Papers on UNIVERGE Hardware

Small-Scale IP Telephony Server: UNIVERGE iExpress5800/Lite

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ABSTRACT As IP networks have been rapidly expanding in recent years, the move toward VoIP (Voice over IP)/IP telephony services has been progressing at a very fast pace. This paper describes the UNIVERGE iExpress5800/Lite, a SIP (Session Initiation Protocol) and Presence server for small- and medium-sized businesses and SOHOs. This server features a call control function based on SIP protocol IETF RFC3261 as well as a presence function.

KEYWORDS **VoIP (Voice over IP), IP phone, SIP (Session Initiation Protocol), Presence server, VoIP gateway**

1. INTRODUCTION

As the broadband environment spreads, an increasing number of companies are aiming for lower network-related TCO, higher communication efficiency, greater business efficiency, a revolution in work style, etc., through the integration of data networks and voice networks and the introduction of IP telephony. Furthermore, carriers have started offering IP telephony services on a full scale, and in the fall of 2003, the 050 IP phone prefix entered service. The IP telephony market is thus on track to expand every year.

Amidst such a market environment, the need for SIP servers that enable the cost effective introduction of IP telephony is rising among small offices and departments. The UNIVERGE iExpress5800/Lite was commercialized to answer this market need.

2. NETWORK CONFIGURATION

The UNIVERGE iExpress5800/Lite consists of the following products.

① SIP (Session Initiation Protocol) server iExpress5800/Lite
② SIP compliant VoIP (Voice over IP) gateway IPMASTER

IPMASTER supports the analog public circuit network, INS Net 64 network, and OD lines (leased lines). SIP group line telephone sets and SIP Softphones are provided as SIP terminals. **Figure 1** shows an example of a network configuration in the case of introduction at two points using iExpress5800/Lite.

The following types of VoIP (Voice over IP) services are possible with the configuration example shown in Fig. 1.

1) Telephone calls between SIP terminals in Base A and Base B (Bases A and B are connected via LAN through an IP-VPN network, etc.)
2) Telephone calls to general subscribers connected to a public telephone network, ISDN network, or IP telephone network via SIP terminals in Base A and Base B, and VoIP gateway (IPMASTER).

3. iExpress5800/Lite (SIP SERVER)

3.1 SIP Server Function

The SIP server function complies with SIP, a standard established by the IETF (Internet Engineering Task Force). This function is that of a proxy server (stateful) that manages subscriber information such as telephone numbers and IP addresses and performs signal control for VoIP communication. The main functions that are provided are described below.

(1) SIP Routing

In SIP, the "user name@domain name" format called SIP-URI is used as the ID that identifies a

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communication party, whereas iExpress5800/Lite supports number routing, which is the routing method used by traditional exchanges, whereby a telephone number is used as the user name part. The dial number routing table is managed in the SIP server, and upon receiving a dial number from a SIP terminal, the SIP server determines the connection destination.

(2) Subscriber Information Management

The telephone number, authentication parameter (user ID, password), IP address, etc., are managed as subscriber information in a built-in database.

(3) User Authentication

IP address authentication and HTTP digest authentication, which is a standard SIP authentication mechanism, are supported as user authentication functions. IP address authentication is performed during terminal registration using the REGISTER message from a SIP terminal. HTTP digest authentication is performed upon receiving the REGISTER message from a SIP terminal or the INVITE message (session initiation request) sent during call origination.

(4) Communication Logging

A function to record the call information generated during the call connection (session initiation) process as a communication log is provided. The call information is recorded not only in the case of completed calls (calls that have been successfully placed) but also in the case of incompleted calls (calls that could not be placed).

(5) Additional Services

It is possible to form groups of multiple SIP terminals and set group lines and main numbers. Proxy answering of calls incoming to a group line, etc., is possible via sharing among the SIP terminals in the group. Automatic call distribution of calls incoming to a main number to available terminals set to the group is also possible.

Furthermore, unconditional, busy, no answer, and logout transfer are also supported. It is also possible to transfer incoming calls when communication is not possible, such as during busy, no answer, etc., or during SIP Softphone logout, by setting in advance a transfer destination for each SIP terminal.

3.2 Network Services

iExpress5800/Lite is a small-scale SIP server that can accommodate from 20 to 50 subscribers per server.

By adding one iExpress5800/Lite server, routing between two servers is possible, allowing the connection of up to 100 subscribers.

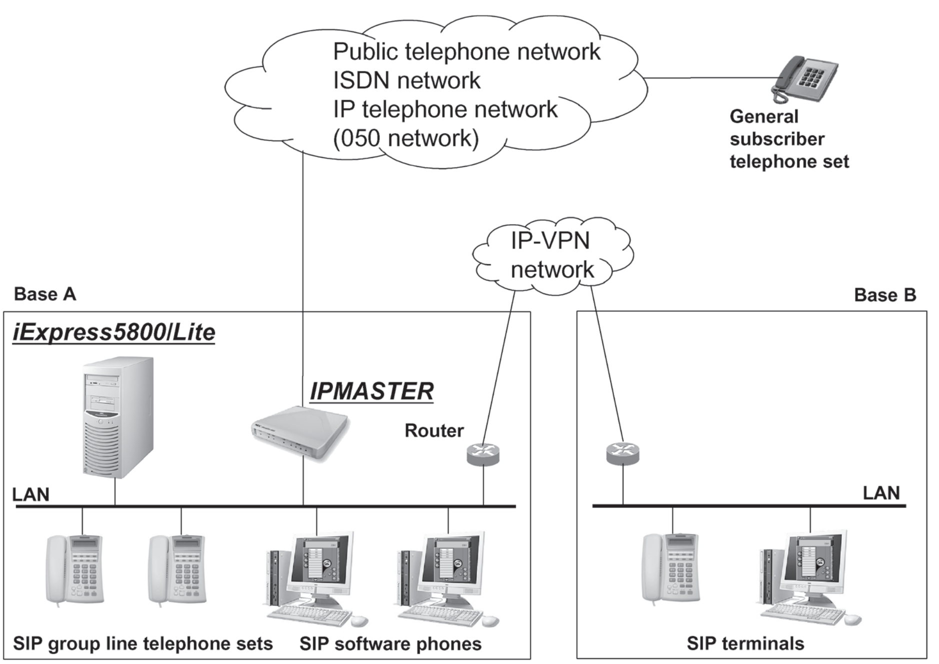

Fig. 1 UNIVERGE iExpress5800/Lite network configuration example.

3.3 Additional Functions

(1) Presence Server Function

iExpress5800/Lite features a presence server function. It supports IETF’s SIMPLE (SIP for Instant Messaging and Presence Leveraging Extensions), works with the SIP server function, manages presence information (status information such as online/offline, busy, absent) shared with SIP software phones, and can notify presence information to other registered SIP software phones.

Figure 2 shows an example of the operation when a SIP Softphone is started, with users B to K registered to the SIP server and in the online status (presence information = “online”), and user A not registered to the SIP server and in the offline status. When the SIP Softphone is started by user A, a REGISTER message is transmitted to the SIP server for terminal registration (①). When terminal registration has been received by the SIP server, the fact that user A has been registered is notified to the presence server (②). The presence server requests the presence information for user A who was newly registered and receives the presence information from user A (③). If the online status, the presence server updates the held presence information online, and notifies the presence information to each SIP Softphone (④). When user B and user K communicate via SIP Softphone, the presence information becomes busy, and their presence information (busy status) is notified and displayed to each SIP software phone.

In this way, using a presence server and SIP Softphones in addition to a SIP server, the status of the other parties (presence) can be checked and communication can be performed via calls, simple messages, e-mails, etc.

Moreover, the presence server function allows linking of presence information and other network services via SOAP.

(2) Simple File Server Function

This function implements a file server on a Windows machine where a shared file area has been set up.

(3) Management Console

The functions offered by iExpress5800/Lite can be managed via HTTP. Backup restore of setting files, SIP server function/presence server function/file server function settings, management of logs, module update, etc., can be performed.

3.4 Software Configuration

Figure 3 shows the software configuration of the SIP server function and presence server function.

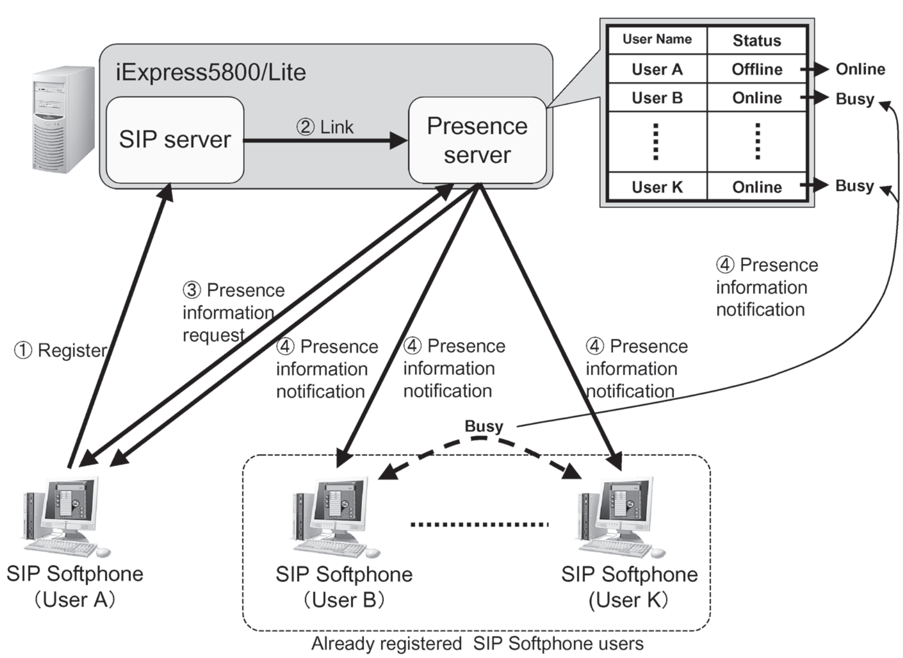

Fig. 2 Presence information operation example.

In addition to the standard SIP control compliant with the SIP protocol on the OS, the SIP server function features a call control section with expanded SIP control for installing a transfer and other functions provided as additional services not prescribed by the SIP protocol. A highly expandable software configuration is realized by separating the protocol-specific processing, and defining service start API (Application Program Interface) for additional services. The call control section features a SIP common control section that provides subscriber data management, a translation function (number translation function), routing control, call information management, etc.

Moreover, the presence server function provides presence protocol stacks on the OS, and over these there is a presence application that controls and distributes presence information.

4. IPMASTER (SIP COMPLIANT VoIP GATEWAY)

4.1 Function Overview

The following IPMASTER models are available depending on the type of circuits on the existing phone network to be connected.

① IPMASTER-1041A (analog public circuit connection)
② IPMASTER-1042A (OD circuit connection)
③ IPMASTER-1043A (INS Net 64 circuit connection)

IPMASTER is a series of SIP protocol compliant gateway devices that realize interconnection with existing telephone networks. These gateway devices support the G.711, G.729a, and G.723.1 voice codec standards and select the standard separately for each call. They also provide linking with SIP servers, allowing the use of an independent call receiving function and PBX dial-in/additional dial-in function (IPMASTER-1041A/1043A).

Figure 4 shows a network configuration example using the various IPMASTER models.

IPMASTER-1041A and IPMASTER-1043A are used connected to an analog public circuit and an INS Net 64 circuit, respectively. IPMASTER-1042A can be connected via a conventional voice network through connection to an installed PBX, KTS, etc., via an OD circuit.

4.2 Protocol Conversion

IPMASTER units can perform protocol conversion between an existing telephone network and SIP protocol to allow interconnection. **Figure 5** shows an example of the call control sequence when using an ISDN network and IPMASTER-1043A (hereafter, VoIP gateway), iExpress5800/Lite (hereafter, SIP

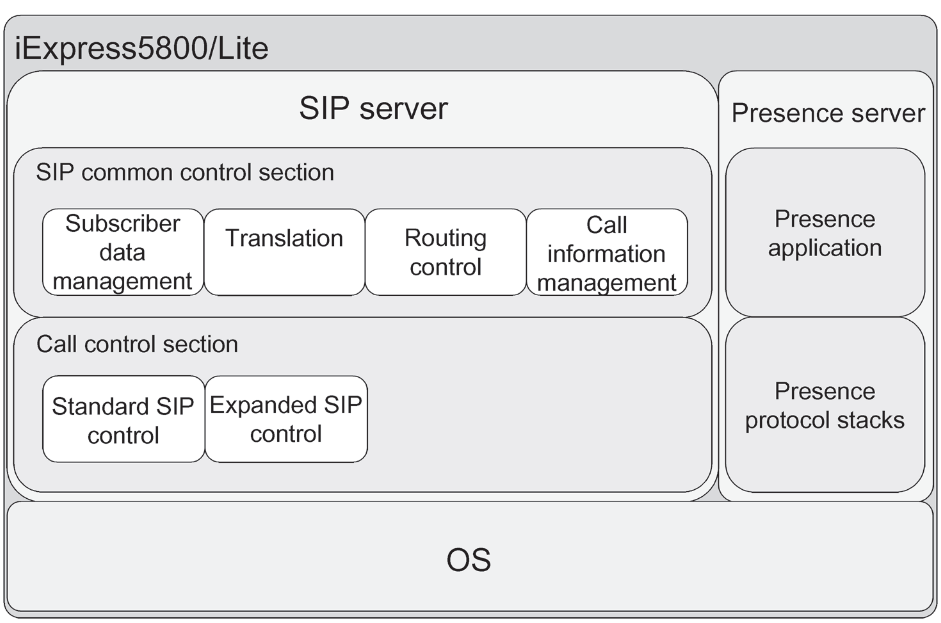

Fig. 3 iExpress5800/Lite software configuration.

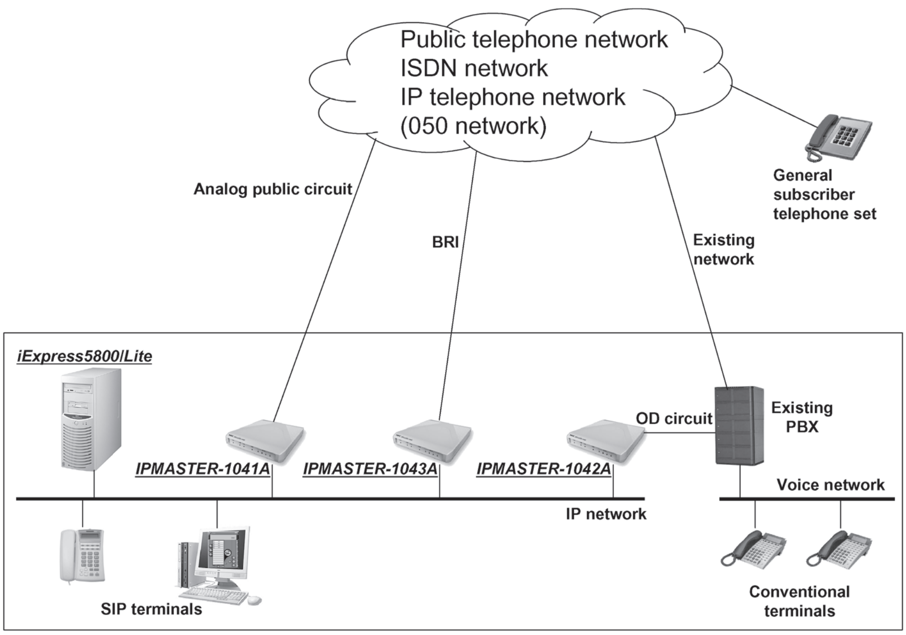

Fig. 4 IPMASTER VoIP gateway configuration example.

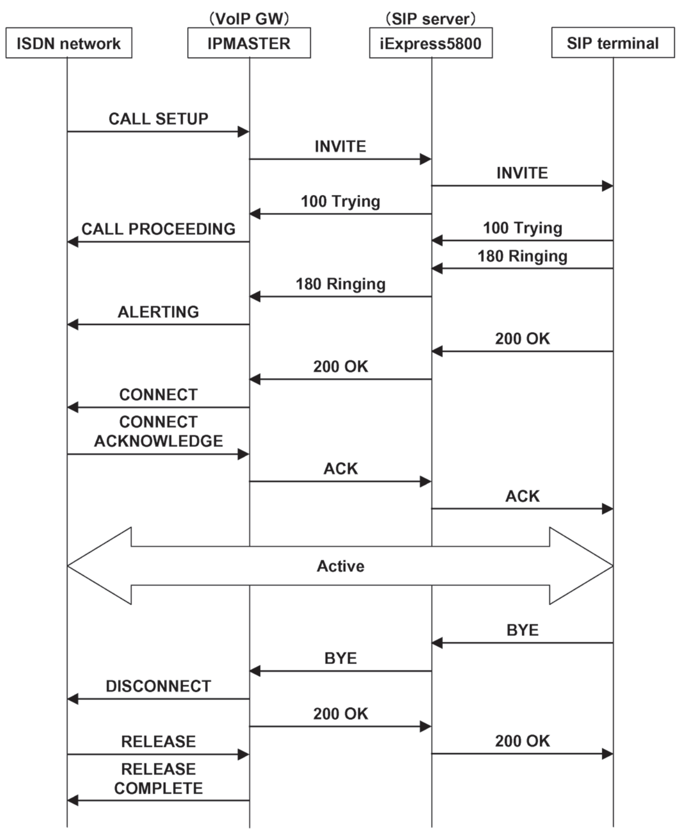

Fig. 5 Example of all control sequence via VoIP gateway.

server).

The sequence shown is that when a call is placed from a general subscriber telephone set connected the ISDN network. When a call is made from a general subscriber telephone set, a call setting message is delivered to the VoIP gateway from the ISDN network via the D channel. In the VoIP gateway, SIP protocol conversion is performed for incoming circuit numbers, and an INVITE message is transmitted to the SIP server. In the SIP server, the SIP terminal is identified from the SIP message based on the preset number translations, and an INVITE message is transmitted. After the transmission, a trying message is returned to the VoIP gateway, and a call setting acknowledgement message is returned from the VoIP gateway to the ISDN network. The status of the SIP terminal that received the INVITE message changes to the ringing status, a Trying message and Ringing message are returned to the SIP server, a Ringing message is returned from the SIP server to the VoIP gateway, and a call message is returned to from the VoIP gateway to the ISDN network.

If there is an answer at the SIP terminal in the ringing status, the answer status code OK is returned to the SIP server, and an answer message is returned to the ISDN network side. After the response, the ACK message, which is the final answer, is transmitted back from the VoIP gateway to the SIP server, and from the SIP server to the SIP terminal, and communication between the general subscriber telephone set and SIP terminal starts via the B channel of the ISDN network. During the communication session, packets are transmitted over the LAN between the SIP terminal and VoIP gateway without traveling via the SIP server.

If the call ends with the SIP terminal disconnecting, a BYE message, which is a session termination request, is transmitted, and a disconnection message is transmitted to the ISDN network side. Following transmission, the OK message, which is the answer status code, is returned from the VoIP gateway, and on the ISDN network side, the channel is released.

5. CONCLUSION

This paper has introduced the UNIVERGE iExpress5800/Lite, a small-scale IP telephony server that realizes a new work style integrating data communication and voice communication.

VoIP/IP telephony services are expected to rapidly expand as IP networks grow, and NEC Corporation will continue releasing flexible products to serve the needs of this market.

* Windows is a registered trademark of Microsoft Corporation in the United States and other countries.

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