Abstract

Internet Telephony is a powerful and economical communication options by combination of the telephone networks and data networks. The ability to use IP network to carry traditional telephone traffic brings both challenges and opportunities to all the long-distance telephone service companies and their resellers. Although a lot of difficulties exist, from technology to social issues, few people doubt that it will bring a great change in communication field and bring a new huge market.

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General Information

● The birth of Internet Telephony

CALL ANYWHERE! TALK FOREVER! NEVER PAY LONG DISTANCE!

Internet telephony uses the Internet to send audio between two or more computer users in the real time, so the users can converse. Vocaltec introduced the first Internet telephony software product in early 1995. Running a multimedia PC, the Vocaltec Internet Phone (and the numerous similar products introduced since) lets users speak into their microphones and listen via their speakers.

Two and half years passed, Internet telephony technology has caught the world's attention. The technology has improved to a point where conversations are easily possible. And it continues to get better. Dozens of companies have introduced products to commercialize the technology, and virtually every major telecommunications company has launched research to better understand this latest threat to their markets.

● Facts and numbers

[http://www.newbridge.com/]

○ In 2010
  ■ 50% of North Americans will use the Internet daily
  ■ a trans-Atlantic phone call will cost 4 cents per minute
  ■ 28% of all international voice communication will take place over the Internet

○ Internet Today
  ■ 9.5 million hosts worldwide
  ■ 240,000 domains worldwide

○ Internet Capacity and Phone Capacity (bytes/month)
  ■ Internet traffic in 1996: $10^{13}$
  ■ International phone traffic in 1994: $10^{15}$

○ Who is using Internet Telephony today?
  ■ 15,000 users today
potentially 20+ million users when Netscape bundles it

Economic Differentiators

- Circuit switching costs: $.15/kbit
- Packet switching costs: $.04/kbit

The factors making Internet Telephony possible

- Voice quality is increasing, thanks to new codec technology;
- There are ongoing improvements in compression techniques;
- Full-duplex PC sound cards enable two-way simultaneous calls;
- The typical PC is getting more and more powerful, making it possible to perform processor-intensive functions without specialized hardware.

Three generations

The first generation systems were introduced to enable voice conversations between users with telephony software-equipped-computers. The software provides the functions of data compression and translation to IP packets, and sends the packets over the Internet to the destination computer where the process is reversed.

The second generation emerged after the development of technologies which overcame difficulties with PSTN interface protocols and the mapping of IP addresses to E.164 phone numbers. Using servers at the ISP's premises, these systems enable a user with a computer and an Internet connection to call any number on the PSTN.

The third generation phone gateways makes Internet telephony start to receive serious attention. These gateways provide a two-way interface between the PSTN and the Internet and allow voice conversations between users with standard phones, without the need of computers or Internet access.

Gateways are the key to bringing Internet telephony into the mainstream. By bridging the traditional circuit-switched telephony world with the Internet, gateways offer the advantages of Internet telephony to the most common, cheapest, most mobile, and easiest-to-use terminal in the world: the standard telephone. Gateways also overcome another significant Internet telephony problem, addressing. To address a remote user on a multimedia PC, you must know the user's Internet Protocol (IP) address. To address a remote user with a gateway product, you only need to know the user's phone number.

Ideal Internet Telephony should be

- high-volume call processing within and between public and private networks
- high-volume, real-time translation between IP and circuit-switched networks
economic scaleability
- broad acceptance and implementation of standards

Right now, a common acceptable requirement for Internet Telephony is:

the ability to establish a connection to the other end 95 percent of the time and a sound quality of at least 4-8 kbps, which is sub-toll quality.

- What users can get now
  - point-to-multipoint voice
  - data conferencing
  - application sharing
  - long-distance telephone savings

- Three factors to success
  - Ease of connectivity in terms of dialing another party directly, whether or not the party is on the Internet
  - An open set of standards that operates independently of which product is used
  - The addition of telephony features that provide a competitive differentiation, other than price, from conventional long-distance services

- Research groups
  - LBNL Network Research Group
    WWW: [http://ee.lbl.gov/](http://ee.lbl.gov/)
  - Cambridge University Engineering Department SVR Group
  - Tarifica
  - MICE Multimedia Index
    WWW: [http://www-mice.cs.ucl.ac.uk/multimedia/projects/mice/mice_home.html](http://www-mice.cs.ucl.ac.uk/multimedia/projects/mice/mice_home.html)
  - Telemedia, Networks, and Systems Group
  - MIT Advanced Network Architecture Group
Current Problems and Solutions

● Current problems

○ Standard

The biggest difficulty that is facing VON (Voice On the Net) technology is the interoperability between Internet telephony products and interworking with legacy PSTN-based systems and services. Currently, no two products are compatible. Users who want to make Internet phone call have to have the same kind of software. Standard development and adoption are the key to ensuring interoperability. At this time, the primary specification issues that have to be resolved are related to the codec format, the transport protocol, and directory services.

○ Quality

Voice performance is measured by delay. Calls on the public switched telephone network usually exhibit 50- to 70-millisecond delay. That latency increases substantially on the Internet, where it typically ranges from 500 milliseconds (an eternity when it comes to voice traffic). So right now, some users still complain the quality when they use the Internet Telephony.

Latency affects the pace of the conversation. Humans can tolerate about 250msec of latency before it has a noticeable effect. Today's Internet telephony products exceed this latency, so most connections sound like traditional calls routed over a satellite circuit (which are usable, but require some getting used to). Even today, the products are well suited to many applications.

○ Capacity

The Internet is an open network of many different ISPs' networks. Consequently, there is no way to get network bandwidth, packet sequence and latency guarantees. One of the main parameters affecting the quality of service on the Internet is lost packets. Packet loss is a persistent problem, particularly with the increasing popularity, and therefore increasing load, of the Internet. Packet loss can occur for a number of reasons. Network congestion due to bandwidth limitation or traffic overload is the main reason. Inadequate network access links, especially local ISP connections to the Internet backbone, are already causing chronic bandwidth congestion.

The heavy traffic loads are also straining the backbone infrastructure and leading to traffic collisions. The network overload also results in delays in packet transmission, with packets arriving too late at the receiver to be played back and are therefore, discarded. Congestion of routers and gateways also leads to packet discards. Error performance of underlying transmission paths is also an affecting factor with packet losses increasing dramatically on transcontinental links involving local ISPs and high error rate local networks. Finally, another reason for packet loss is the heavy loading of the servers leading to scheduling difficulties in multi-task operating systems. Current Internet telephony applications repair
lost packets with silence, which leads to the speech clipping effects currently experienced by many Internet telephony users. Since comparatively large packets are used, even the loss of individual packets has a serious impact on the intelligibility of speech.

- Social issues

Regulation of Internet telephony is still largely a question mark. Traditionally, telephone service has been heavily regulated. In most countries, governments or government-sanctioned entities retain monopolies for provisioning telephone service. Moreover, even without the Internet, telephony service is deregulating in many countries around the world, although the deregulation process is time consuming and heavily political.

Internet telephony has stirred fears from carriers throughout the globe, many of whom are reacting by seeking regulatory protection from the new technology. In the US, the America's Carriers Telecommunications Association (ACTA), a coalition of the smaller long distance carriers, filed a petition with the Federal Communications Commission (FCC).

ACTA argues that the major reason Internet telephone calls are cheaper than traditional circuit-switched calls is the access charge exemption ISPs enjoy. Hence, they argue, the Internet-based providers have an "unfair" advantage in offering cut-rate long distance phone service. Fortunately for the Internet telephone industry, and the Internet industry itself, the FCC does not seem to agree.

- Solutions to current problems

- Standard

The ITU H.323 recommendation which was ratified in May of last year, defines the core technology for VON (Voice On the Net) applications. H.323 is based on the real-time protocol (RTP/RTCP) and is an extension of H.320 to include packet switched networks. H.323 is composed of a set of recommendations including G.729 specifications for audio codecs, ratified by the ITU in November 1995. The initial objective of the recommendation was to identify a voice compression algorithm that could transport voice with quality equivalent to 32 kbps ADPCM at only one-fourth of the bandwidth. The ratified standard compresses signals to 8 kbps while delivering 4 kHz speech bandwidth with toll quality. The adopted CS-ACELP algorithm meets stringent requirements and objectives such as toll quality under clean channel conditions, robust performance in the presence of random bit errors and detected erased frames. In fact, the algorithm delivers an exceptionally high level of voice quality with minimal delay and hopes are high that this state-of-the-art vocoder will be adopted by major vendors.

H.323 also calls out T.120 for data conferencing. T.120 enables products from different vendors to interoperate without terminals assuming prior knowledge of the other systems. It specifies the network interfaces and wire formats, along with a data transmissions facility.

As for the transport protocol, RTP is finding acceptance as the standard means of transporting time related applications over the Internet. RTP has been introduced as a new protocol layer to provide support for applications with real-time properties including timing.
reconstruction, loss detection, security and content identification. RTP provides a
time-stamp and control mechanisms for synchronising different streams with timing
properties.

Since RTP does not address the issue of resource reservation or QoS control, it relies on the
resource reservation protocol (RSVP) to provide these capabilities. Currently a draft
standard protocol, the RSVP is part of various efforts to enhance the current Internet
architecture with support for QoS flows to be able to handle real-time traffic more reliably.
RSVP is a new signalling protocol to be implemented in Internet routers to provide for new
classes of services by reserving paths for sessions on an end-to-end basis. This is achieved
through three main functions, admission control, packet classification and packet
scheduling. Each vendor has, however, its own strategy for performing these functions to
provide the controlled load service at a first stage and the ultimate guaranteed service later.
Although some vendors have already announced plans for providing telephony gateways
with RSVP support, the reality is that after 13 consecutive revisions, RSVP specifications
are still only a draft and testing of products and services for interoperability in public
networks is still in the planning stages.

- Quality

Serveral ways are used to improve the quality:
- improvements in protocols: RSVP
- dedicated service lines with managed traffic loads
- co-locating telephone access with backbone nodes
- bigger routers
- new network architectures

- Capacity

Right now, the average routers hop number of trans-Atlantic call are 20 to 30. Since delay
increase with each router hop, one solution is to increase routing speed rather than putting in
more routers. People prefer the bigger routers is the solution to capacity. A gigarouter can
handle at least 10 times more traffic than a conventional router.
- Traditional routing: 500,000 - 1,000,000 packets per second
- Gigarouting: 5,000,000 - 20,000,000 packets per second

The per-packet cost of gigarouting is 3 to 4 times less than traditional routing.

- Social issues

Although there are a lot of society issues, people always encourage new technology. The
trend of technology development can not be prevented by anything. More than 100
companies are involed in the development of Internet Telephony, including AT&T, MCI and
Sprint. These long distance service giants do not regard the Internet Telephony technology
as their threats but opportunities. Thinking about the supports from industry and keeping
promises that encourage competitions in 1996, FCC deny the petition from ACTA and provides the official support to Internet Telephony.

Future Developments

The future is exciting. There are two dimensions in which the Internet telephony can be improved. One dimension is technology itself. It will go to much more high quality, big capacity, multi-function, etc. The other dimension is the combination with other technology, such as Intranet, to confirm a new product or tool to meet the increasing needs to the communication. For example, some forward-thinking organizations will begin deploying Internet telephone gateways for internal use, particularly for things like internal helpdesks. Calls will be routed over Intranets.

Right now, S.100 and other standards from the Enterprise Computer Telephony Forum are provided. They are important when building business communication platforms with integrated IP phone capability. Nearly 75% of U.S. companies already have or plan to incorporate intranets, according to Business Research Group (Newton, Mass.). So it can be foreseen that in the near future, the Internet Telephony will work with Intranet together. It will come to most employees' desk and everybody will familiar with them.

Products

The Top VON Products of 1996 are:

<table>
<thead>
<tr>
<th>Internet Telephony</th>
<th>Points</th>
<th>URL</th>
</tr>
</thead>
<tbody>
<tr>
<td>1 VocalTec Internet Phone</td>
<td>(1176)</td>
<td><a href="http://www.vocaltec.com">http://www.vocaltec.com</a></td>
</tr>
<tr>
<td>2 TeleVox</td>
<td>(316)</td>
<td><a href="http://www.voxware.com">http://www.voxware.com</a></td>
</tr>
<tr>
<td>3 NetMeeting</td>
<td>(314)</td>
<td><a href="http://www.microsoft.com/netmeeting">http://www.microsoft.com/netmeeting</a></td>
</tr>
<tr>
<td>4 Intel Internet Video Phone</td>
<td>(216)</td>
<td><a href="http://www.intel.com">http://www.intel.com</a></td>
</tr>
<tr>
<td>5 CU-SeeMe</td>
<td>(202)</td>
<td><a href="http://www.wpine.com">http://www.wpine.com</a></td>
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<td>6 CoolTalk</td>
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<td>7 WebPhone</td>
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<td>8 FreeTel</td>
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<td>9 VDOPhone</td>
<td>(111)</td>
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<tr>
<td>10 Net2Phone</td>
<td>(107)</td>
<td><a href="http://www.net2phone.com">http://www.net2phone.com</a></td>
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Internet Telephony Software

Windows Clients

- **Speak Freely (John Walker)**
  http://www.fourmilab.ch/netfone/windows/speak_freely.html
- **Internet Phone (VocalTec)**
  http://www.vocaltec.com/
- **Powwow (Tribal Voice)**
  http://www.tribal.com/
- **Televox (Voxware)**
  http://www.voxware.com/
- **Internet Call**
  http://www.gathertalk.com/
- **Softphone (SilverSoft)**
  http://www.pak.net/
- **Webphone (Internet Telephone Company)**
  http://www.itelco.com/
- **Farsite (databeam)**
  http://www.databeam.com/Products/FarSite/
- **PGPfone (MIT)**
  http://web.mit.edu/network/pgpfone/
- **CoolTalk (Netscape/Insoft)**
  http://live.netscape.com/
- **Microsoft NetMeeting**
  http://www.microsoft.com/ie/
- **Internet Conference (Insitu)**
  http://www.insitu.com/
- **Precept Software, Inc.**
  http://www.precept.com/
- **IRIS Phone**
  http://alpha.acad.bg/

Mac Clients

- **PGPfone (MIT)**
Unix Clients

- Nautilus Secure Phone
  - http://www.lila.com/nautilus/
- Vat
  - ftp://cs.ucl.ac.uk/mice/videoconference/vat/
- Mtalk (Linux)
  - ftp://sunsite.unc.edu/pub/Linux/apps/sound/talk/