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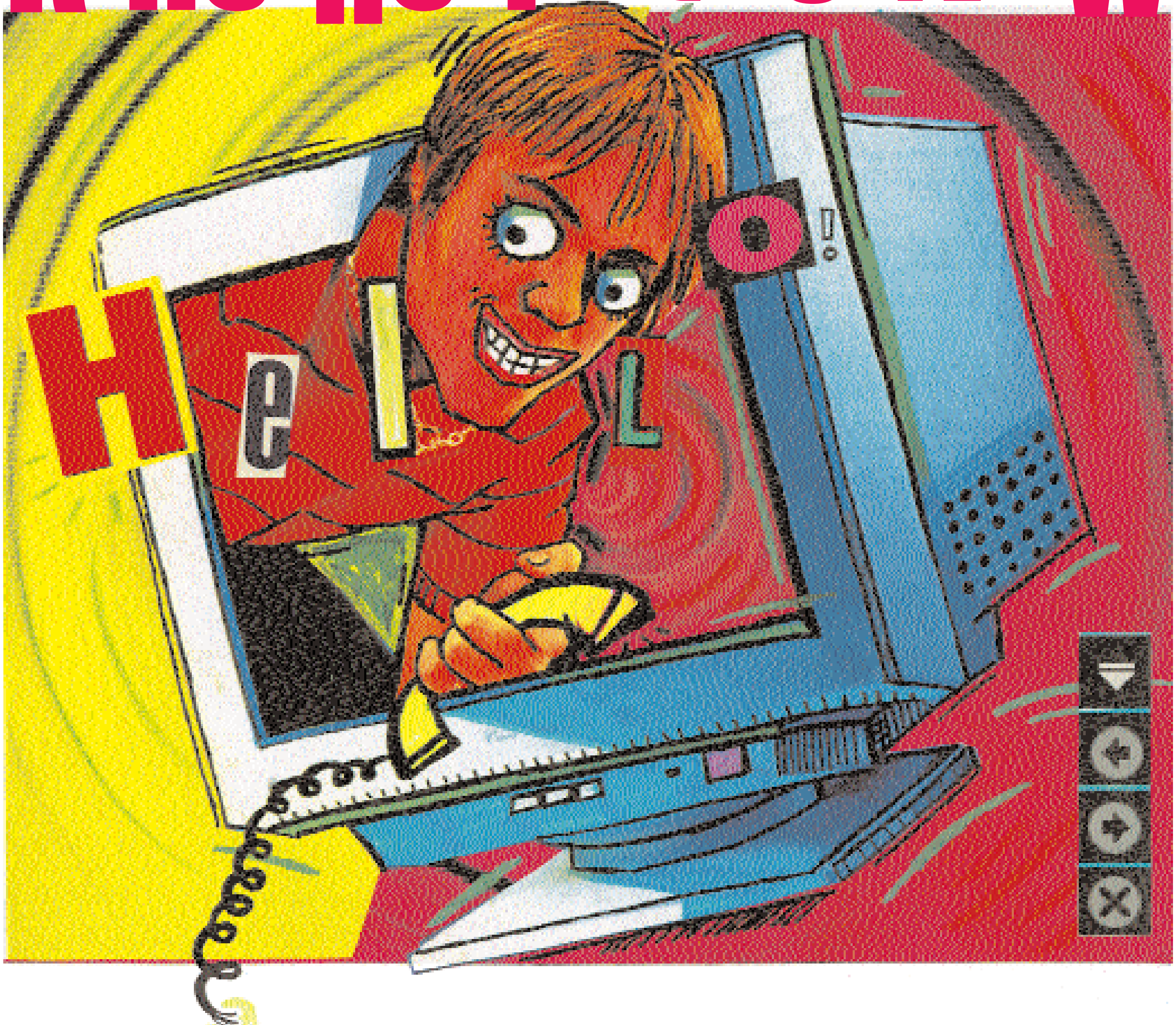
Ringing in the New



FARZANA COOPER

A phone call from Mumbai to New York cheaper than a phone call from Mumbai to Delhi? It's true! Technologies such as Voice over IP add a whole new dimension to traditional means of communication

Chatting on the phone with your overseas friend or associate has now become as economical as calling up your next-door neighbour. The evolution of standards has paved the way for smooth transmission of voice and video on Internet Protocol (IP) networks, and service providers in many countries have been restructuring their network services accordingly.

Voice, video and data traffic used to be mutually exclusive because they evolved independently of one another. Besides, the applications of these fundamental services used to be inherently different. The trend today, however, is towards greater interactivity among these technologies, and the convergence of voice, video and data traffic has simplified and enhanced the way we communicate.

Yet to become 'legal' in India, Internet Telephony has caused many a heated debate about telecom pricing policies around the world. There is a widespread feeling that the rates charged by telephone companies are not justified, especially in view of the easy transmission of voice and video over the Internet. Are VSNL and MTNL listening?

What is Internet Telephony?

In Internet Telephony, calls are made over normal telephone networks, but are delivered in part through the Internet, instead of a Public Service Telephone Network (PSTN). This kind of routing enables these calls to cost less than normal phone calls. Of course, the voice quality may not be as good.

Internet Protocol can be used to communicate across any set of interconnected networks and is particularly ideal for Local Area Networks (LANs) as well as Wide Area Networks (WANs). A call via IP Telephony would now require a normal telephone and not necessarily a computer. Depending on the IP telephony service, one may have to dial a special code to route the call to the service.

The first part of the call goes over the normal Public Switched Telephone Network. This call is then sent to a special IP voice gateway that can be located anywhere, but is most often at the telephone company. This gateway converts voice signals to digital data and compresses it, since the data files of an uncompressed voice call could be too large to deliver in time across the Internet.

The compressed voice signals are further broken by the gateway into what are called IP packets and sent across the Net using TCP/IP. These voice packets are sent to the IP voice gateway nearest to the destination of the phone call. The receiving IP voice gateway uncompresses the voice packets, converts them back to their original form and sends them through the PSTN. To the person at the other end, this would sound just like a normal phone call.

Speak Easy

Like cable television and online credit card transactions, Internet telephony is making itself useful—in spite of governmental indifference, even active disapproval. Though the idea of voice over private IP networks is widely accepted today, many countries do have regulations against Internet telephony.

However, this does not mean that the rules are always followed. The best part of using Internet Telephony is the price factor. You need to pay only for your Internet connection, just as you would to browse the Web or send e-mail. But do not throw out your telephone yet.

Though a number of products enable you to talk over the Net, not all of these can communicate with each other. So unless you have the same software as your caller, there is no standard as yet that helps you make or receive Internet calls easily.

GLOSSARY

Bandwidth: It is a measure of information carrying capacity of a communications channel; the higher the bandwidth, the greater the amount of information that can be carried.

Codec: A method or device that converts analog signals to digital signals. Codecs are used in digital telephone systems to transmit voice over digital lines.

Frames: A packet data format consisting of streams of bits. A frame includes start bits, data bits, an optional parity bit, and stop bits in addition to the payload.

Gateway: A system that interconnects networks (or applications) that communicate using different protocols, and bridges their differences by transforming one protocol into another.

IP (Internet Protocol): This provides for the routing of packets of data over multiple networks on their way to their final destination.

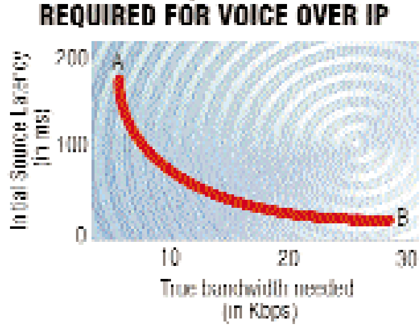
ISDN (Integrated Services Digital Network): A digital switched network that providers very fast, simultaneous transmission of voice, data and images over a single telephone line. It enables the transport of hundreds of communications and multipage faxes as well as medium-quality video images, at once, in seconds instead of minutes.

Jitter: A signal variation in timing caused by the constancy of a source clock rate or differences between a source s clock and a receiver s clock.

Latency: The amount of time before a requested network or communications channel is available for transmission or the amount of time required for a transmission to reach its destination.

Packet: On the Internet, data is broken up into small chunks called packets; each

LATENCY V/S BANDWIDTH REQUIRED FOR VOICE OVER IP

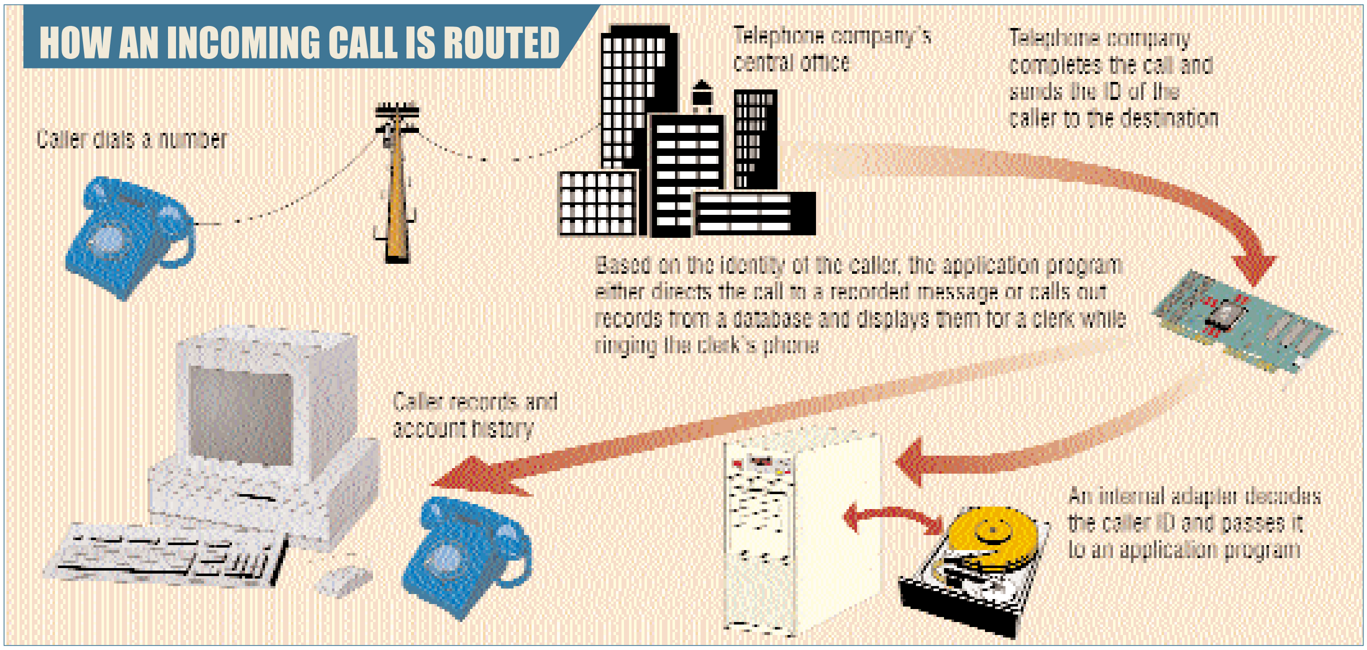


A sound technique

In a Voice over IP (VoIP) system, the analog voice signal is picked up by a headset-mounted microphone and sent to an audio processor within a PC. Here, a software or hardware code performs analog-to-digital conversion and compression. The nominal bandwidth required for telephone-type voice ranges from 2.9 Kbps to 13 Kbps, which is the GSM cellular standard. The code is output into packets that do not operate continuously. Instead, they sample the voice over a short period of time—these are called 'frames' and are like little bursts of data.

Waiting longer to fill the IP datagram reduces overhead, which in turn reduces the true bandwidth needed to send the digitised voice. However, this waiting creates a corresponding delay at the source, and too much of this 'latency' makes for a difficult conversation. This, along with jitter (changes in latency) has

Graphics: JAYA SHETTY



a degrading effect on voice quality. Therefore, real-time voice quality is difficult to maintain over a large wide area packet network.

A solution to circumvent this involves buffering the code data at the receiver. A large buffer can be filled irregularly but emptied at a uniform rate. This permits good quality reproduction of voice, and is also called audio streaming. (*See 'What You Get Is What You See' CHIP April 1999*). Internet 'radio' stations utilise this technique to deliver audio with FM radio broadcast quality. All the same, it may be some time before we switch en masse to Internet Telephony.

Better never than late

Termed as one of the most powerful emerging technologies that has been developed along with Voice over IP, Video over IP supports a variety of applications ranging from distance learning and collaboration, to conferencing and video-on-demand.

Like audio transmission, video transmission over a packet network can be greatly simplified by buffering at the destination, though the overall data rate does not change for the quality of video and audio desired. The quality can be maintained over a loaded network that does allows guarantees of bandwidth. Quality of Service (QoS) techniques are designed to balance the needs of voice, video, and data across the network and reserve a portion of the network bandwidth for the predictable data traffic.

A voice or video packet that misses its 'play' window and arrives late is not of much use. A network would rather drop the packets that are late, than use them and clog the already scarce network sources.

Videoconferencing

One of the chief applications of video over IP, videoconferencing offers better communication, easier management and shorter schedules. One can use the same meeting tools–auxiliary cameras, document cameras and electronic whiteboards–even as you access more than one site and still have one screen.

Videoconferencing is not merely point-to-point communication; it is a much richer three-dimensional medium. High-quality interactive video conferencing increases the productivity and competitiveness of business resources, regardless of time and distance. The server monitors your meetings, constantly searching for the site that has clearest and most continuous audio. It automatically sends the video from that site to all the other sites. When someone at a different site starts talking, the server switches everyone's view to that site. Each time the view switches to show a new speaker, the new speaker continues to see the previous speaker.

Future Speak

Internet Protocol (IP) will become the dominant networking protocol over the next decade. IP networks are on a steep slope of innovation that will gradually make them the long-term carriers of all traffic types worldwide. It has already begun to make its presence felt in India in a small way. Tata Infotech, for instance, has implemented this service in 14 centres across India.

Though Voice over IP has cut down long-distance telephone costs, infrastructure, operational costs and scalability, it is not without its drawbacks. Packets are sometimes lost, or take alternative routes, jumbling even coherent sentences. There could also be problems at the user end (for example, the PC could pick up background noise) and selection of software would require careful consideration of parameters.

Nevertheless, every major software, hardware and information technology industry has already implemented IP products and is spending significant dollars to develop next-generation IP products and services.

Some of the software that vendors have been using–such as information, connection, call servers and gateways that redirect the calls from PCs to standard phones–are now being targeted at corporate companies.

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