(12) INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(19) World Intellectual Property Organization
International Bureau

(43) International Publication Date
7 September 2001 (07.09.2001)

PCT

(10) International Publication Number
WO 01/65786 A1

(51) **International Patent Classification**[7]: **H04L 12/66**, 12/43, H04J 3/02, H04L 1/00

(21) **International Application Number:** PCT/US00/28386

(22) **International Filing Date:** 13 October 2000 (13.10.2000)

(25) **Filing Language:** English

(26) **Publication Language:** English

(30) **Priority Data:**
09/516,857 2 March 2000 (02.03.2000) US

(71) **Applicant: 3COM CORPORATION** [US/US]; 5400 Bayfront Plaza, Santa Clara, CA 95052-8145 (US).

(72) **Inventors: EREKSON, Rich**; 4760 South 3350 West, Roy, UT 84067 (US). **ALDRIDGE, Timothy, W.**; 939 East Edgefield Road, Sandy, UT 84094 (US). **PRICE, Tim, U.**; 568 East Water Lily Drive, Salt Lake City, UT 84106 (US).

(74) **Agents: MASCHOFF, Eric, L.** et al.; Workman, Nydegger & Seeley, 1000 Eagle Gate Tower, 60 East South Temple, Salt Lake City, UT 84111 (US).

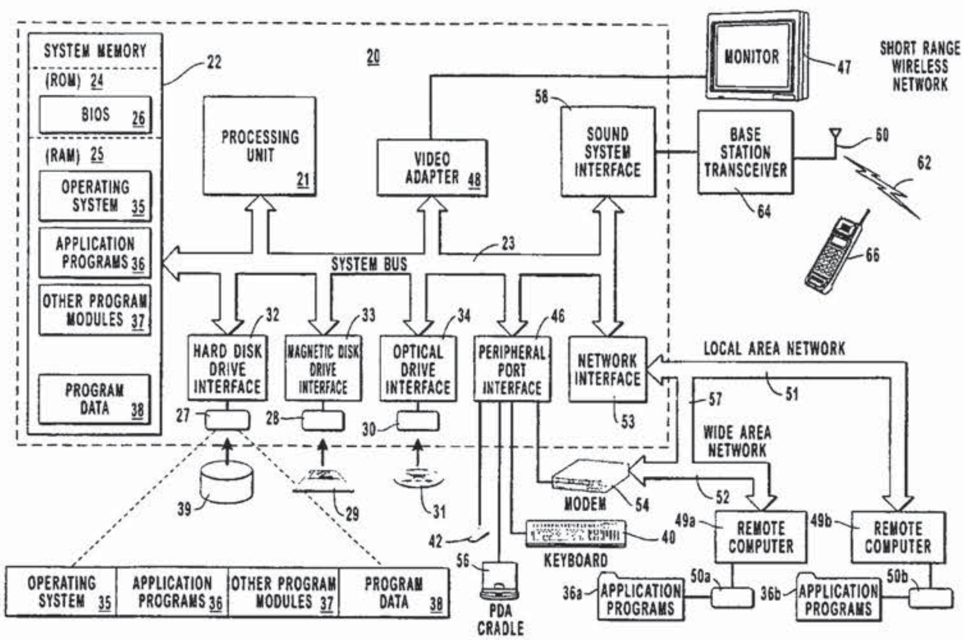
(81) **Designated State** *(national)*: JP.

Published:
— *with international search report*

For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.

WO 01/65786 A1

(54) **Title:** VOICE-OVER-IP INTERFACE FOR STANDARD HOUSEHOLD TELEPHONE



(57) **Abstract:** The present invention enables a traditional analog telephone (Fig. 1) to be used with VoIP applications. For example, the user should connect their standard 900 MHz telephone to this invention, establish a VoIP call and enjoy the freedom of movement their cordless telephone provides. The preferred embodiment of the present invention minimizes overhead to the host computer via a dual CODEC modem that incorporates a DSP (610) capable of simultaneous communication with the two CODEC modules. This architecture facilitates a latency and communication overhead reduction as the analog voice signals effectively "stream" from the first CODEC (620) to the second CODEC (630).

VOICE-OVER-IP INTERFACE FOR STANDARD HOUSEHOLD TELEPHONE

BACKGROUND OF THE INVENTION

1. The Field of the Invention

This invention is in the field of Voice Over Internet Protocol (VoIP) communications and, more particularly, to a system and method of interfacing a standard telephone to a VoIP compatible communication network.

2. The Prior State of the Art

Voice Over Internet Protocol (VoIP) is an emerging technology that allows the systems and wires that connect computer networks to act as an alternative to phone lines, delivering real-time voice to both standard telephones and PCs. VoIP allows an individual to utilize their computer connection to transmit voice encapsulated data packets over available local communication lines, such as the Internet, to another user on another computer, thereby creating a long distance phone call at a local connection price.

How VoIP works

In a Voice-over-IP (VoIP) system, the analog voice signal is typically picked up by a microphone and sent to an audio processor within a PC. There, either a software or hardware CODEC performs analog-to-digital conversion and compression. Considerable research has been devoted to voice compression schemes that are well know to those skilled in the art. The nominal bandwidth required for telephone-type voice ranges from 2.9 Kbps (RT24 by Voxware) to 13 Kbps (GSM cellular standard).

In placing the CODEC output into packets, there is a trade-off between bandwidth and latency. CODECs do not operate continuously. Instead, they sample the voice over a short period of time, known as a frame. These frames are like little bursts of data. One or more frames can be placed in a single IP datagram or packet, and then the packet payload is wrapped in the necessary packet headers and trailers. This packet overhead is at least 20 bytes for IP and 8 bytes for the User Datagram Protocol (UDP). Layer 2 protocols add even more overhead. Waiting longer to fill the IP datagram reduces overall overhead, which in turn reduces the true bandwidth needed to send the digitized voice. However, this waiting creates latency at the source, and too much total latency makes for a difficult conversation. Chart 1 shows the basic trade-off for initial latency versus true bandwidth.

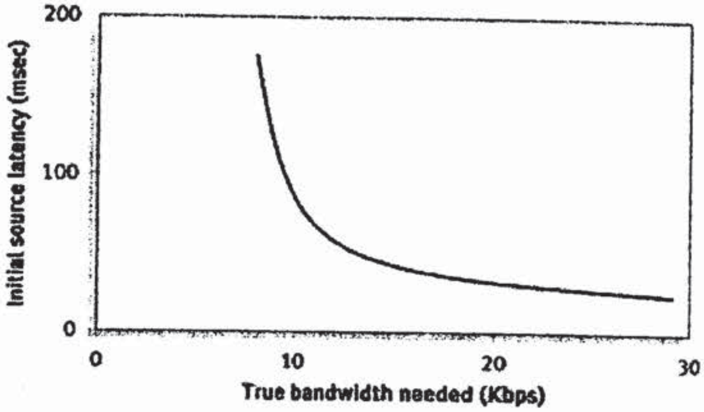


Chart 1. *Latency vs. Bandwidth Required for Voice-over-IP*

The total network latency and jitter (changes in the latency) have a degrading effect upon voice quality. Therefore, real-time voice quality is difficult to maintain over a large wide-area packet network without priority handling. As previously mentioned, VoIP converts standard telephone voice signals into compressed data packets that can be sent locally over Ethernet or globally via an ISP's data networks rather than traditional phone lines. One of the main difficulties with VoIP connections is that the communication network supporting a VoIP platform must be able to recognize that VoIP data packets contain voice signals, and be "smart" enough to know that the communication network has to move the data packets quickly.

Presently, serious voice traffic does not use the public Internet but runs on private IP-based global networks that can deliver voice data with minimal congestion. As such, transmission of voice signals over private data networks offers businesses some great advantages. For ISPs, merging voice and data on one single network allows them to expand their services beyond simple information access and into the realm of voice, fax, and virtual private networking. For businesses, the benefit is big savings on long-distance service. The Internet right now is a free medium on many networks. If businesses can send voice over a computer network, businesses can conceivably make long-distance or international calls for the cost of a local call. VoIP further facilitates electronic commerce by allowing a customer service rep using one data line to answer telephone questions while simultaneously placing a customer's order online, perusing the company's web site, browsing an online information/product database, or sending an E-mail. Similarly, VoIP also creates new possibilities for remote workers, who for the cost of a local call can log in remotely, retrieve voice mail from their laptop PCs, and keep their E-mail and web applications running while conducting multiple voice and data calls over one phone line. Presently, this type of expanded VoIP functionality is exclusively limited to those with access to private IP based networks, such as business users and not the typical household user.

In fact, most household computer users are generally limited to the congested public Internet and cannot implement the VoIP standard effectively. If latency and jitter are too high, or the cost of reducing them is excessive, one alternative is to buffer the CODEC data at the receiver. A large buffer can be filled irregularly but emptied at a uniform rate. This permits good quality reproduction of voice. Such a buffering technique is known as audio streaming, and it is a very practical approach for recorded voice or audio. Unfortunately, excessive buffering of the audio signals leads to generally unacceptable one-sided telephone conversations, where one party dominates the transmissions. What is needed is a packetized telephone system that is able to compensate for latency and jitter, without introducing noticeable buffering.

Traditionally, the operating environment for a household user with a VoIP connection is either a laptop or desktop general-purpose computer. The recording and transmission or interpretation of the VoIP packets takes place in the sound system or modem DSP found on the laptop or desktop. As such, the desktop system has a minor advantage over the laptop, because the desktop sound system traditionally provides stereo surround speakers and an accurate microphone. Thus, the desktop system can more accurately capture an individual's voice for retransmission of these voice signals to the user on the other end of the connection. VoIP telephone software buffering and control structures help improve the connection, but even though the audio signal has been accurately sampled, the processor delays and transmission latency associated with the desktop VoIP connection over the public Internet tends to result in a barely audible VoIP call. What is needed is a household compatible packetized telephone system that is able to compensate for communication network delays and hardware limitations, without introducing noticeable degradation into the voice signal.

One of the main difficulties with using VoIP in a household system is that the protocol requires the user to follow numerous steps in order to establish a voice connection. In addition to the normal boot-up process associated with general-purpose computers for the operating system and the Internet telephone application, there are several details difficult for the household user to provide. For example, if a user were trying to contact another individual, they would need to know the individual's IP address and punch the address into their software application or web browser to contact the individual. Once the user contacts the individual through either E-mail or at the website, the user must notify them that the user wishes to initiate a VoIP connection. Then the individual being contacted would enable their VoIP to allow the user to begin streaming voice packets between the two devices. What is needed is a simple method of using

DOCKET ALARM

Explore Litigation Insights

Docket Alarm provides insights to develop a more informed litigation strategy and the peace of mind of knowing you're on top of things.

Real-Time Litigation Alerts

Keep your litigation team up-to-date with **real-time alerts** and advanced team management tools built for the enterprise, all while greatly reducing PACER spend.

Our comprehensive service means we can handle Federal, State, and Administrative courts across the country.

Advanced Docket Research

With over 230 million records, Docket Alarm's cloud-native docket research platform finds what other services can't. Coverage includes Federal, State, plus PTAB, TTAB, ITC and NLRB decisions, all in one place.

Identify arguments that have been successful in the past with full text, pinpoint searching. Link to case law cited within any court document via Fastcase.

Analytics At Your Fingertips

Learn what happened the last time a particular judge, opposing counsel or company faced cases similar to yours.

Advanced out-of-the-box PTAB and TTAB analytics are always at your fingertips.

API

Docket Alarm offers a powerful API (application programming interface) to developers that want to integrate case filings into their apps.

LAW FIRMS

Build custom dashboards for your attorneys and clients with live data direct from the court.

Automate many repetitive legal tasks like conflict checks, document management, and marketing.

FINANCIAL INSTITUTIONS

Litigation and bankruptcy checks for companies and debtors.

E-DISCOVERY AND LEGAL VENDORS

Sync your system to PACER to automate legal marketing.

WHAT WILL YOU BUILD? | sales@docketalarm.com | 1-866-77-FASTCASE

fastcase®
Smarter legal research.