4-4 Approach of VoIP/SIP Interoperability Task Force

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In this research, it achieved interoperability of VoIP systems using SIP in both Multi-vendor and Multi-provider environments, and “VoIP/SIP interoperability task force” (thereafter only “VSTF”) was established with the JPNIC・WIDE project as an activity organization.

This VSTF provides and operates a test-bed for interoperability verification/evaluation, and provides the minimum requirement of evaluation and test specifications. The interoperability is verified by using the TTC standard as a standard specification between VoIP terminals and the career’s SIP server, or IP-PBX. When trouble occurs at a verification, we report on the trouble case to a domestic standardization organization.

Keywords
SIP, VoIP, Interoperability, VoIP/SIP interoperability task force

1 Introduction

In 2002, a major ISP launched Japan’s first IP telephony service. Since then, IP telephony has been spreading not only among companies but also among general users. However, individual carriers and vendors initially developed VoIP services on their own, and adequate interoperability has yet to be established among different servers and terminals.

These vendors and providers must establish basic interoperability if VoIP systems are to become as widespread as conventional phones and if VoIP-based multimedia services are to find wide use in society at large. With the aim of establishing interoperability among different VoIP systems in multi-provider and multi-vendor environments, we established the Task Force, in collaboration with the JPNIC and WIDE Projects.

In this paper, we will explain the Task Force’s activities, give an overview of SIP, and illustrate actual models used to verify interoperability. Additionally, we will summarize the results of interoperability verification experiments performed to date.

2 Activities of VoIP/SIP interoperability task force

2.1 Goals of activities

(1) To perform technical verification activities to establish interoperability among VoIP systems using SIP under the following two environments:
  (i) Multi-vendor environment
  (ii) Multi-provider environment

(2) To make preparations for interoperability evaluation/testing:
  (i) Minimum specifications for evaluation/testing
  (ii) Evaluation/testing software based on the above specifications
  (iii) Testbed environment and specific activities in interoperability evaluation/testing

(3) To contribute to the promotion of global cooperation and business activities, as follows, for the purpose of achieving the above goals:

(i) Creation of scenarios for VoIP system evaluation/verification
(ii) Release of software for VoIP system evaluation/verification
(iii) Improved software quality and establishment of interoperability of VoIP-related devices
(iv) Establishment of interoperability among VoIP systems
(v) Establishment of portability among VoIP devices
(vi) Presentation of findings and suggestions to domestic and international standards organizations (IETF: Internet Engineering Task Force, ITU-T: International Telecommunication Union-Telecommunication Standardization Sector, TTC: Telecommunication Technology Committee, HATS: Harmonization of Advanced Telecommunication Systems Conference, etc.)

2.2 Overview of SIP

SIP (Session Initiation Protocol) is an application-layer signaling protocol that initiates, modifies, and ends multimedia sessions over an IP network. SIP was initially proposed by the IETF SIP Working Group, and subsequently standardized as RFC3261. SIP can be used to implement a wide range of services, including IP telephony, videoconferencing, instant messaging, and the presence function.

The ITU-T protocol H.323 is similar to SIP in terms of functions. However, SIP is viewed as simpler and is considered to consume fewer resources than H.323. SIP initiates, modifies, and ends sessions, but does not define the data to be exchanged during these controlled sessions. SIP can therefore be used flexibly according to the type of data exchanged over the network—for example, IP telephony (voice), videoconferencing (voice and image), and instant messaging (text messages). Due to its significant scalability and ease of integration with other systems, SIP is now receiving attention as a standard protocol for real-time communications.

We will now describe how a session takes place. For example, if Alice uses an IP phone to call Bob, the devices involved in this session include the two IP phones (the "Terminal Equipments (TEs)"), and SIP proxy servers A (atlanta.com) and B (biloxi.com) for these IP phones. A SIP proxy server, equivalent to an exchange in PSTN, receives requests from the TEs or proxies and issues these requests to the appropriate TE or proxy.

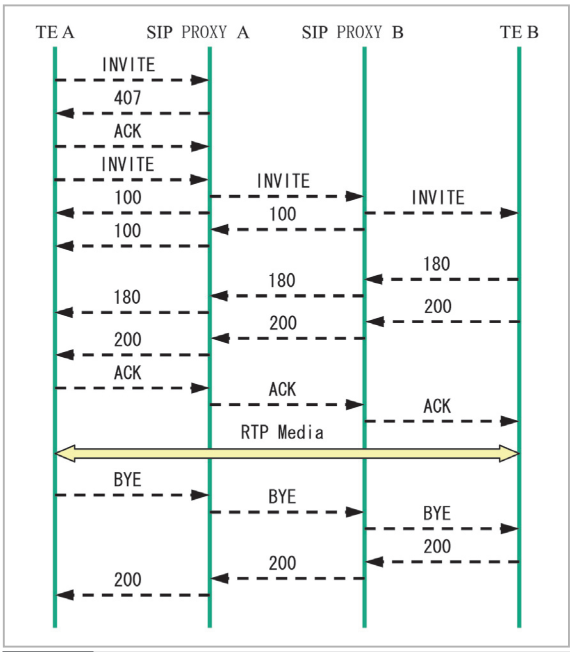

Fig. 1 *Example of IP phone session from establishment to disconnection*

A session starts with an INVITE message. URIs (Uniform Resource Identifiers) such as "sip:alice@atlanta.com" and "sip:bob@biloxi.com" are used to identify the TE. Alice sends an INVITE message to sip:bob@biloxi.com to establish a session with Bob.

Figure 2 shows an example of an INVITE

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INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bK776asdhds
Max-Forwards: 70
From: Alice <sip:alice@atlanta.com>;tag=1928301774
To: Bob <sip:bob@biloxi.com>
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142
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Fig.2 *Example of INVITE message*

Table 1 *List of status codes*

1xx: Informational	Request has been received; continuing to process.
2xx: Success	The action was successfully received, understood, and accepted.
3xx: Redirection	Further action is required in order to complete the request.
4xx: Client Error	The request contains erroneous syntax or cannot be fulfilled at this server.
5xx: Server Error	The server failed to fulfill an apparently valid request.
6xx: Global Failure	The request cannot be fulfilled at any server.

message sent from Alice to Bob:

As shown in the figure, SIP uses messages in user-readable ASCII code. Since this format is similar to HTTP and SMTP, these messages are easy to understand.

Upon reception of the INVITE, proxy A will send it to proxy B (biloxi.com) because the destination is bob@biloxi.com (Fig.1). Proxy A will also send a provisional response “Informational 100: Trying” to Alice to indicate “Sending INVITE to Proxy B” (Fig.1).

“100” is one of the status codes indicating request results. As shown in Table 1, SIP uses a set of status codes that is essentially an extended version of those specified for HTTP:

IP phone terminals or SIP servers are supposed to be interoperable if they are compliant with IETF’s RFC3261 standard. However, vendors initially tended to develop servers (VoIP exchanges) and IP phones in combination for use in closed systems, and they sometimes added extended functions on their own. Furthermore, URI formats differ from vendor to vendor, and some aspects are not specifically defined by the RFC standard. As a result, adequate interoperability has yet to be ensured among VoIP providers or among different vendors’ SIP-enabled products.

To solve this problem, we are working to establish interoperability in terms of standards and implemented functions.

2.3 Models for testing

To verify interoperability under actual operational conditions, we created simulation models for testing as described below:

(1) TE–ISP testing

This model is designed to test a UNI (User Network Interface) between SIP terminals connected to a SIP server, verifying interoperability in a multi-vendor environment in which different vendors’ terminals are connected to a single provider. Specifically, interoperability is assessed between a SIP server and a SIP terminal, as well as interoperability among different vendors’ SIP terminals connected to a single SIP server. Figure 3 shows the model under test.

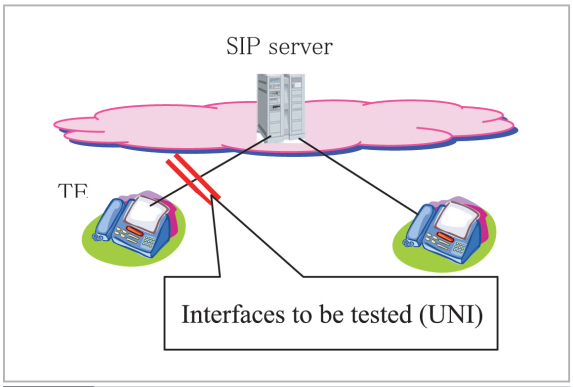

Fig.3 *Model for TE–ISP testing*

(2) ISP–ISP testing

This model is designed to test the NNI (Network–Network Interface) between SIP servers, verifying interoperability in a multi-provider and multi-vendor environment in which different vendors’ terminals are connected to different providers’ SIP servers. Specifically, interoperability is verified between SIP terminals connected to different SIP servers. In this case, the first SIP terminal is connected to one SIP server, and the second

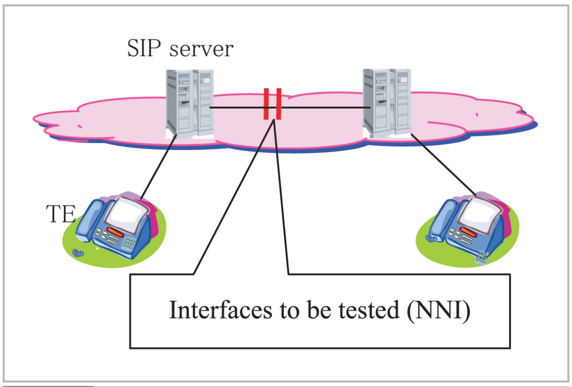

Fig.4 Model for ISP–ISP testing

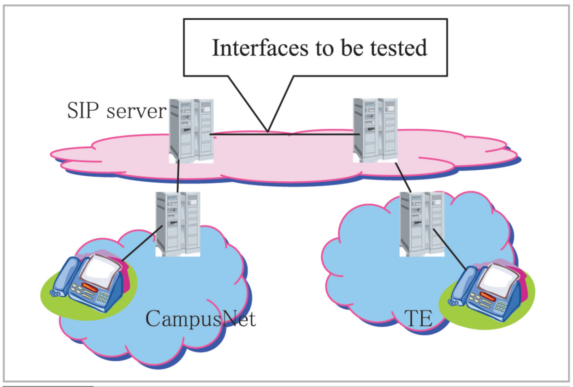

Fig.6 Model for CampusNet–CampusNet testing

SIP terminal is connected to another SIP server. Figure 4 shows the model under test.

(3) CampusNet–ISP testing

This model is designed to test the UNI/NNI between providers' SIP servers and a SIP server (such as IP-PBX) located within a private network, verifying interoperability in a multi-provider (multiple SIP servers) and multi-vendor (multiple SIP terminals) environment. Specifically, interoperability is verified between SIP terminals connected to providers' SIP servers and SIP terminals connected to a SIP server located within a private network. Figure 5 shows the model under test.

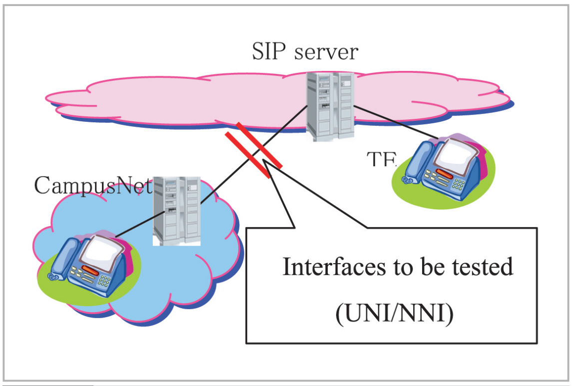

Fig.5 Model for CampusNet–ISP testing

(4) CampusNet–ISP–(???)–ISP–CampusNet testing

This model is designed to test UNI/NNI between SIP servers (such as IP-PBXs) located within different private networks connected via multiple providers' SIP servers, verifying interoperability between SIP terminals within different private networks connected via multiple providers' SIP servers. Figure 6 shows the model under test.

2.4 Verification results obtained to date

Finally, we will report on the results of verification testing we have performed since the establishment of the Task Force.

We performed TE–ISP testing once or twice with each of the following providers: Fusion Communications Corporation, KDDI Corporation, NTT Group (NTT Service Integration Laboratories, NTT Communications Inc., NTT East Corporation, NTT West Corporation), and Japan Telecom Co., Ltd. Participant vendors include Iwatsu Electric Co., Ltd., INTEC Web and Genome Informatics Corporation, NEC AccessTechnica, Ltd., Cisco Systems, Inc., Softfront, Hitachi Communication Technologies, Ltd., FUJITSU Ltd., and Yamaha Corporation. In each case the success rate was above 99%. (Note that if a terminal did not offer a certain function, we omitted the relevant verification items). Nevertheless, we discovered a number of problems through these verification experiments and requested both the providers (of the SIP servers involved) and the vendors (of the applicable SIP terminals) to enact measures based on the TTC standard. For problems considered to be of particular importance, we made suggestions to TTC, a domestic standards organization.

Table 2 List of TE–ISP tests

Provider	No. of terminals used	No. of test items (Basic + extended)
Fusion Communications	6 (1st time) 4 (2nd time)	7 + 10 7 + 54
KDDI	6	7 + 30
NTT Group	5	7 + 54
Japan Telecom	5	7 + 54

Fig.7 Interoperability verification testing

We also performed ISP–ISP testing with the following upstream providers: Fusion Communications Corporation, KDDI Corporation, NTT Group (NTT Service Integration Laboratories, NTT Communications Inc., NTT East Corporation, NTT West Corporation), and Japan Telecom Co., Ltd. We used seven terminals in total from six vendors: Asgent Inc., Iwatsu Electric Co., Ltd., Oki Electric Industry Co., Ltd., Cisco Systems, Inc., Fujitsu, and Yamaha Corporation.

We verified interoperability for 23 items with a basic configuration and achieved a 99.6% success rate. (Note that if a terminal did not offer a certain function, we omitted the relevant verification items). For problems considered to be of particular importance, we made suggestions to TTC.

2.5 Conclusions

In this paper we outlined the activities of the VoIP/SIP Interoperability Task Force and presented a discussion on SIP technology. We also reported on the results of verification testing performed to date. The models described above were used in testing, and wherever a problem was discovered, we made the appropriate suggestions to domestic standards organizations.

Going forward, we will design more verification scenarios with new combinations of components and will continue in our tests. Specifically, we intend to invite foreign SIP terminal vendors to participate in our activities, with the aim of establishing interoperability among VoIP systems, both domestically and internationally.

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