IP-Telephony Technology and Solution Families Covering One Box Startup to Carrier Grade Service

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ABSTRACT In recent years, IP telephony services using VoIP (Voice over IP) technology has been attracting a great deal of attention all over the world as a means of telecommunication cost reduction. This paper describes the IP telephony market trend and various important technologies for not only IP telephony for public service and enterprise use but also IP Centrex services, a new solution for voice communications aimed at the needs of business customers, just launched last autumn in Japan. It covers packetization, voice codecs, echo cancellation and call processing protocols. Also, it is suggested that the possibility of convergence of information and network technologies will bring a new work style of business.

KEYWORDS **VoIP (Voice over IP), IP telephony, SIP (Session Initiation Protocol), IP Centrex service**

1. INTRODUCTION

IP telephony services using VoIP (Voice over IP) technology is attracting a great deal of attention. The number of IP telephony subscribers in Japan is projected to reach 4.4 million in March 2004[1]. One characteristic of IP telephony services for residential users is their low calling rates. Calls between subscribers belonging to the same ITSP (Internet Telephony Service Provider) in particular are free of charge. Since the autumn of 2003, IP Centrex services for businesses have also been available that use V-LAN (Virtual Local Area Network) lines as access lines. Meanwhile, the Ministry of Public Management, Home Affairs, Posts and Telecommunications laid out a dedicated numbering scheme for IP telephony in 2002. Telecom carriers are already using this numbering scheme.

At present, however, these services have several problems — notably, subscribers having only IP telephony dedicated numbers cannot make emergency calls to the police or fire department nor can they access some well-known PSTN (Public Switched Telephone Network) services such as free phone service (0800 service). These problems were already experienced with the proliferation of mobile phones. It is expected that the problems will be solved gradually as changes are made to meet the needs of society.

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2. VoIP MARKET TREND

The VoIP market for businesses in Japan breaks down roughly into two segments — outside line IP telephony and premises IP telephony. The market for outside line IP telephony is further divided into two categories of services. One is IP trunk telephony service that deploys IP for the trunk circuits of existing telephone networks. The other is IP telephony service whereby IP networks are accessed directly from within enterprise users' premises.

Of the companies subscribing to outside line IP telephony services, 8,000 were using the IP telephony service and 131,000 the IP trunk telephony service in 2002. It is expected that in 2005, 170,000 and 123,000 businesses will be using the IP telephony service and IP trunk telephony service, respectively[2].

With respect to the premises IP telephony market, the number of companies using IP-PBX is projected to expand to 19,000 in 2005 from 9,000 in 2002. It is also expected that, while no businesses deployed IP-Centrex in 2002, 5,000 companies will be using the service in 2005. There is an evident tendency for the growth of the VoIP market to accelerate in all its segments.

One powerful factor prompting users to deploy VoIP is their need for cost reduction. On the other hand, users who are now considering the use of VoIP have high expectations particularly about future service enhancement.

Office work involves many different kinds of information, often in an inconsistent way, in many different forms such as calls, telephone directories, message slips, whereabouts bulletin boards/schedule charts, electronic mail, fax, individual documents and shared documents. Furthermore, these different kinds of information are used for collaboration between divisions (or even with affiliated companies) and the processing and integration of data is accomplished based on individual sets of such information.

There is an increasing need to improve productivity at the office by linking these distinct parts of information with adequate aspects of work.

The following two examples explain how this can be achieved.

1) Combine the “telephone directory” and “calls” while at the same time linking them with the personnel database. This not only eliminates the need for the “paper-based telephone directory” but also realizes an “electronic telephone directory.” The electronic telephone directory can be updated in real time in response to changes and allows one to communicate with the other-end party just by clicking a telephone number of mail address displayed in a Web page on the PC. Furthermore, a “call routing function” can be implemented which automatically redirects the call to a voice mail system or mobile phone by identifying the called party's status (presence information indicating that the called party is temporarily absent, out, in a meeting, busy on another call, has gone home, etc.).

2) Deploy a Web-based multipoint video conferencing function. Unlike its counterpart over telephone lines, the function enables one to grasp who is participating in the conference at a glance. If multiple workers can collaborate while viewing the same shared document or drawing, benefits such as savings in travel time and a faster decision-making process will be achieved.

3. BASIC TECHNOLOGIES FOR IP TELEPHONY

In addition to the known basic technologies such as the packetization technique that segments voice signals into packets and voice codecs, this section describes the key technologies needed to maintain levels of voice quality that are equivalent to those provided on the PSTN when sending voice over the IP network.

3.1 Voice Processing

There is a growing demand among users for VoIP to achieve a voice quality level comparable to that offered on the conventional PSTN. The deployment of IP has negative effects on some factors such as voice delays and echoes. The use of the latest technologies, however, has improved the voice quality of VoIP to a level almost equal to that of the PSTN, thus letting users engage in voice conversations without noticing the negative effects of IP. Comparing VoIP with the PSTN, the following elements are considered to be vital factors in determining voice quality. How to address these factors is the key to providing high voice quality with VoIP.

- Voice delay
- Jitter buffer control (absorption of IP packet jitter)
- QoS control in the network
- Bandwidth guarantee in the network
- Echo control

Figure 1 shows a simple block diagram illustrating how these factors are addressed, and explanations of the factors are given below.

(1) Voice Delay (Jitter Buffer Control)

Delay can occur when voice signals are segmented into IP packets, when they are compressed, and when

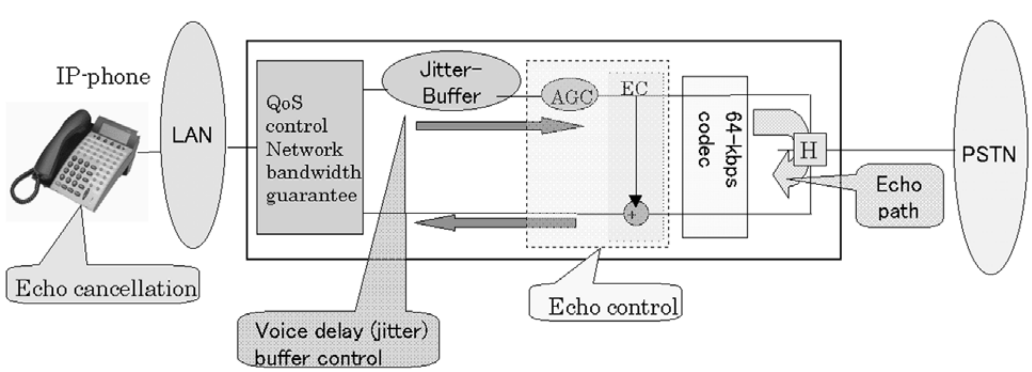

Fig. 1 Allocation of voice processing technologies.

they go through the jitter buffer. If the length of delay exceeds a certain level, the conversation is crippled, making it impossible for users to continue talking in the normal way. One important factor in controlling the delay is jitter buffer control. The jitter buffer is intended to absorb fluctuations in intervals at which packets arrive at the network, thus ensuring that the conversation proceeds without any sound losses. Packet arrival intervals, however, fluctuate from moment to moment. Whereas providing a large jitter buffer size reduces the probability of sound losses to almost zero, it causes the delay to become greater. Therefore, it is technically important to minimize the jitter buffer size based on the network condition. To meet this challenge, packet arrival times and fluctuations are evaluated using statistical techniques and packets are processed based on information such as an abrupt change in the packet jitter caused by a sudden increase in the network load. The size of the jitter buffer is adjusted so as to minimize packet losses in the network and to make delay as small as possible while preventing sound losses in conversations. Furthermore, since clock differences between terminals are inevitable in an IP network, the jitter buffer is required to have a mechanism for compensating for those differences. To this end, the jitter buffer is also designed to detect the slightest difference in the packet transmission intervals and to compensate for the difference through the use of a statistical technique.

(2) QoS Control in the Network

Priority control (QoS) over voice packets is exerted in the network to prevent voice quality from degrading. To achieve QoS, Layer 2 VLAN (IEEE 802.1q) and Layer 3 IP Precedence and DiffServ are supported. These capabilities prevent voice quality degradation not only for communications within subnets but also for those across wide-area networks.

(3) Bandwidth Guarantee in the Network

Suppose that calls are set up between two networks. If more calls than expected originate, the resulting traffic exceeds the bandwidth allotted to the networks, potentially leading to sound losses and other phenomena associated with voice quality degradation. To avoid this situation, the amount of voice packet traffic being carried between the networks (the bandwidth currently in use) is monitored and, if the traffic exceeds a preset threshold, further connections are restricted to prevent voice quality from degrading.

(4) Echo control

Since voice delay is intrinsic to VoIP, echoes occur when signals are looped at the 2-wire/4-wire conversion circuit of the legacy interface or at the receiver of the IP telephone set. Such echoes are canceled by the G.168 canceller (line echo canceller) mounted in the gateway and the G.167 canceller (acoustic echo canceller meant to cancel echoes at the receiver) contained in the IP telephone set. There may be some echo, however, in the initial period of the call during which the echo canceller is still not trained. To prevent such echoes in the initial period of the call, a function is provided that sends a unique echo training tone at the beginning of the call. Also, in the case of a gateway connected to multiple legacy lines of different types, the line of the same type as the one that the echo canceller used most recently is given precedence in line seizure, in order to avoid echoes occurring in the initial period of the call. In addition, to make full use of the echo canceller's functionality, a function is also supported that adjusts the level of the signal that is input to the echo canceller without changing the voice level.

3.2 SIP Based VoIP Signaling Network

The call processing protocols for VoIP include: H.323[3], a standard that the ITU-T developed and promoted ahead of other organizations; MGCP (Media Gateway Control Protocol)[7]/MEGACO (H.248)[4], which is a master/slave protocol mainly used for gateway control; and the IETF-developed SIP (Session Initiation Protocol)[5] which is currently the mainstream protocol for VoIP services and is expected to evolve further in the future.

SIP was developed as a means to set up sessions between applications for the purpose of implementing multimedia communications over the Internet. It was published as RFC 2543 in 1999 and renumbered RFC 3261 when it was revised in June 2002. SIP is a text-based protocol built, referring to such protocols as HTTP (Hypertext Transfer Protocol)[8] and SMTP (Simple Mail Transfer Protocol)[9]. For this reason, many parts of the descriptions of the SIP functions and messages are common to those used for the Web and e-mail, resulting in SIP inheriting the advantages of these applications.

In order for SIP terminals to accomplish voice communication, they exchange their media information and set up a session by exchanging the following SIP messages - INVITE, 200 OK and ACK (see **Fig. 2**). During this process, the SIP proxy server located in the call center relays these SIP messages to their respective destinations. While the session is active, the SIP terminals engage in peer-to-peer voice communication using the RTP (Real-time Transfer Protocol)[6] or other adequate protocol. To disconnect the call, they tear down the session by exchanging the BYE and 200 OK messages. When compared to other call processing protocols, the call setup, communication and tear down procedures are simple as described above, which is one of the reasons why SIP is widely used for VoIP.

Because SIP is a highly extensible protocol that is remarkably compatible with the Internet, an array of its extensions are being developed by the IETF. It also has been adopted by 3GPP, a standardization body working on the third generation mobile specifications, as a protocol for implementing multimedia communications. As suggested by these and other facts, expectations are running high that SIP will be instrumental in connecting various kinds of things and data and thus implementing ubiquitous network services, including video conferencing, Net-enabled consumer electronics, network gaming and advertisement delivery, for the coming all-IP networking age.

3.3 IP Telephony Service Networks

Even after IP telephony equipment has been deployed that takes full advantage of the cutting-edge technologies explained above, there still remain some issues to be considered in relation to networks. Since telephone systems are very important core systems for enterprise users, thorough security measures need to be taken when introducing an IP telephony system. Another prerequisite for deployment of such a system is a flexible system configuration that can be changed to meet the user's specific needs. For example, the user may want to leave part of the conventional lines intact in the new IP telephony system configuration in order to secure some center-powered telephone terminals in protection against power failure or to continue using those lines for which IP deployment is difficult for cost reasons (e.g., local dedicated circuits, long-distance extension lines, etc.). It is necessary to accommodate these needs.

As factors that impede businesses from introducing IP telephony systems, many cite the problems with voice quality and reliability. Not a few enterprise users feel that deploying IP for their entire telephony systems would be a risk. Also, many of the telephone uses in a corporate environment presuppose the advanced telephony capabilities offered by existing PBX systems, and they are closely tied to the work styles unique to individual workplaces. A system configuration is therefore needed that allows enterprise users to migrate to an IP telephony system smoothly while maintaining the telephony environment they currently enjoy.

There are two ways to provide IP telephony system

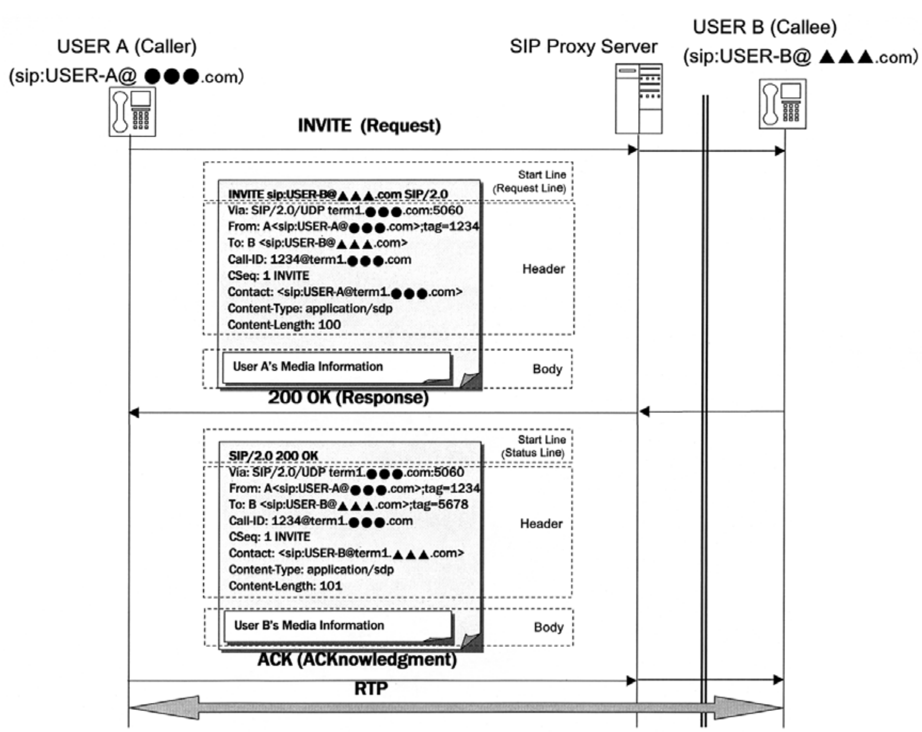

Fig. 2 SIP call flow.

services to enterprise users. One is by installing a voice server individually at each site of business activity, and the other is by installing voice servers at a central location and managing IP telephones at multiple sites in a centralized manner (Centrex type).

When installing a voice server individually at each business activity site, it is necessary to equip the voice server with an additional device capable of accommodating existing lines so as to ensure that some traditional center-powered telephone terminals can be used together with IP phones in protection against power failure, which is one of the weaknesses of IP telephony, as mentioned earlier. In that case, of course, communication between IP phones and traditional phones should be possible and the same level of telephony service should normally be available for both types of phones.

When voice servers are installed at a central location, several worst-case scenarios can be expected in such cases as a failure of any of the voice servers installed at the center, tear down of sessions due to some trouble on the IP network or a router fault in a site, leading to the outage of the entire IP telephony system or the disruption of all voice communications within the site where the problem has occurred. Measures to respond to these situations need to be formulated. Two concrete patterns of response are detailed below.

(1) Pattern 1

Under normal circumstances, the media gateway (which inter-works with the fixed public network) installed at each site is used for call origination and termination between the site and the fixed public network. In an emergency, it allows one or more predetermined telephones to be connected directly to the fixed public network, thereby enabling call origination and termination between at least those telephones and the fixed public network.

(2) Pattern 2

Emergency call processing devices are installed at individual sites on an as-needed basis. These devices conduct a health check at regular intervals with voice servers that are usually located in the center. All IP telephone terminals of a site operate under the control of a voice server in the center. If the emergency call processing device of a site detects the disconnection of communication with the center server, the control over the IP telephone terminals of that site is automatically transferred to the device. Then, the emergency call processing device recovers most of the daily telephony services, such as call origination to and call termination from the fixed public network via the media gateway, intercom, call forwarding and multiline telephones, thus preventing the telephony system from getting out of service. Of course, upon recovery of the communication with the voice server in the center, the control over the IP telephone terminals of the site is automatically returned from the emergency call processing device to the voice server, putting the system back in the normal operating condition. The merit of this approach is that, by preparing different types of emergency call processing devices for different scales of individual sites, the system can be deployed at lower cost than when procuring an IP-PBX or main system for each site individually.

4. ENTERPRISE APPLICATIONS

Table I lists NEC's IP telephony products. Residential IP telephony services are expected to approach the PSTN service offerings in terms of both quality and content. On the other hand, efforts are underway to explore new enterprise IP telephony services. The following sections discuss the available enterprise applications in detail.

4.1 IP-PBX

The communications environment surrounding the business community is changing rapidly.

Particularly, as the IP networks are being developed as social infrastructure with the rapid spread of the Internet in recent years, it is increasingly necessary for business communications to achieve productivity improvement and TCO reduction that are enabled by the IP technology. As the volume of IP data traffic carried over the network is growing at a phenomenal pace, an array of WAN solutions are emerging including VoIP gateways and routers that realize

Table I NEC's IP telephony product overview.

Use	Series name	Product
Residential service	CX	Call-agent, SIP server, Signaling GW, Media GW
IP-PBX	APEX/NEAX	7600i, 3600i, SV7000, soft-phone etc.
IP Centrex	IParty	SIP server, VoIP NAT, GWs, soft-phone etc.
IT-NW convergence	iExpress5800	(All in one box type)

low-cost communications by delivering voice signals as IP packets. Businesses are actively introducing these products as a means to trim their communications costs. This trend extends so far as to reach the floor of the workplace, with an increasing number of companies considering the deployment of IP telephony for their LAN solutions, i.e. using LANs for internal telephone facilities.

The concept of IP-PBX, which is aimed at integrating data infrastructure with PBX capabilities, is derived from two distinct platforms - PBX systems and data devices. SV7000, developed by NEC, implements the diverse services of the conventional PBX system on a soft switch. It is a server-type product that encompasses the entire suite of the traditional digital PBX capabilities while at the same time making it possible to provide new services for IP phones and IP networks. Without sacrificing the high level of telephony service offered by the existing APEX series systems, SV7000 allows accommodation and coexistence of not only conventional telephone terminals, including PHS handsets, and multifunction IP phones but also SIP single-function phones, SIP multifunction phones and wireless terminals.

Especially for the SIP multifunction terminals, we have originally expanded the SIP protocol to inherit the traditional PBX features to be used on SIP. With this, various features provided for ordinary multifunction phones will be supported while at the same time making it possible to provide the collaboration of opened application such as presence feature and instant messaging feature based on SIP protocol.

Furthermore, it is foreseen that PC based soft phones widely spread. For traditional phone users, however, simply installing the telephone functions in PCs incurs inconveniences, for example:

· Significantly differed operation method
· Available only when PC is operating
· Difficulty in carrying

To solve them, our own technology enabled the integration of the PC based soft phone and the hard phone including the wireless terminal. With this technology, users can choose their preferable terminals, for example, PHS (Personal Handy-phone System) terminals or wireless LAN telephone, in place of USB hand-sets connected with PCs, still enjoying the use of the electronic telephone directory.

SV7000 has a rich set of maintenance functions as well, including voice quality management, SNMP capability, FTP-based program downloading and provision of Web-based maintenance consoles. Featuring these functions, the system enables businesses to implement TCO reduction solutions.

Its telephony integrated capability based on CTI (Computer Telephony Integration) provides very useful telephony applications even in a fully IP-enabled environment, by accomplishing higher-level interworking with the IP integrated infrastructure. This makes it possible for SV7000 to contribute to improving productivity of businesses.

4.2 IP-Centrex

IP-Centrex (IP-CTX) is a Centrex service that uses the VoIP technology. Generally, Centrex provides an extension telephone service for corporate users through the use of central office switches on the public network. It allows companies to outsource the capabilities of PBX and key systems they install at their business sites as well as the work of operating and maintaining such systems, thereby enabling them to cut the TCO of internal communications. With the traditional Centrex service that does not use the VoIP technology, call signals are sent back and forth via the Centrex system located at the central office on the public network, even if the call is intended for internal communication between sites of a company. This not only raises security concerns but also poses other problems — e.g., a network spanning multiple sites tends to be expensive and difficult to build.

IP-CTX delivers voice signals in the form of VoIP packets and accomplishes communication processing by means of high-performance servers, thus offering services and cost-effectiveness that cannot be achieved with the conventional Centrex system. With IP-CTX, terminals communicate with each other on a peer-to-peer basis. Since communication inside a site is closed within the internal network of that site, IP-CTX outperforms the conventional Centrex service in security as well. Also, by linking sites on the IP network, a wide area network can be established less expensively than with the traditional approach. Thanks to these advantages, IP-CTX is now receiving attention as one of the most promising solutions for future corporate communications.

There are several powerful protocols deemed vital to implement VoIP. SIP is the most promising candidate protocol.

The NEC IParty series offers IP-CTX products that use the SIP provided by NEC. **Figure 3** shows an example of the IP-CTX architecture implemented using the IParty series products. The IParty solution provides the following capabilities:

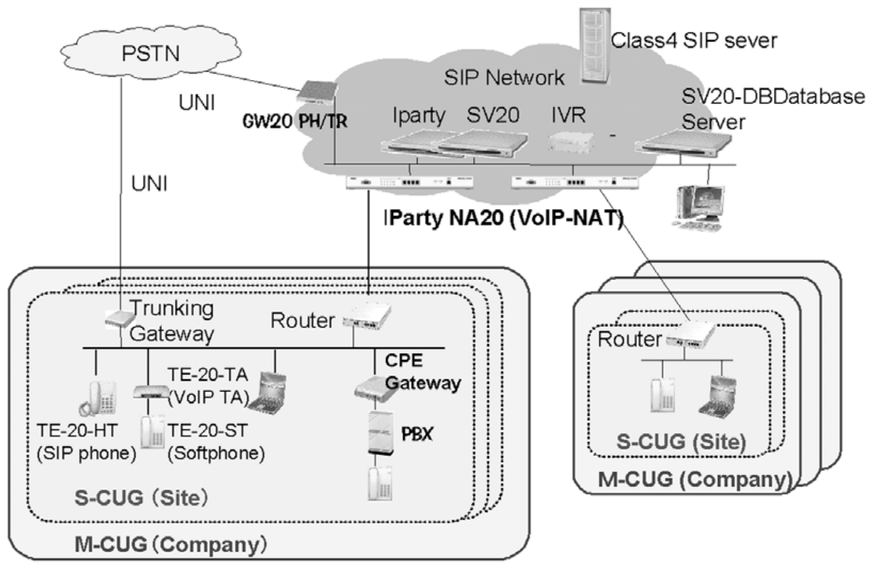

Fig. 3 IParty architecture.

(1) Provision of Shared Service

IParty enables communication among multiple companies or multiple sites of a company to be handled by a single server.

A single IParty SV20 SIP server is capable of serving up to 128 companies independently and accommodating telephone lines for up to 1,024 sites of each of those companies. The IParty NA20 VoIP-NAT allows each company and site to have an independent IP address (private IP address).

(2) Coexistence of the Public Network and Intranets

IParty permits a network based on the public telephone numbering schemes and a corporate dedicated-line intranet to overlap in a single network architecture. The numbering schemes of the public networks (8-digit numbers prefixed with 050 for the public IP networks in Japan and 10-digit numbers prefixed with 0 for the PSTN) can be used along with individual companies' unique numbering schemes within the same network.

(3) Accommodation of Existing Networks and Devices by Gateways

A variety of gateways are available to allow the IP-CTX system to accommodate different types of existing public networks and existing PBX and key systems. The trunking gateway can connect emergency service calls (in Japan, those made by dialing such a number as 110 or 119), which currently cannot be set up via the IP network, and route local calls to the public network point closest to the site. In cases where the user wishes to build a network by linking the sites that are operating independently while keeping the existing PBX capabilities intact, the CPE (Customer Premises Equipment) gateway can accommodate the PBX or key systems.

(4) Centralized Provisioning Management

The SV20-DB server enables system configuration settings, subscriber profiles and various other information to be managed centrally on a per-company and per-site basis. This makes it possible to deploy IP-CTX for a large company that otherwise cannot be accommodated by a single SIP server.

4.3 Web Applications Related to IP Telephony

NEC's iExpress5800 is a next-generation server system that combines IT and network technology to give anyone secure access to data, voice, video and other types of information anywhere, regardless of the communication environment. While maintaining the traditional PBX services using the SIP server and VoIP gateway, iExpress5800 features the roaming server and Radius server to provide secure communication services. Other components comprising the system include devices intended to offer added value, such as the conference server and integrated operation and management device, and SIP phones and softphones (See **Fig. 4**).

The features of iExpress5800 are explained below.

(1) VoIP Function

Since SIP (Session Initiation Protocol) is adopted for call control, iExpress5800 allows applications to be extended with ease and provides services that

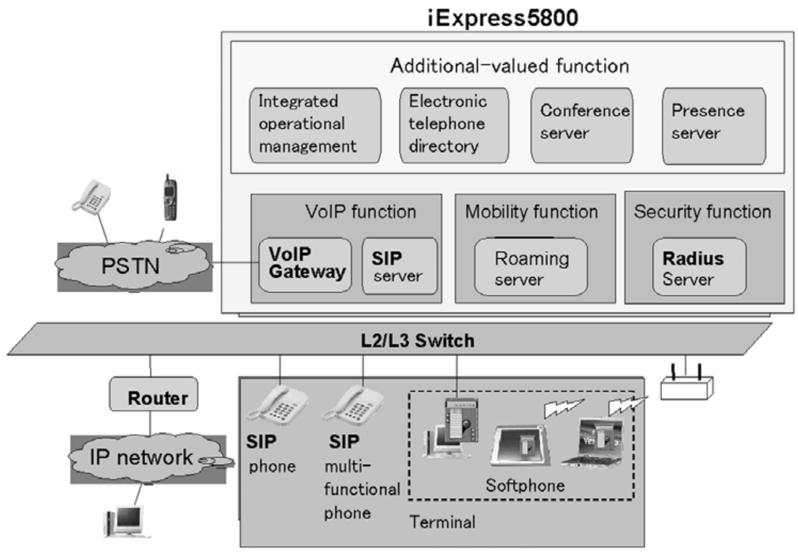

Fig. 4 iExpress5800 architecture.

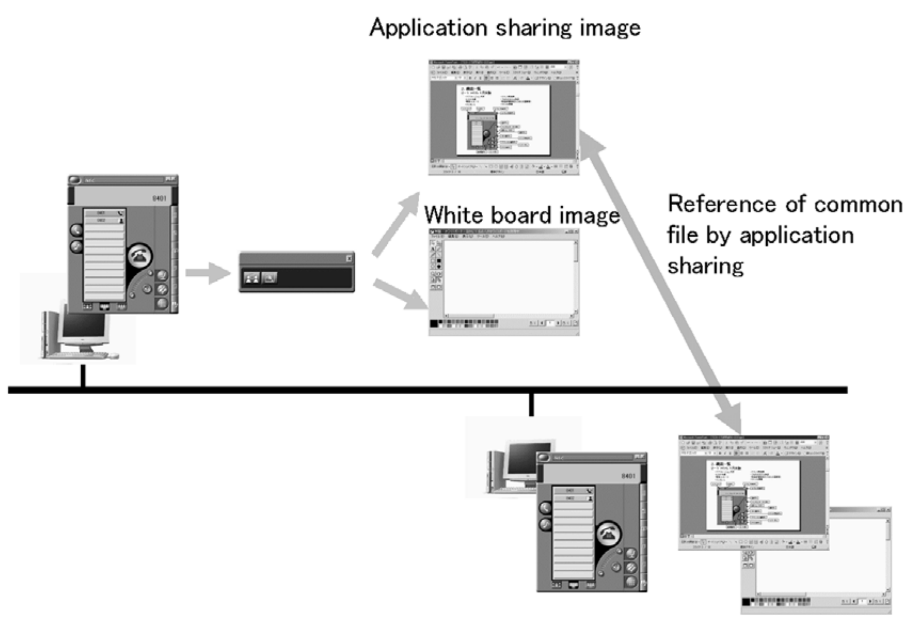

Fig. 5 Application sharing.

inter-work with multimedia applications, such as the electronic telephone directory and conference server, in addition to the voice telephony service (See **Fig. 5**).

If softphone terminals are used, such features as application sharing and video conferencing become available, making the communication environment more convenient. Furthermore, by inter-working with a presence server, the VoIP function can display the softphone status (busy, absent, online, etc.), making it possible for the user to choose the most suitable means of communication (telephone, short message, e-mail, etc.).

(2) Mobility Function

The mobility function of iExpress5800 provides the following services by using the technologies supported by the roaming server, such as mobile IP, IPSec, NAT (Network Address Translation)/NAPT (Network Address Port Translation) and traversal.

(3) Seamless Roaming Service

The mobile IP technology enables seamless roaming by automatically switching between networks such as PHS, mobile service (2G/3G), wired and wireless LANs.

(4) Mobile VPN Service

IPSec provides security when the user accesses the intranet while at home or outside the company.

(5) Security Function

iExpress5800 features the IEEE802.1x-compliant RADIUS authentication server function, which guarantees a level of security that is high enough for core corporate tasks even in a wireless LAN environment.

5. SOCIAL IMPACT AND FUTURE PROSPECT

Background factors surrounding corporate users deploying IP telephony systems include the following:

(1) Drastic Change in User Needs

With the traditional telephony service, the biggest issue has been how to cut the communications cost. The increasingly severe economic climate, however, has brought about a drastic change in user needs for communications. Users are now seeking not only to reduce communications cost but also to improve their cash flow drastically on an across-the-board basis by slashing personnel costs and other expenses incurred in operation and planning and avoiding deploying equipment wherever possible.

(2) Focus on Core Tasks

Management consider it increasingly important to outsource the work involved in the conventional telephony service and to relieve employees of the burdensome job of operating the corporate network, thus letting them focus on the company's core tasks.

(3) Provision of Cutting-Edge Network Solutions

With technology advancing rapidly in the broadband age and carriers releasing new services one after another, users want to make full use of such innovations as useful business tools.

As stated above, the key point will be how to best use the features of IP as support tools not only for achieving communications and operational cost reductions but also for reforming the work style upon which the company's survival hinges. In this context, the "establishment of and improvement on highly flexible, high-capacity, high-speed and highly reliable IP infrastructure" and "applications" that run on such infrastructure occupy an important position. It is likely that the infrastructure and applications will continue to grow in importance and need to support a highly open environment. Given this trend, it can be considered that demands for servers and other peripheral equipment other than the infrastructure, plus potential needs for services for outsourcing the work of operating such equipment, will create a huge business opportunity and market. NEC, too, intends to focus on this field of business.

6. CONCLUSION

IP telephony is already in practical use and is expected to grow as a new means of communication for both public and corporate uses. With the basic technology already established, efforts will be focused on implementing network services that are easier to use. On the other hand, much hope is being placed on the development of IT-integrated applications that streamline work and improve work flows. It is NEC's intention to increase the focus on boosting its product series to meet these needs.

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