Corporate Network System Using VoIP Technology

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OVERVIEW: The rapidly growing pervasiveness of the Internet is dramatically increasing the amount of data conveyed through networks, including corporate networks. To reduce communication costs, the need for voice over Internet protocol (VoIP) technology is growing because it integrates voice and data for efficient transmission over IP networks. Hitachi has developed VoIP gateways for wide area networks (WANs). The NT-1000 gateway supports Common Channel Signaling System No. 7 (SS7) and the NT-40 supports the H.323 version 2 protocol. These gateways provide optimal corporate network solutions for small and large networks, solutions that meet the users' needs. Hitachi used the NT-1000 gateway for rebuilding and expanding its own corporate network. From this expansion, we have acquired the know-how needed to build large-scale VoIP networks, and we are making use of this know-how to develop VoIP solutions for WANs.

INTRODUCTION

THE rapidly growing pervasiveness of the Internet is dramatically increasing the amount of data conveyed through networks, and the volume of data traffic will soon surpass that of voice traffic. To cope with this situation, communication carriers and other related companies are establishing backbone networks that can handle faster and more efficient transmission of Internet protocol (IP) data. This trend is also penetrating into corporate networks. A significant challenge for companies in dealing with this increasing amount of data is rebuilding their networks to enable efficient transmission of voice and data. There is thus an increasing the need for voice over IP (VoIP) technology, which integrates voice and data.

In April 2000, Hitachi released a VoIP-supporting version of its main force CX8000 series private branch exchange (PBX) and the NT-1000 VoIP gateway, which can be connected to PBXs other than those in the CX8000 series. Hitachi is now planning to build VoIP networks on a large scale.

In this paper we describe the features and functions of the NT-1000 gateway and the CX8000 PBX with IP trunk. The features and functions of both these products can be used together to create a corporate network system using VoIP technology. We also describe Hitachi's own corporate network as an example of a large-scale network using VoIP technology.

DEVELOPMENT OF VOIP NETWORK FOR WANS AND HITACHI'S APPROACH
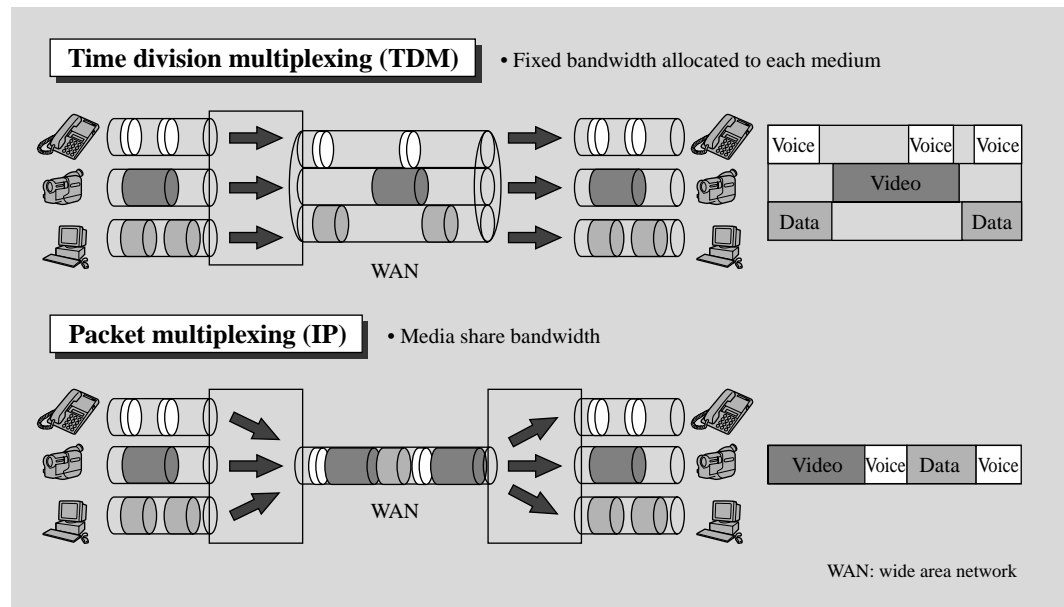
VoIP Development

With the diversification and lower prices of the line services provided by communication carriers and the changes in the business environments of companies, conventional corporate voice and data networks are being integrated. Time division multiplexer (TDM) is being used to make operations more effective, so as to reduce running costs and increase efficiency. IP-based networks are being built based on the background of the rapidly expanding use of the Internet and the diversification and lower prices of services. For example, communication carriers are starting to offer IP virtual private networks (IP-VPNs)[1, 2].

VoIP Technology

As shown in Fig. 1, in the TDM system, a fixed bandwidth is allocated to each medium, and it is reserved for each medium even when it is not being used. In the IP system, on the other hand, bandwidth is shared, thereby improving usage efficiency and reducing communication costs. For example, all the bandwidth can be used for media other than voice when there are no voice communications.

However, an IP network is a best-efforts one, so it can be a problem to handle voice signals that require real-time processing. Measures are needed to prevent situations that may degrade voice quality, such as delay,

Fig. 1—Comparison of TDM and IP Networks.
In the packet multiplexing of IP networks, the bandwidth is shared by the media, thereby improving line usage efficiency and reducing communication costs.

jitter, and packet loss, which are not problems in conventional TDM networks.

With VoIP technology, the router can prevent voice packets from being delayed or discarded. The VoIP gateway unit can absorb jitter, i.e., variations in the packet arrival times. Its echo canceller can remove undesired echoes caused by delay.

Hitachi's Approach

Hitachi released the GI-IP100 VoIP gateway, a VoIP-supporting product, for small-scale WANs in January 2000, the CX8000 series PBXs supporting built-in IP trunks and the NT-1000 VoIP gateway in April 2000, and the small NT-40 VoIP gateway in August 2000. In October 2000, Hitachi released the VoIP-supporting version of the MX key telephone system for small businesses. With this lineup, Hitachi provides optimal corporate network solutions that meet users' extensive needs for small to large-scale networks.

CX8000 PBXS AND NT-1000 GATEWAY

The NT-1000 gateway and CX8000 PBXs are the core systems for implementing a large-scale VoIP solution for WANs. Fig. 2 shows an example configuration of a VoIP network using these systems. Fig. 3 and Table 1 show the appearance of the NT-1000 and the major specifications of the NT-1000 and the CX8000 with IP trunk, respectively.

A WAN using these systems has several important features.

(1) Reduced communication costs

Voice is converted into IP data for transmission, enabling data and voice lines to be integrated, creating a more efficient network.

(2) Supported PBX voice applications

PBX voice applications are supported, including inter-office transfer, reset call between offices, calling number display, PHS (personal handyphone system) roaming between PBXs, and PBX network functions over an IP network. They are supported because the IP trunks and gateways use SS7 for the call-control protocol, which VoIP technology uses. SS7, together with the Dch common channel signaling system, is a representative example of signaling systems derived from the field of exchange systems. It is based on the idea of implementing VoIP technology while retaining the operability of conventional telephone systems as much as possible. This enables the systems to provide the operability of conventional PBXs to an IP network. There are also call-control protocols derived from the field of computers; a representative example is the ITU-T Recommendation H.323 protocol.

Note, however, that only voice communication can be provided in connection with PBXs other than CX8000, CX5000MS, CX3400MS, and CX3000MS, including other companies'.

(3) Traffic measurement

Incoming-call loss and per-destination traffic volume can be measured.

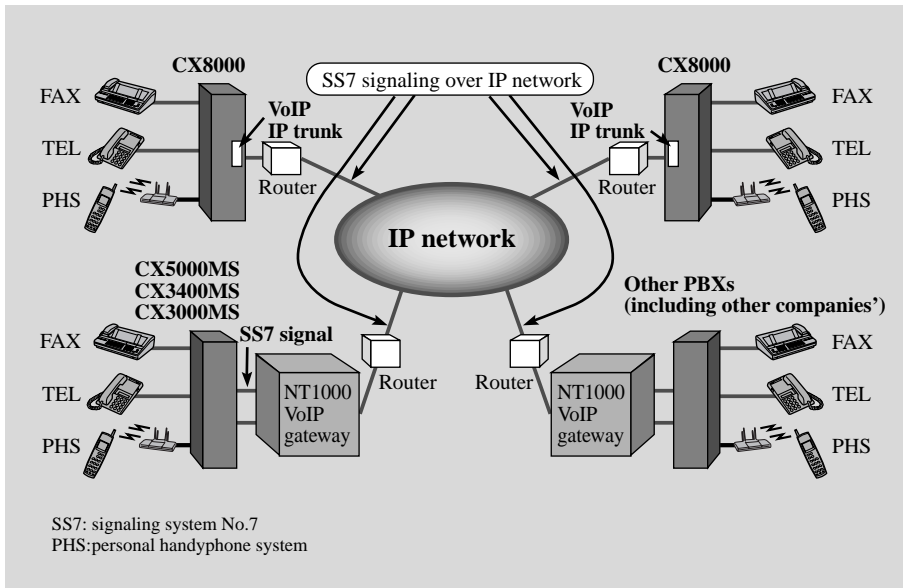(4) ITU-T Recommendation G.729A voice compression

Fig. 2—Example Configuration of VoIP Network Using NT-1000 Gateways and CX8000 PBXs.
The PBX network functions can be given to an IP network by using the CX8000 PBX with IP trunk, the NT-1000 VoIP gateway, and SS7. Note, however, that only voice communication can be provided in connection with PBXs other than CX8000, CX5000MS, CX3400MS, and CX3000MS, including other companies'.
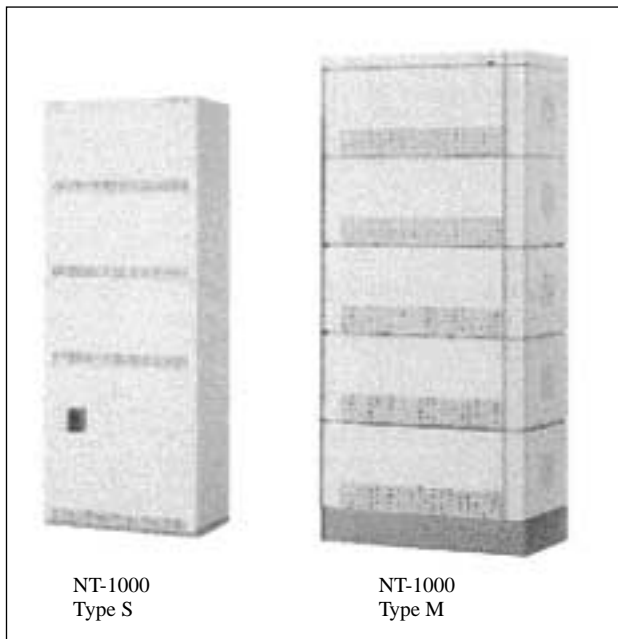

Fig. 3—Appearance of NT-1000 VoIP Gateway.
There are three types of NT-1000 VoIP gateways: type S can transfer 16 voice channels (max.) over an IP network, type M1 can transfer 96 voice channels (max.), and type M2 can transfer 248 voice channels (max.).

TABLE 1. Major Specifications of NT-1000 VoIP Gateway and CX8000 PBX with IP Trunk
The NT-1000 gateway can be given PBX functions by adding a card and software applications.

	NT-1000	CX8000 with IP trunk
LAN interface	10BASE-T	
PBX interface	2-Mbit/s TTC OD trunk	
Voice compression	ITU-T G.729A ITU-T G.711	
FAX communication	G3	
DTMF transmission	Enabled	
Call-control protocol	SS7	
Echo canceller	Enabled	
Buffer for absorbing jitter	Enabled	

Voice compression is done using 8-kbit/s CS-ACELP (ITU-T Recommendation G.729A), thereby achieving voice quality equivalent to that of 32-kbit/s ADPCM, resulting in highly efficient transmission with high quality.
(5) Ensured speech quality
Close-to-natural speech is ensured by use of functions to absorb jitter and prevent delays.
(6) Fax communication
Fax signals are automatically recognized so that users can transparently use G3 fax communication.
(7) Transmission of dual-tone multi-frequency (DTMF) signals
Transparency for DTMF signals is provided by distinguishing DTMF signals from voice signals and then encoding the DTMF signals for transmission.
(8) Accommodation of PBXs of other companies
Connections with PBXs of other companies are accommodated via the out-band-dialing (OD) trunk or the TTC 2-Mbit/s interface for speech by VoIP.
(9) Addition of PBX functions to NT-1000

The NT-1000 gateway can be given PBX functions by adding the necessary cards and software applications.

BUILDING LARGE-SCALE VOIP NETWORK
Considerations

The standardization of priority control, such as differentiated service (Diff-Serv), and the enhancement of operability, such as the transfer of common channel signals over an IP network, have reached the point where large-scale voice-and-data-integrated networks can now be built using VoIP technology.

An IP network is a best-efforts one, as mentioned earlier, so it must be designed considering delay, jitter, and loss of voice packets, especially when it is a large-scale network. Also must be considered are the IP addresses and numbering plan for extensions.

Hitachi's VoIP Corporate Network
Background

Hitachi's current corporate network was built more than a decade ago. It has privately owned access points, each of which consists of a TDM, a packet switch, and a toll switch-private branch exchange (TS-PBX) and is connected with adjacent branch offices by high-speed digital lines. The trunk lines between access points were upgraded to asynchronous transfer mode (ATM) private lines to provide higher-speed communications and reduce costs.

Hitachi is rebuilding this TDM-based corporate network to make it an IP-based one. An IP network is needed that can meet the need for high-speed transmission of IP data to create a supply chain in the company and to share resources among research laboratories, as well as to reduce communication costs.

Implementation

In the rebuilt network, Internet applications such as intranet Web and e-mail systems and conventional applications such as PBX extension systems and communication between mainframe computers are being unified by using the IP, GR2000 gigabit routers[3], NT-1000 VoIP gateways, and CX8000 PBXs with built-in IP trunks. The result is higher-speed Internet communication that enables implementation of VoIP technology.

ATM private lines and high-speed digital lines are being used to construct a network with a star topology, with its center at Omori (see Fig. 4). A star topology is possible because the TS-PBXs are no longer needed,
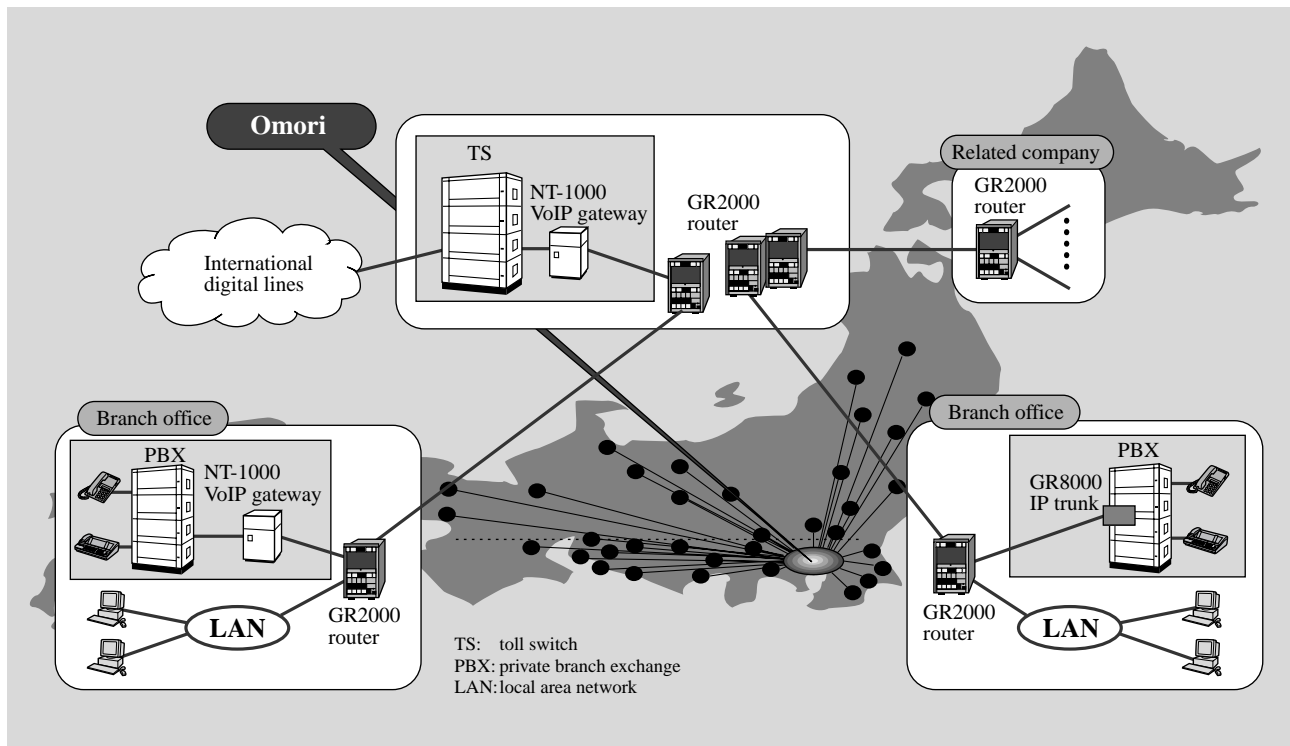

Fig. 4—Image of Hitachi's Corporate Network.
Using GR2000 routers, NT-1000 gateways, and CX8000 PBXs, Hitachi is rebuilding its corporate network.
Based on a star topology with its center at Omori, this new IP network can meet the need for high-speed
transmission of IP data, as well for reduced communication costs.

and the access points can be removed due to adoption of VoIP technology. This topology greatly simplifies the network facilities and reduces operation costs. Further cost reduction will be achieved by using IP-VPNs for the trunk lines.

IP technology has been introduced at several locations, and field evaluations are presently underway. By April 2001, IP technology will have been introduced at all branch offices. It will then be extended to related companies.

CONCLUSIONS

We have discussed the functions and features of the NT-1000 gateway, a VoIP-supporting product for WANs, and of the CX8000 PBX. Hitachi's own large-scale corporate network was described as an example system using these products.

IP technology will be increasingly adopted into corporate networks as priority control technology is improved.

Hitachi will apply the know-how it has acquired from upgrading its own corporate network to customer systems and will develop and enhance the network products needed to meet users' needs.

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