as *phonemic restoration*), which is not possible if there is silence [2, 3].

Packet Repetition — Replaying the last correctly received packet in place of the lost one is another way of recovering from lost packets. The Global System for Mobile Communications (GSM) recommends that the repeated signal be damped or faded to ensure a better quality.

Packet Interpolation — Interpolation-based repairs use the characteristics of the packets in the neighborhood of the lost one to produce a replacement. This ensures that the replacement will follow the changing characteristics of the whole voice stream. Studies have shown that interpolation recovery using the waveform characteristics of the sound before and after the lost packet gives sound quality better than that achieved using silence substitution or packet repetition [4]. There are variations in interpolation methods, with one method concentrating more on the pitch of the voice signal, and another on the timescale.

Frame Interleaving — The effect of packet loss can be reduced by interleaving voice frames across different packets. The procedure involves rearrangement of the original frames to ensure that previously consecutive frames are separated at transmission and rearranged back in their original sequence at the receiver. With interleaving, the loss of a single packet will only result in multiple short gaps in different streams of the received data, which the receiver is able to tolerate, compared to one long gap consisting of consecutive frames, which would occur in a noninterleaved data stream. Frame interleaving has the disadvantage of increasing the delay. Frames that were originally consecutive are spread over a number of packets, only to be rearranged at the receiver. However, if packet interleaving can be implemented within the constraints of the delay budget, it is an attractive loss recovery technique because it does not introduce overheads in the network.

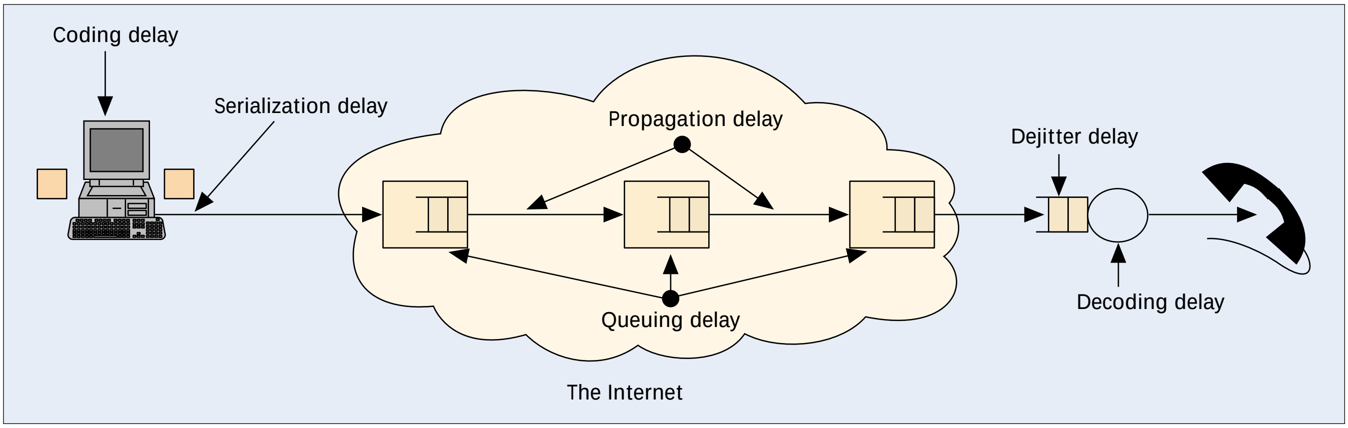

Figure 2. *Delays encountered in IP telephony.*

Forward Error Correction — In FEC, the information in a packet is redundantly transmitted in subsequent packet(s). In the event that the original packet is lost, it can be reconstructed from subsequent packets. The redundancy may either be independent of the data stream, or use the stream characteristics to enhance the repair process. Since Real-Time Transport Protocol (RTP) is the protocol used to support IP telephony over the Internet, it is appropriate to have mechanisms within RTP to carry redundant voice packets. An RTP payload format to carry redundant voice packets is discussed in [5].

PACKET DELAY

Timing is an important characteristic of voice. Two syllables of a word are uttered with an interval. This interval is as much a part of the voice as the uttered syllable. If additional delay is inserted between syllables, the rhythm of voice is lost. Too much delay can impair voice in several ways. First, long delays cause two speakers to enter a half-duplex communications mode, where one speaks, and the other listens and pauses to make sure the other is done. If the pausing is ill timed, the speakers end up "stepping" on each others' speech. Second, long delay exacerbates echo, because the reflected signal comes back to the sender after the sender finishes transmission.

The question that arises naturally is, what is the value of the delay threshold for voice? The basic guideline is that delays below 150 ms are acceptable for most applications. As delays exceed 150 ms, users start to run into echo and step on each others' speech. However, delays between 150 and 400 ms are still acceptable for long distance communications, such as between Melbourne, Australia, and New York, United States, since users are mentally prepared for long delays due to unavoidable signal propagation delay. Voice quality deteriorates significantly above 400 ms and is not acceptable in most cases.

Delay is not a big issue with current circuit-switched telephone networks, such as the integrated services digital network (ISDN) and PSTN. The primary source of delay in such networks is the signal propagation delay, which depends directly on distance. Since the signal travels almost at the speed of light, it can be kept well below 400 ms even for long distance satellite links.

Unfortunately, one of the biggest technical challenges facing IP telephony is delay. With packet-switching networks, such as IP networks, there are many factors contributing to the delay, the most significant being the queuing delay, discussed below. While some delays are fixed and known in advance, others are variable and unpredictable. Figure 2 illustrates the various types of delays that are encountered as voice travels from one end to other in IP telephony. The total delay can easily exceed the 400 ms mark.

Following is a discussion of the various sources of delay in IP telephony and the approaches that can be taken to reduce or eliminate delay where possible.

Codec Delay — The primary function of a codec is to convert analog voice to digital data and vice versa. Codecs also perform voice compression to reduce the bandwidth requirement of voice transmission over digital networks. Analog–digital conversion and voice compression introduce delays in the codec. Higher compression is achieved at the price of longer delays. Two factors that contribute to the total encoding delay are *frame processing delay* and *lookahead delay*. Frame processing delay is the delay to process a single voice frame, the amount of voice to be packed in one packet. Lookahead delay is the delay to process part of the next frame to exploit any correlation in successive voice frames. Decoding delay at the receiver is typically half the encoding delay at the sender [6]. Table 1 lists encoding and decoding delays for several voice coding standards (mainly to facilitate IP telephony) recently standardized by the International Telecommunication Union (ITU).

Serialization Delay — Serialization delay is the time it takes to place a packet on the transmission line, and is determined by the speed of the line. With higher line speeds, serialization delay can be greatly reduced. For example, it takes 125 μs to place 1 byte of information on a 64 kb/s line, while it takes only 0.05 μs to place the same amount of information on an OC-3 line, which runs at 155 Mb/s. Note that serialization delay also depends on the frame size used by a codec. Longer frames result in higher delay in transmitting the packet.

Coding standard	Compression algorithm	Bit rate (kb/s)	Frame processing delay (ms)	Lookahead delay (ms)	Total encoding delay (ms)	Typical decoding delay (ms)
G.711	PCM	64	0	0	0	0
G.729	CS-ACELP	8	10	5	15	7.5
G.723.1	ACELP	5.3/6.4	30	7.5	37.5	18.75

Table 1. *Encoding and decoding delays.*

Queuing Delay — Queuing delay occurs at the various switching and transmission points of the network, such as routers and gateways, where voice packets wait behind other packets waiting to be transmitted over the same outgoing link. Since the number of packets waiting in the queue depends on the statistical nature of the arrival process, queuing delay on the Internet varies significantly from packet to packet.

Queuing delays can be reduced in a number of ways. Faster links can be used, but this is only applicable in a network where users have control of the infrastructure, such as a corporate IP network. The Internet Engineering Task Force (IETF) is working on mechanisms, such as differentiated services (DiffServ) [7] and Resource Reservation Protocol (RSVP) [8], to prioritize voice packets over data packets to minimize queuing delay for voice and other delay-sensitive applications.

Propagation Delay — The time required by signals to travel from one point to another is fixed, and is determined by the speed of light. This delay becomes significant where long distances are involved. Long fixed delay is apparent in calls that are routed over a satellite link, especially a geostationary earth orbit (GEO) satellite. Propagation delay for long distance satellite calls is also a problem for traditional telephony.

One of the ways to reduce the long delays in connections involving GEO satellites is to use a cluster of low earth orbit (LEO) satellites. In the case of LEO satellites, a connection from the earth is handed over from satellite to satellite as the earth station goes out of the footprint of one satellite and comes under the footprint of another satellite. This, however, gives rise to variable delay paths since LEO satellite are moving with respect to the ground station, and also results in buffering at the nodes during connection handover.

Other Sources of Delay — Some delay sources are specific to certain implementations of VoIP systems. In dialup networks, there are delays caused by modems. Such delays can be avoided by using digital lines. Packet voice systems using multimedia PCs also incur delays due to operating system inefficiencies and sound card delays. Such problems can be addressed by using gateway cards that use fast, specialized digital signal processors (DSPs).

Network Jitter

Variance in the interframe arrival times at the receiver is called *jitter* and is potentially more disruptive for IP telephony than the delays discussed above. Jitter occurs due to the variability of queuing delays in the network and propagation delays in the case of connections utilizing LEO satellites. IP packets belonging to the same stream may even take different paths in the Internet and experience different delays. Network jitter can be significant even for low average network delays. If an IP packet is inordinately delayed, it will not arrive in time at the receiver and will be considered lost. If this happens too often, the quality of voice will be affected significantly.

To allow for variable packet arrival times and still achieve steady stream of packets, the receiver holds the first packet in a jitter buffer for a while before playing it out. The amount of this hold time is the measure of the size of the jitter buffer. For example, a 50 ms hold time means 50 ms jitter buffer.

The jitter buffer hold time adds to the overall delay. Therefore, for high jitter, the overall perceived delay is high even if the average delay is low. For example, the overall delay is only 55 ms for a moderate average delay of 50 ms with a 5 ms jitter buffer. In contrast, if the network has a low average delay of 15 ms, but occasionally the packet is delayed by 100 ms, the delay buffer would have to be 100 ms; the overall delay in this case is 115 ms!

Selection of jitter buffer size is crucial to IP telephony systems. An optimum buffer size has to be found which balances removing jitter and limiting delay to tolerable levels. If the buffer is set too low, some packets may be lost; if set too high, higher delays result. The jitter buffer size may be determined by using the ratio of late packets to those that arrive in time. Ideally, the jitter buffer size should be modified dynamically to suit varying network conditions. Common buffer sizes range from 50 to 100 ms. Cisco, Hypercom, and Netrix, among others, offer intelligent buffers that adjust automatically according to network variability.

Integration of IP and PSTN

There are several architectures for end users. The first one is a PC–PC architecture where users are equipped with multimedia-capable computers directly connected to the Internet using network interface cards in the case of a LAN or via modem/cable modem when the connection is through an Internet service provider. All the sampling, compression, coding, and decoding happens at the computers. This places an enormous load on the CPU unless a hardware card is used to carry out the above functions. Calls between users are established using IP addresses. In this architecture, IP and PSTN continue to operate independently.

PC–phone is an alternative IP telephony architecture which allows a PC user to establish a call with a conventional PSTN phone user. This architecture leads to the issue of integration of IP networks and PSTN. An extension of PC–phone architecture is phone–Internet–phone architecture which uses the Internet in the background to reduce telephone costs for conventional phone users. In a phone–Internet–phone architecture, conventional telephone sets are used. A user intending to call another user calls a particular number which is the gateway between the PSTN and the Internet, and then enters the desired telephone number. All sampling and coding takes place at the gateway. Voice packets are then carried over the Internet to a gateway close to the second user. The second gateway does all the decoding and conversion to an analog signal, which is then carried over the PSTN to the second user.

Products and Market Segments

In the previous sections we described the services that are possible with Internet telephony and the technical challenges to be solved to make IP telephony a success. In this section we discuss the emerging IP telephony products and the targeted market segments.

Internet telephony products appearing in the market can be classified as belonging to three market segments: carriers, enterprises, and small businesses/single users. The carriers trying to provide IP telephony service to the general public and businesses form a large market segment. The second market segment exists for enterprises who want to switch their intracompany telephone calls from the PSTN to their existing data networks. Individual users using PCs and conventional telephones can be classified in the third category. In the following subsections, we discuss these three market segments and some of the products for each market segment.

Carriers

Perhaps the largest market segment for IP telephony products is for next-generation telecommunications companies (next-gen telcos), whose sole purpose is to route voice traffic over the Internet, and conventional telephone carriers that have seen the potential of Internet telephony and are adopting it as one of their core services. Since these carriers try to switch calls from the existing PSTN to the IP network and vice versa, the primary product required for this market segment is the gateway between the PSTN and IP network. Among other things, the gateway addresses the interoperability issues, such as addressing and signaling, between the PSTN and IP networks.

The other two important functions for carriers trying to offer IP telephony service are network management and accounting of the hybrid IP-PSTN networks. While some products implement these functions in the gateway itself, it is possible to buy separate units that run these functions as separate modules in separate boxes. The basic architecture of IP telephony products and architectures targeting the carrier market segment is illustrated in Fig. 3. The central module facilitates communication between different modules and integrates all elements of the product to provide a single-product appearance.

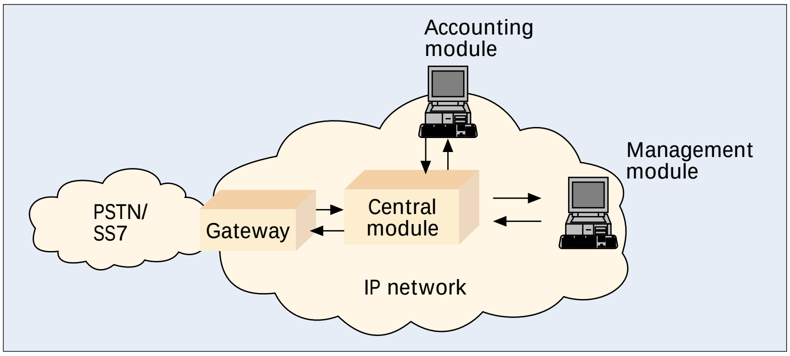

Figure 3. *The basic architecture of carrier-class IP telephony products.*

Product specification \ Vendor	Ericsson	Mockingbird Networks	Ascend
Product name	IP Telephony Integrated Solution	Nuvo100	Ascend Multi-Voice for IP
Target market	Small telecommunications companies, backbone network providers, corporations, Internet service providers, Web hosting companies, and next-generation telecommunications companies	Telecommunications carriers, ISPs, and large corporate customers	Next-Generation carriers, Internet Service providers and enterprise customers
Services/features	Least cost routing, dynamic call routing, silence suppression error concealment, echo cancellation support for multiple codecs, support for IVR, Remote Authentication Dial In User Service (RADIUS) based accounting and authentication, Web-based network management	Dynamic/static network routing, support for GSM coding standard and echo cancellation, billing using Bellcore Call Detail Records (CDR), distributed architecture for fault tolerance	Supports DTMF tone detection and generation. "absolute" QoS offering, multiple codecs support, billing, control, back office services and SS7 supportbased on Navis, an Ascend network management tool. Multiple codecs support
Line interface	E1 and T1	Up to four T1/E1 interfaces	Most PSTN interfaces, including T1, ISDN BRI and PRI
Signaling	H.323 SS7	ITU H.323 SS7	ITU-T H.323 for PC–PC and phone–PC support for SS7
Network management	SNMP-based	SNMP/TNM	Management using Navis

Table 2. *Characteristics of a few carrier class products.*

	Symphony	InfoGate 2.0	Vega100
Hardware platform	Compact hardware unit (0.5 kg)	233 MHz Pentium CPU for gateway, Windows NT 4.0 workstation	Compact Unit, built on DSP and RISC platform
Simultaneous number of calls	4	4	120
Interface	10 Base-T Up to 4 PSTN connections per single gateway	10/100 Base-T, Analog PSTN lines-RJ-11	10/100 Base-T, ISDN Primary Rate, T1 and E1
Standard compliance	H.323, SGCP, MGCP and H.GCP	H.323v2	H.323
Codecs Supported	G.711, G.723, G.726 and G.728	G.729A InfoSpeak proprietary voice coding	G.711, G.723, and G.729A
Other features	Supports complete PBX functionality	IVR, dynamic bandwidth allocation, silence detection, network management through SNMP.	Echo cancellation, jitter removal, voice activity detection, and lost packet compensation

Table 3. *Selected products for enterprise products.*

Software	Web site
NetMeeting	www.microsoft.com/netmeeting
Internet Phone	www.vocaltech.com
FreeTel	www.freetel.com
WebPhone	www.netspeak.com
CU-SeeMe	www.wpine.com

Table 4. *Pointers to some well-known Internet telephony software.*

In the last year or so, many vendors released IP telephony products targeting the carrier segment. While an exhaustive survey of the products is beyond the scope of this article, we list a few of them and summarize their features in Table 2.

ENTERPRISES

The enterprise market segment is different from the carrier segment in several ways. First, enterprises do not carry public calls and hence need products on a smaller scale. Second, enterprises would like to replace their traditional PBXs with IP telephony products that connect to their company intranets. Therefore, products targeting this market segment must provide basic PBX functionality, such as call hold and call transfer. Table 3 summarizes features of some of the many products targeting the enterprise market segment.

SMALL BUSINESS/SINGLE USER

This market segment represents the clients or users, rather than providers, of IP telephony, and includes home users or small businesses with one or two PCs. Products targeting this market segment are mainly designed to allow callers to make PC–PC long distance calls over the Internet at the cost of browsing the Internet. With improvement in technology, some of these products are now offering PC–phone calls, utilizing gateway services provided by carriers.

The basic architecture of these products include a voice card to be plugged in at the back of a PC, a microphone with headset and speakers or the conventional telephone set connected to the voice card, and the IP telephony software installed in the PC. The user needs to have Internet access, usually through the local ISP. There are many IP telephony softwares on the market for different platforms. Table 4 lists the official Web sites of some of the well-known software.

CONCLUSION

We have described the new services possible using Internet telephony. However, a number of technical challenges are still to be addressed to bring the quality of Internet telephony to a level comparable to traditional telephony . Solutions for some of the challenges exist but are difficult to implement because of the enormous size of the Internet. New protocols and techniques need to be incorporated in the current Internet to provide true QoS to end users. Research on incorporating QoS in the Internet and developing protocols to support IP telephony is currently being carried out in the Differentiated Service and IP Telephony groups of the IETF. The architecture, services enabled, and standards compliance of a number of commercial products have been described and compared in this article. The high number of available products for the different market segments indicates a very strong interest in and commitment to Internet telephony from the commercial sector.

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Research on incorporating QoS in the Internet and developing protocols to support IP Telephony is currently being carried out in the Differentiated Service and IP Telephony groups of the IETF.