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Voice Over IP

How can voice over the Internet claim a greater share of the worldwide phone market from the voice infrastructure dominated for more than 100 years by the public-switched telephone network?

Voice has been transmitted over the public-switched telephone network (PSTN) since 1878 while the U.S. long-distance market has grown to about $100 billion a year in business and residential demand. The desire of businesses and consumers alike to reduce this cost, along with the investment over the last decade in IP-based networks, public and private, has produced substantial interest in transmitting voice over IP networks. The possible re-emergence of Internet service providers (ISPs) and others as Internet telephony service providers (ITSPs) is likely to further increase competition among all phone service providers. Many communication technology vendors are rolling out hybrid IP/PBX systems. Both traditional and recently established carriers are beginning to offer voice over IP (VoIP) network connectivity to both business and residential customers (see the sidebar "PC-to-Phone Providers).

VoIP involves sending voice transmissions as data packets using the Internet Protocol (IP), whereby the user's voice is converted into a digital signal, compressed, and broken down into a series of packets. The packets are then transported over private or public IP networks and reassembled and decoded on the receiving side (see Figure 1). Residential customers can connect to IP-based networks by using the local loop from the PSTN or high-speed lines, including ADSL/DSL and cable modems.

Several recent industry surveys and projections estimate that VoIP could account for over 10% of all voice calls in the U.S. by 2004. It's likely to be used first in places with significant IP infrastructure or where cost savings are significant; an example might be a company with multiple sites worldwide connected through a private or public IP network. However, VoIP deployment may not be possible everywhere, as some countries restrict the use of VoIP to prevent harming their monopolistic telecommunication markets. VoIP might also be suitable for highly distributed companies or for companies with seasonally variable voice-service demand.

The idea of VoIP, or voice over the Internet or IP telephony, has been discussed since at least the early 1970s [6] when the idea and technology were developed. Despite this history, VoIP didn't establish a commercial niche until the mid-1990s. This gradual commercial development can be attributed to a lack of IP infrastructure and the fact that circuit-switched calling was and still is a much more reliable alternative, especially in light of the poor quality of early VoIP calls. In 1995, Vocaltec (www.vocaltec.com) produced the first commercially available VoIP product requiring both participants in the call to have the software on a PC as well as Internet access. Unfortunately, it did not allow traditional calls through the PSTN.

Following the rapid growth of the public mass-market Internet, especially the Web, during the early 1990s and accompanying investment in IP networking infrastructure by businesses, vendors, and carriers, VoIP has finally become a viable alternative to sending voice over the PSTN. A number of factors are influencing the adoption of VoIP technology. First and foremost, the cost of a packet-switched network for VoIP could be as much as half that of a traditional circuit-switched network (such as the PSTN) for voice transmission [9]. This cost saving is a result of the efficient use of bandwidth requiring fewer long-distance trunks between switches. The traditional circuit-switched networks, or the PSTN, have to dedicate a full-duplex 64Kbps

Figure 1. A possible scenario for VoIP for business customers.

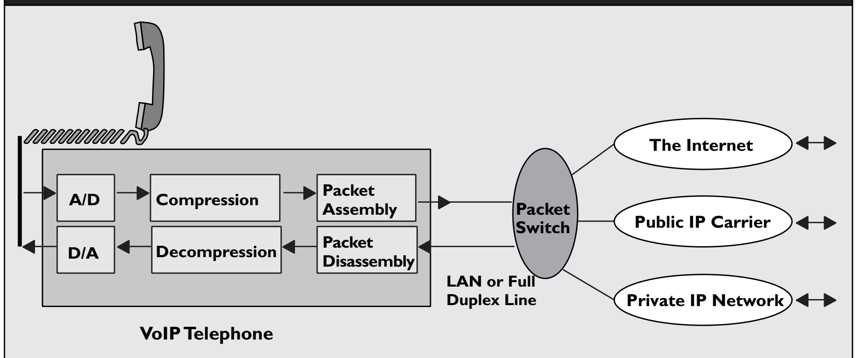


Table 1. A qualitative comparison of voice over PSTN and over IP.

Concept	Voice over PSTN	Voice over IP
Switching	Circuit switched (end-to-end dedicated circuit set up by circuit switches)	Packet switched (statistical multiplexing of several connections over links).
Bit rate	64Kbps pr 32Kbps	14Kbps with overhead*
Latency	< 100ms	200–700ms depending on the total traffic on the IP network. Lower latencies possible with private IP networks.
Bandwidth	Dedicated	Dynamically allocated
Cost of access/billing	**Business customers.** Monthly charge for line, plus per-minute charge for long distance, cost of PBX, and other telephony equipment. **Residential customers.** Monthly charge for line, plus per-minute charge for long distance, cost of simple phone.	**Business customers.** Cost of IP infrastructure, Hybrid IP/PBX, and IP phones. **Residential customers.** Monthly charge for line, plus monthly charge for ISP, cost of computer, and other equipment.
Equipment	Dumb terminal (less expensive); intelligence in the network	Integrated smart programmable terminal (expensive); intelligence not in the network.
Additional features and services	Requires reprogramming or changes in the network design but fast enough to add if advanced intelligent networks (AIN) are in use.	Easy to add without major changes, due to flexible protocol support, but standards are needed for traditional user services.
Quality of service	High (extremely low loss)	Low and variable, but traffic is sensitive depending on packet loss and delay experienced.
Authorization and authentication	Only once when the service is installed	Potentially required, per-call basis
Regulations	Many at federal and state levels	Few yet, but regulatory uncertainty; future regulations may reduce the cost advantages of VoIP.
Network availability	99.999% up time	Level of reliability is not known.
Electrical power failure at customer premises	Not a problem; powered by a separate source from phone company.	Will have problems, as equipment may be down. Power from other sources is not easy to obtain.
Security	High level of security because one line is dedicated to one call.	Possible eavesdropping at routers.
Standards/status	Mature (simplified interworking among equipment from different vendors).	Emerging possible problems in interworking.

*Only when speaker is talking

Figure 2. Managing temporary overflow of calls using VoIP.

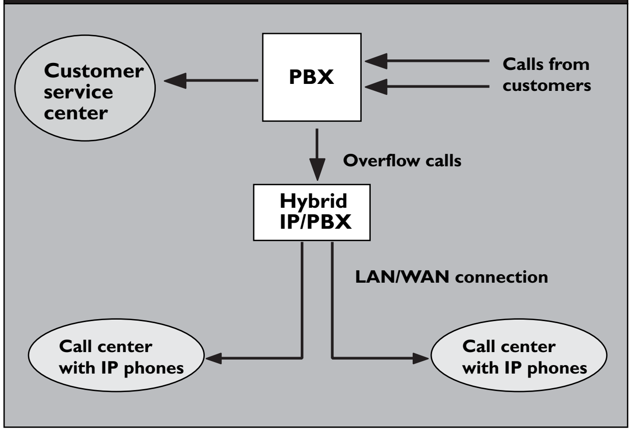


Table 2. The delay factors in VoIP.

Cause of Delay	Length of Delay
Processing at a switch/router	Variable, depending on the speed and traffic on the switch; usually 5–10msec per packet per hop.
Transmission time, or time to put packets online.	Packet size in bits divided by line speed in bits/sec.
Propagation delay, or the actual time it takes the signal to pass between two switches.	Fixed time for a given length of the segment.
Variable delays, or litter, introduced when packets get out of order and must be buffered and reordered before play.	Variable, depending on traffic on routers and switches in the IP network.
Speech encoding, compression, and decompression.	5–10msec per packet.

channel for the duration of a single call. The VoIP networks require approximately 14Kbps, as voice compression is employed, and the bandwidth is used only when something has to be transmitted. More efficient use of bandwidth means more calls can be carried over a single link, without requiring the carrier to install new lines or further augment network capacity; Table 1 compares voice over PSTN and over IP.

Besides cost savings and improved network utilization, VoIP offers other features, including caller ID and call forwarding, that can be added to VoIP networks at little cost [5]. VoIP allows Internet access and voice traffic simultaneously over a single phone line. This function could eliminate the need for two phone lines in a home, one for data and one for voice, by using the same line to carry all traffic without concern for missed calls or being disconnected from the ISP. Other high-speed media, such as ADSL and cable modems, can be used to carry both data and voice to IP networks while letting home customers use regular phone lines for voice calls to and from the PSTN. In this way, VoIP service offered by ISPs and ITSPs might indirectly benefit existing telephone companies and cable providers by increasing the potential number of ADSL and cable modem subscribers nationwide.

Long-distance carriers in the U.S. pay an average of $0.0171 per minute in interstate access charges to the regional Bell operating companies, that is, the local phone companies [8], a total of $9 billion a year. One current VoIP cost advantage is that ISPs pay no access charges, due to a U.S. Federal Communications Commission exemption under enhanced-service-provider regulations. However, any changes in regulation requiring ISPs and ITSPs to pay access charges or treat calls to ISPs as long-distance calls may diminish the VoIP cost advantage.

One VoIP application might involve managing temporary overload call volume for business users. Using a regular PBX, most traffic can be serviced with existing telephony equipment, and any excess or overload traffic can be routed to an IP/PBX system that can then be serviced by remote call centers with IP infrastructure (see Figure 2).

Technical Issues

Among the many technical issues in VoIP, a major one is end-to-end delay, or latency. To ensure good voice quality, latency for voice communication should not exceed 200 milliseconds, as demonstrated in the 1980s when carriers tried to offer voice services over geosynchronous satellites; users deemed the 270-millisecond delay unacceptable. However, under certain circumstances, VoIP might suffer from more latency, leading to unacceptable quality (due to the uncertainty as to whether the other person is talking, possibly leading to interruptions). Latency is influenced by a number of variables. First, other traffic on IP networks directly affects the delay for voice packets. Another is packet size, with smaller packets receiving less end-to-end delay, due to faster routing and other factors, while increasing overhead on the system. Latency is also related to the number of routers and gateways that packets have to travel through before reaching their destinations. Table 2 outlines the four most common causes of packet delay over IP-based networks, public and private.

Some VoIP systems send test messages to several routers over IP networks to find the paths with better quality in terms of less delay. These smart techniques do not always yield better quality, especially over public IP networks like the Internet, due to

Figure 3. Proposed evolution path for traditional PSTN carriers.

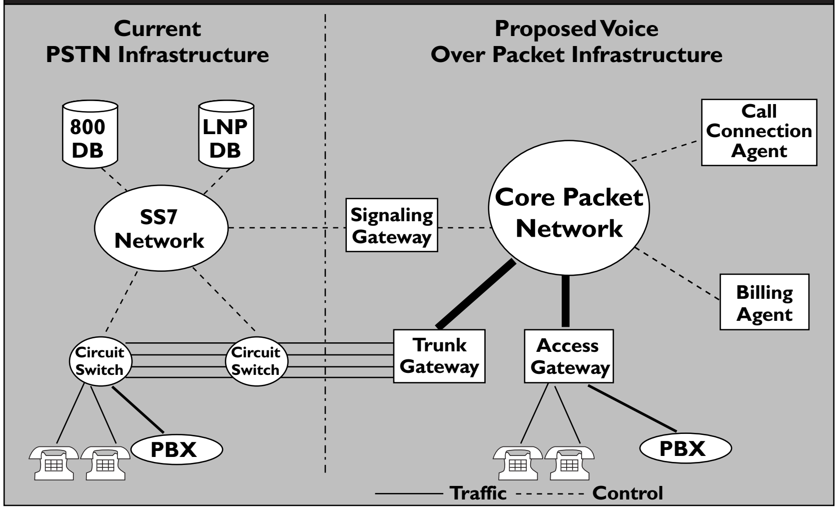


Table 3. Some VoIP implementations.

Approach	Description	Pros	Cons	Example
PC Web phones	Software that allows any PC with a sound card and a microphone to transmit voice to similarly equipped machines.	Only cost is computer and connection to ISP.	QoS issues, along with the requirement that both users have similar equipment/ software.	Vocaltec (www.vocaltec.com) and Net2phone (www.net2phone.com)
VoIP gateway	Used between the PBX and an IP network/LAN, translating and routing the calls to other gateways.	Cost savings in local and long-distance calls, better utilization of network resources.	QoS issues need to be addressed; high initial coast of the gateway equipment.	Quicknet (www.quicknet,net)
Public IP voice carriers	Phone companies that completed bypass the PSTN and provide just VoIP. May still need the local loop.	Currently immune from line-access charges, cheaper phone services, more advanced features; reduced infrastructure costs.	Not as reliable as PSTN; QoS issues still need to be resolved; only available in limited areas; future regulation may affect.	Net2phone (www.net2phone.com) allowing PC-to-phone calls (not the other way round) at low rates
Voice-enabled browsers	Combining voice access with Web browsers.	Good for services like live customer service for Web sites and e-commerce solutions.	Regular dialup connections limit bandwidth available for the combined services.	Both Netscape and Explorer have plug-ins available.

possible rapid fluctuations in the amount of traffic and resulting increase in delays experienced by the people speaking and listening on the line.

VoIP systems use the User Datagram Protocol (UDP) as a transport layer protocol on top of IP to avoid acknowledgments for lost packets. Acknowledgments trigger undesirable retransmission of voice packets and increase network traffic (and end-to-end delay) and thus affect the quality of service (QoS) for VoIP. Some packet loss is tolerable; for example, many voice encoders can handle up to 1% packet loss [2].

specialized equipment, Internet telephony gateways can be used where two users communicate without having a computer at either of their locations. A gateway's basic architecture involves a user connection via the PSTN. The gateway computer then searches for another gateway computer near the target location and makes a connection using circuit switching. When this connection is made, the second gateway utilizes the local PSTN to complete the collection of the call. Though this type of call isn't completely IP, it does suggest possible future solutions for integrating the current PSTN and the VoIP system. Table 3 compares four implementations for supporting VoIP.

However, data packets traveling through the Internet may not be secure and may require encryption, adding overhead by increasing the necessary bit rate beyond 14Kbps, hence reducing the bit rate advantage of VoIP over PSTN. Encryption also increases the end-to-end latency caused by the processing delay for encryption and decryption.

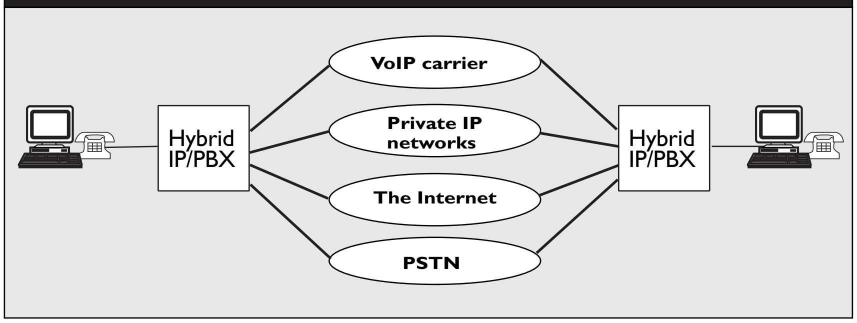
Meanwhile, technology support for VoIP has begun to mature on a number of fronts. The newer generations of routers and switches are faster and better able to handle the added load of real-time data packets. Beyond the advances in compression and equipment, protocol support in the form of the Resource Reservation Protocol (RSVP) and IP version 6 (IPv6) are also starting to mature. These protocols offer ways to prioritize voice traffic over the Net, helping improve QoS, especially when the network is congested.

Protocol Support

Just as in conventional telephony, VoIP needs a connection between users, though in the case of VoIP, a virtual connection. VoIP architecture col is needed to set up individual sessions for voice connections between users [2]. Once a session is established, a transport protocol can be used to send the data packets. Directory access protocols are another important part of VoIP, providing routing and switching information for connecting calls.

A signaling protocol handles user location, session establishment, session negotiation, call participant management, and feature invocation. Session establishment is invoked when a user is located, allowing the call recipient to accept, reject, or forward the call [6]. Session negotiation helps manage different types of media, such as voice and video, transmitted at the same time. Call participant management helps control which users are active on the call, allowing for the addition and subtraction of users. The signaling protocol also involves feature invocation, at which time call features, such as hold, transfer, and mute, are controlled.

Figure 4. Possible coexistence scenario for PSTN and VoIP.



The Realtime Transport Protocol (RTP) can be used to support the transport of real-time media, including voice traffic, over packet networks. RTP-formatted packets contain media information and a header, providing information to the receiver that allows the reordering of any out-of-sequence packets. Moreover, RTP uses payload identification to place an identifier in each packet to describe the encoding of the media so it can be changed in light of varying network conditions [7]. The Real Time Control Protocol (RTCP), a companion protocol for RTP, provides QoS feedback to the sending device, reporting on the receiver's quality of reception. The Real Time Streaming Protocol (RTSP) can be used to control stored media servers, or devices capable of playing and recording media from the server. This added RTSP-based control allows the integration of voice mail and prerecorded conference calls in VoIP environments. The ability to integrate these advanced services is important to the future growth of VoIP. The Session Initiation Protocol (SIP) can be used to establish, modify, and terminate multimedia calls.

To encourage rapid, widespread deployment of VoIP services, several standards bodies have generated agreements based on groups of existing protocols and standards. The two most important are the H.323 recommendation from the International Telecommunication Union and Media Gateway Control Protocol (MGCP) from a branch of the Internet Engineering Task Force. Neither is a standalone protocol but relies on other protocols to complete their jobs [1]. The H.323 architecture is based on four components: terminals, gateways, gatekeepers, and the multipoint control unit (MCU). Gateways are used for protocol conversion between IP and circuit-switched networks. Gatekeepers are used for bandwidth management, address translation, and call control. H.323 provides a foundation for audio, video, and data communications across IP-based networks, including the Internet. Complying with H.323 enables different multimedia products to interoperate. H.323 depends on other standards, such as H.245, to negotiate channel usage and capabilities, modified Q.931 for call signaling and call setup, Registration Admission Status for communicating with a gatekeeper, and RTP/RTCP for sequencing audio/video packets. The MCU supports multicast conferences among three or more end points by using H.245 negotiations to determine users' common capabilities [1].

The Media Gateway Control Protocol (MGCP) defines communications among call agents (media gateway controllers) and telephony gateways. Call agents have the intelligence for call control and other functions and manage telephony gateways used for protocol conversion. A call agent in MGCP is analogous to a gatekeeper in H.323 [1]. The MGCP can use the Session Initiation Protocol (SIP), which uses the HTTP format to allow a user to initiate a call to be initiated by clicking on a browser.

Although H.323 and the MGCP have been standardized by two different standard-setting bodies, some of their functions are quite similar. Both the gatekeeper in H.323 and the call agent in MGCP manage and control gateways and participate in setting up, maintaining, and terminating the VoIP's telephone connection. The MGCP can also be used as part of H.323 for simplified interworking.

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