IP Telephony Terminal Solutions for Broadband Networks

Masami Mineo
Atsushi Niimura
Haruyasu Ooboshi
Masaaki Tanaka

OVERVIEW: The current trend toward the use of broadband networks through access services using xDSL (digital subscriber line), FTTH (fiber to the home), and CATV (cable TV) lines involves the widerspread availability of an "always-on" connection environment. As telecommunication service providers are beginning to provide IP (Internet protocol) telephony services using these network infrastructures, the Japanese Ministry of Public Management, Home Affairs, Posts and Telecommunications is developing a telephone-number plan for IP telephones to help establish an environment for the widespread use of IP telephony services. Targeting these markets, we have developed elemental technologies including: 1) communication protocol stacks, H.323 and SIP (session initiation protocol), 2) jitter control, which is a key technology for voice-over-packet transfer, 3) a real-time video communication technology, and 4) DSP (digital signal processor) middleware for telephony. We have applied these technologies to IP video phone software for PCs (personal computers), and developed a prototype of a "residential gateway" that can be connected to existing telephones to provide IP telephony services. We have been working on these elemental technologies to establish IP telephony terminal solutions in order to develop embedded IP telephone-related systems. Also, to enable the use of IPv6 (Internet protocol version 6) in the future, we have developed a prototype IPv6 version of the residential gateway that operates in the IPv6 environment.

INTRODUCTION

VOICE over Internet protocol (VoIP) is a technology for transferring voice over IP packets. VoIP was introduced in corporate networks to reduce communication costs by integrating voice and data. The current trend toward the use of broadband IP networks using xDSL, FTTH, and CATV lines increases the available bandwidth, improving the voice quality in VoIP and the availability of visual communication. This also makes an "always-on" connection environment widespread, creating an environment for the P2P (peer-to-peer) communication among homes.

Telecommunication service providers such as telecommunication carriers and ISPs (Internet service providers) are starting to provide IP telephony services on these network infrastructures[1]. The Japanese Ministry of Public Management, Home Affairs, Posts and Telecommunications created a study group on IP network technology to establish a number plan for IP telephony and quality guidelines on voice, transmission, and connectivity. This also helps build a network environment for the widespread use of IP telephony.

This report describes elemental technologies including communication protocols (H.323 and SIP), jitter control, real-time video communication, and voice and video codecs for terminal solutions in IP telephony and its products.

TRENDS IN IP TELEPHONY SERVICES PROVIDED BY TELECOMMUNICATION SERVICE PROVIDERS

A report published in February, 2002, by the study group on IP Network Technology of the Japanese Ministry of Public Management, Home Affairs, Posts and Telecommunications[2], defines IP telephony and the Internet telephone as follows:

IP telephony: Telephony service provided by using IP network technology over part or whole of a network.

Internet telephone: An IP telephone that uses the same IP network (the Internet) as the one used in

Internet applications such as the World Wide Web.

Since the Internet telephone is inexpensive and easy to use, Internet telephone services are mainly provided by ISPs to tie in their customers.

Broadband [ADSL (asymmetric digital subscriber line), FTTH, etc.] telecommunication service providers also plan to offer IP telephony services to entice customers to use their IP networks.

The above report states that the quality of an IP telephone is defined by the transmission quality of the network and the characteristics of the PC and IP telephony terminal itself. Therefore, IP telephony service providers are looking for PC applications and IP terminals that provide high-quality voice communication and try to improve the transmission quality of their networks.

Realizing that an IP telephony system requires (1) higher interoperability, and (2) high-quality voice and video transmission, we have developed a number of elemental technologies and established IP telephony terminal solutions.

ELEMENTAL TECHNOLOGIES COMPOSING AN IP TELEPHONY SYSTEM

Overview

An IP telephony system is composed of various elemental technologies such as communication protocols and digital voice processing.

Fig. 1 shows the configuration of an IP telephony system.

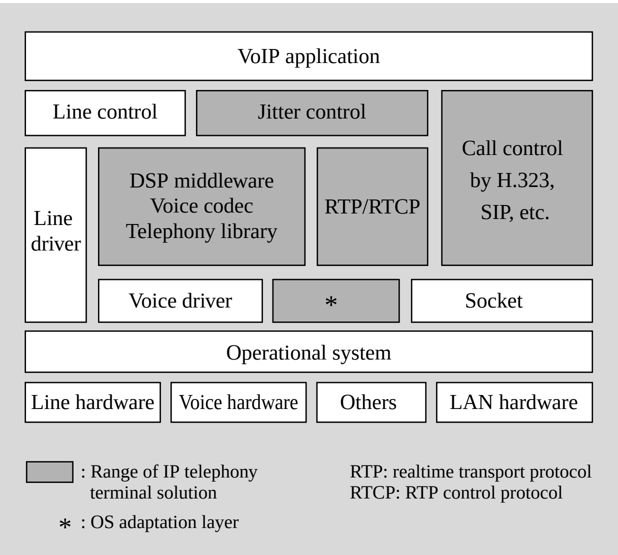

Fig. 1—IP Telephony System Configuration.
IP telephony system is composed by various components.

Communication Protocols

Communication protocols used in the IP telephony system include H.323, which is derived from the existing video telephone system, SIP, which has a strong affinity with other Internet protocols, MGCP (media gateway control protocol), which is used in CATV, and NOTASIP (nothing other than a simple Internet phone), developed in Japan.

To enable the interoperability of the system with products of other vendors, and its future development into a system for multimedia communication including video, we have developed H.323 and SIP protocol stacks as software products.

Even if IP telephony products are compliant with international standards, the interoperability among products from other vendors may not be ensured because of the difference in understanding product specifications and implementing products among different vendors.

A number of actions are being undertaken to solve these interoperability problems and enable high product interoperability. In Japan, the HATS (Harmonization of Advanced Telecommunication Systems) promotion conference is a typical organization for it.

We participated in all interoperability tests for H.323 and SIP performed by the HATS promotion conference, and have incorporated the results of these tests in our protocol stack products to enable users to easily build a system with high interoperability.

Jitter Control

The IP telephony system uses the user datagram protocol (UDP) for voice and video transmission to ensure real-time traffic. It provides a lower processing load but is also less reliable. The real-time transport protocol (RTP) is implemented on UDP.

With UDP, packets may be lost, and there is fluctuation (jitter) in packet arrival intervals due to traffic conditions over the IP network. Playing these packets at the time of arrival may result in packet dropout and image/sound degradation. To prevent this, an approach is being considered where a receive buffer is used to accumulate a certain number of packets before playing them to avoid degradation of voice and images. However, a large buffer causes delays in voice and images, making it difficult to communicate in real-time.

Jitter control is a technology to solve these problems. With jitter control, the buffer size is dynamically controlled depending on the network conditions, and voice and image data in dropout packets are interpolated. To control embedded systems, we have implemented a load-adaptive control method[3]), developed by Systems Development Laboratory of Hitachi, Ltd., as a library.

Fig. 2 shows the effect of this jitter control method.

Real-time Visual Communication Technology

We use MPEG-4 (moving picture experts group phase 4) as a video codec, which provides a high information compression rate and error-resistant encoding and is therefore suitable for real-time communication. It enables high-quality, real-time, low-delay video communication at a bit rate of 32 kbit/s to 2 Mbit/s depending on the network bandwidth. This is a software codec developed by Central Research Laboratory of Hitachi, Ltd.

DSP Library

As a platform for embedded systems, we use SH3-DSP, a RISC microcomputer for multimedia processing. As general-purpose voice DSP middleware including voice codecs (G.723.1 and G.729A) and echo cancellers (G.165 and G.167), we use products developed by Semiconductor & Integrated Circuits, Hitachi, Ltd.[4]), and we have also developed DSP middleware products specifically designed for telephony, including DTMF (dual tone multi frequency) and CID (caller identification).

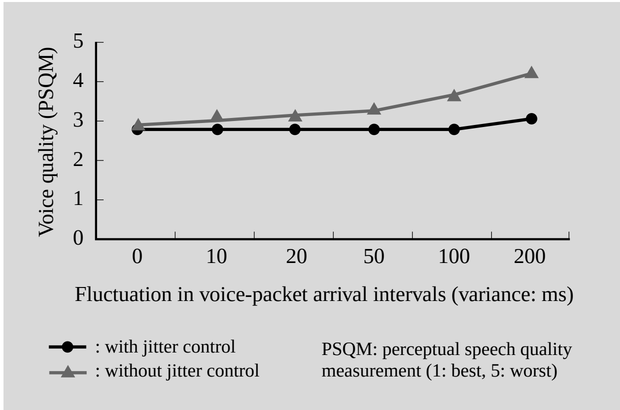

Fig. 2—The Effect of Jitter Control.
Jitter control can effectively improve voice quality under heavy network conditions.

Fig. 3—Network Communication Software.
Cell phone-like interface provides user-friendly operation.

APPLICATION EXAMPLE

Network Communication Software

Our network communication software enables real-time voice and video communication over IP networks. The widespread use of broadband networks, higher PC performance, and lower PC cost have opened up possibilities to provide videophone functionality at a lower price. This software can be used by a variety of professional organizations including educational institutions, security companies, and organizations using remote monitoring, as well as by local communities and recreational facilities.

The network communication software has the following features:

(1) High-quality voice-call capability

It provides high-quality voice transmission by using our unique jitter control technology.

(2) High-quality video-communication capability

It enables high-quality, real-time video communication depending on the network bandwidth by using our unique real-time video communication technology for MPEG-4.

(3) High interoperability

It uses H.323 as a call control protocol to ensure the interoperability of the IP telephony system with gateway products to enable connecting ordinary telephones of other vendors to the system.

(4) Easy-to-use

The software provides a cell phone-like user-friendly interface and such functions as an address book and call history.

Fig. 3 shows a screen image of the network communication software.

Residential Gateway Prototype

To demonstrate the application example to an embedded IP telephone system, we developed a

TABLE 1. Specifications of Residential Gateway Prototype
This device supports two call control protocols, which makes it possible to be used in various network environments.

	Item	Description
1	CPU	SH3-DSP (SH7729R) 200 MHz
2	OS	VxWorks*
3	Call control protocol	H.323 (V2), SIP
4	Voice codec	G.711 (A-law, μ-law), G.729A, G.723.1
5	LAN interface	100Base-TX/10Base-T RJ-45
6	Line interface	Two-wire analog telephone interface RJ-11 × 2
7	Analog line interface	Two-wire analog central office line interface RJ-11 × 1

* VxWorks is a registered trademark of Wind River Systems, Inc.

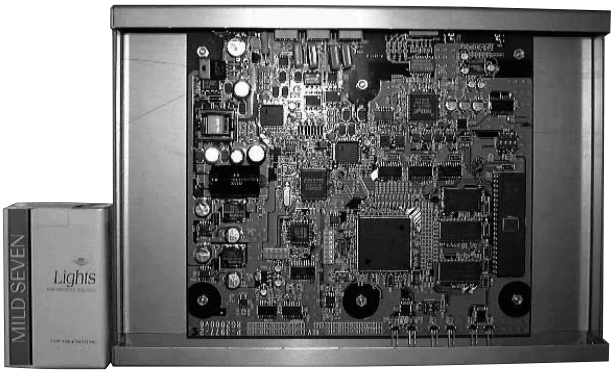

Fig. 4—Residential Gateway Prototype.
Hitachi RISC (reduced instruction set computer) microcomputer SuperH3-DSP (SH3-DSP) simplifies the circuitry.

prototype of a residential gateway that can be connected to ordinary telephones to provide IP telephony services. This device provides IP telephony services to homes through telecommunication service providers.

Table 1 shows the specifications of this system.

Fig. 4 shows an external view of the prototype hardware.

We will promote our IP telephony terminal solutions using this hardware platform.

Support for IPv6

The problem of IP address depletion in IPv4 (Internet protocol version 4) networks is particularly critical for P2P communication systems like IP telephony. To solve this problem, we developed a prototype of a VoIP telephone adapter that supports IPv6 on the above hardware. We tested this adapter in a demonstration experiment of an IPv6 access network and information home electrical products conducted by IPv6 Promotion Council.

We plan to improve its quality and functionality based on the QoS (quality of service) control and security functions that are features of IPv6.

CONCLUSIONS

In this report, we described elemental technologies and products that compose our IP telephony terminal solutions, and showed some examples of application products.

The IP telephony system is expected to further evolve and become more widely available. To achieve this, we need to improve the performance of the system by linking its hardware and middleware, reducing its costs, and improve its functionality using the functional extendibility of software.

We will continue to make an effort to develop IP telephony systems with higher quality and performance by combining our communication software technology with Hitachi's middleware and hardware technology.

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ABOUT THE AUTHORS

Masami Mineo

Joined Hitachi Communication Systems, Inc. in 1979, and now works at the VoIP Solution department of the IT Solution Center. He is currently engaged in the research and development of VoIP-related systems. Mr. Mineo is a member of Information Processing Society of Japan and Japan Society for Software Science and Technology, and can be reached by e-mail at masami-mineo@hitachi-cs.co.jp.

Atsushi Niimura

Joined Hitachi Communication Systems, Inc. in 1980, and now works at the IP Solution Department of the IT Solution Center. He is currently engaged in the development of Internet communication software. Mr. Niimura is a member of IEICE, and can be reached by e-mail at atsushi-niimura@hitachi-cs.co.jp.

Haruyasu Ooboshi

Joined Hitachi Communication Systems, Inc. in 1984, and now works at the VoIP Solution department of the IT Solution Center. He is currently engaged in the development of VoIP system products. Mr. Ooboshi can be reached by e-mail at haruyasu-ooboshi@hitachi-cs.co.jp.

Masaaki Tanaka

Joined Hitachi Communication Systems, Inc. in 1979, and now works at the VoIP Solution department of the IT Solution Center. He is currently engaged in the development and design of VoIP hardware. Mr. Tanaka can be reached by e-mail at masaaki-tanaka@hitachi-cs.co.jp.