TTC標準
Standard

JJ-22.03

Technical Specification on User Information Interface between Private SIP Networks

First Edition

Established on August 27, 2007

THE TELECOMMUNICATION TECHNOLOGY COMMITTEE

Introduction

This document provides the TTC original Standards formulated and put into effect by the Technical Assembly. It contains unabbreviated version of ‘JJ-‘ Standards, which have not been defined as international standards.

In case of dispute, the original to be referred is the Japanese version of the text.

We trust that greater understanding of TTC Standards by a wider range of users will further contribute to the development of telecommunications.

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1. Introduction

Since last year, the Private Network Interface Sub-Working Group of the Private Network Special Committee has implemented the standardization of IP (Internet Protocol) based private networks (IP networks) between IP-PBXs (Private Branch eXchanges). Considering future trends in markets and international recommendations, it is necessary to study supplementary service technology and application linkage technology based on SIP (Session Initiation Protocol) within private networks. In addition, there is a need for unique services within private networks that use individual headers specified in this standard.

Because of the background and reason stated below, this standard summarizes technical specifications on inter-connection interface between private SIP networks.

- Makes it possible to promptly accommodate unique, supplementary services required in a private network by newly defining the SIP protocol for private network.
- Increase connectivity by defining a light protocol for use in an operating agency, different from the one used in a carrier.

2. Revision History

Edition	Date of establishment	Description
First Edition	August 27, 2007	Established.

3. Miscellaneous

(1) Recommendations, standards, etc., referenced

TTC standard: JJ-22.00 The Guideline for the Architecture of the Technical Specifications for Private SIP in TTC

Revison.11 December 6, 2007

TTC standard: JJ-22.01 Technical Specifications on Inter-connection Interface between Private SIP Networks

Revison.11 December 6, 2007

TTC standard: JJ-22.02 Inter-work Specifications between Private SIP Network and private ISDN Network.

Revison.11 December 6, 2007

(2) Associations with other domestic standards

No associations with other domestic standards.

4. Organizational Unit Preparing Standards

First Edition: Private Network Special Committee

1. Overview

1.1 Scope of this standard

This standard is based on a connection interface (interface C) between inter-connected private SIP networks in a network connection architecture specified in JJ-22.00 <The Guideline for the Architecture of the Technical Specifications for Private SIP in TTC>.

Also, this standard is intended to ensure that in inter-connected private SIP networks, management in the private networks is facilitated while high interconnectivity is maintained, on the assumption that the private SIP networks conform to the provisions of this standard.

1.2 Purpose and provisions of this standard

Standardization is promoted by mainly focusing on conformance to ISO/IEC standards and IETF standards ,but when we look at the present domestic market, we can see a number of cases in which each enterprise cannot receive supplementary services using standard SIP in a timely manner because of excess attention to international and domestic standardization.

If we pay attention to standardization protocols, we can consider supplementary services in conformance with ISO/IEC 22535 (tunneling), but they are merely for transferring Qsig information, and cannot send user-specific information.

By the consideration above, there arises a need for frame information in which the user can implement services uniquely, thus this standardization material is created.

1.3. Terms

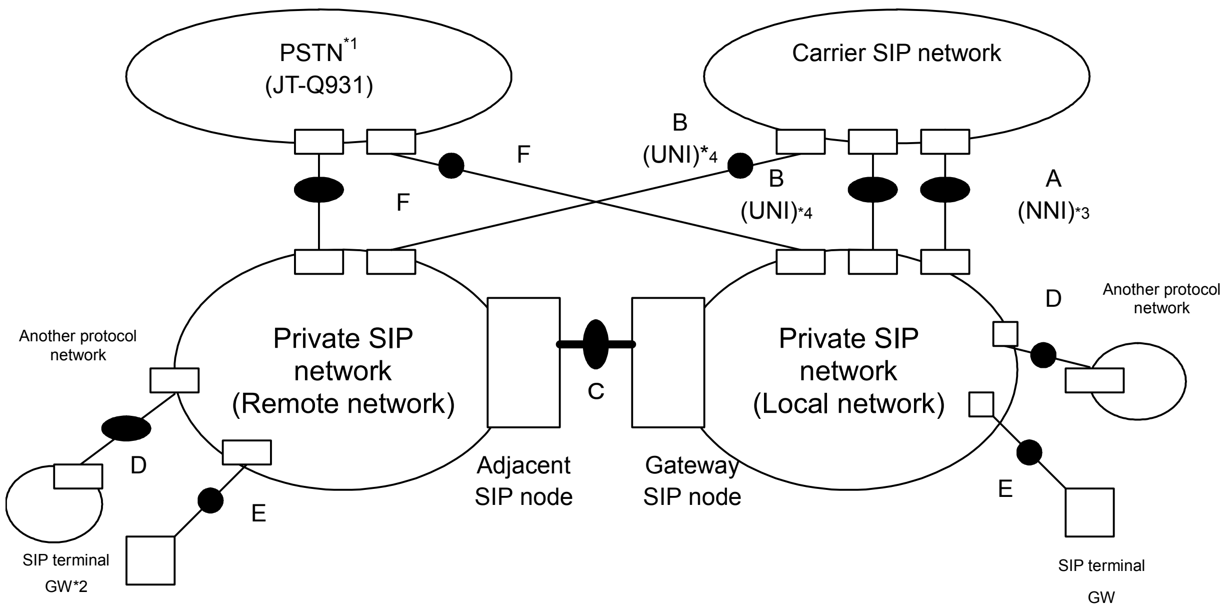
The terms used in this standard shall conform to JJ-22.00, JJ-22.01, and JJ-22.02.

2. Connection Configurations

2.1 Basic connection configuration

This standard describes the conditions for connection interfaces to managed private SIP networks that are applicable to interface C specified in the private SIP network interconnection model shown in Figure 0-1.

In this standard, a private SIP network that has an interface conforming to the provisions of this standard is called a "managed private SIP network". It is assumed in the remainder of this standard that the term private SIP network refers to a "managed private SIP network". (The private SIP network interconnection model shown below is the one cited from JJ-22.00.)



*1:PSTN･･･Public Switched Telephone Networks
*2:GW･･･GateWay
*3:NNI･･･Network Network Interface
*4:UNI･･･User Network Interface

Figure 0-1　　Private SIP network interconnection model

2.2 Scope of this standard

This standard defines the provisions inter-server connection (C) .

3. List of Abbreviations

C	C reference point
IP	Internet Protocol
PINX	Private Integrated Services network eXchange
Qsig	Signalling information flows at the Q reference point
SIP	Session Initiation Protocol
SCM	Signalling Carriage Mechanism
SS	Supplementary Service
NWL	Network Layer

4. Definition Related to the User-Specific SIP Signalling Method

4.1. Definition

Mechanism for sending non-Qsig information notifications and for implementing application services in accordance with each private network in a timely manner.

4.2. Definition in comparison with ISO/IEC22535 (tunneling)

- Flow of signalling information in ISO/IEC 22535

Transmit Qsig information through the SIP body part. It is applied to transit of Qsig information.

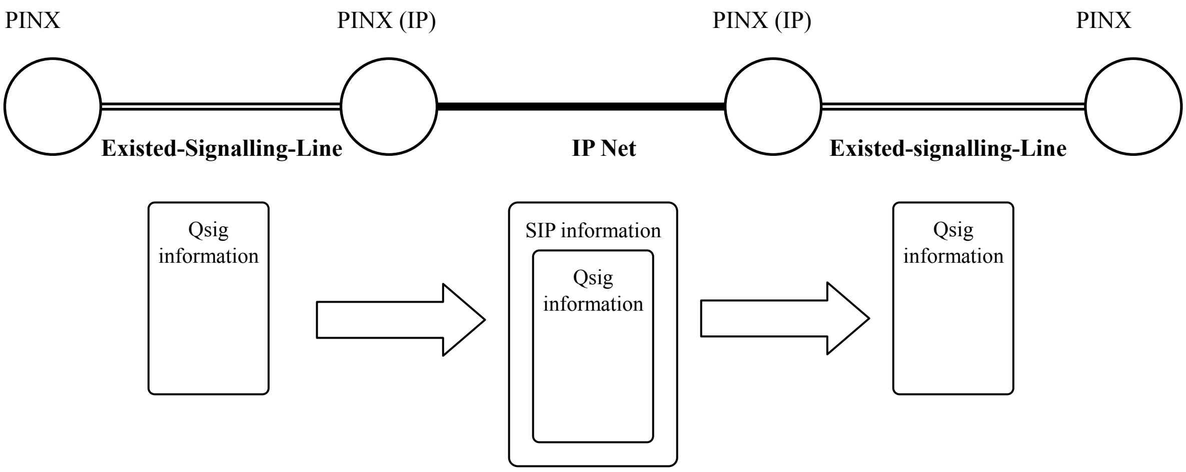


Figure 4-1 Flow of signalling information in ISO/IEC standards

- Flow of signalling information in the user-specific SIP protocol

The main function is to send information about application serviced in each device.

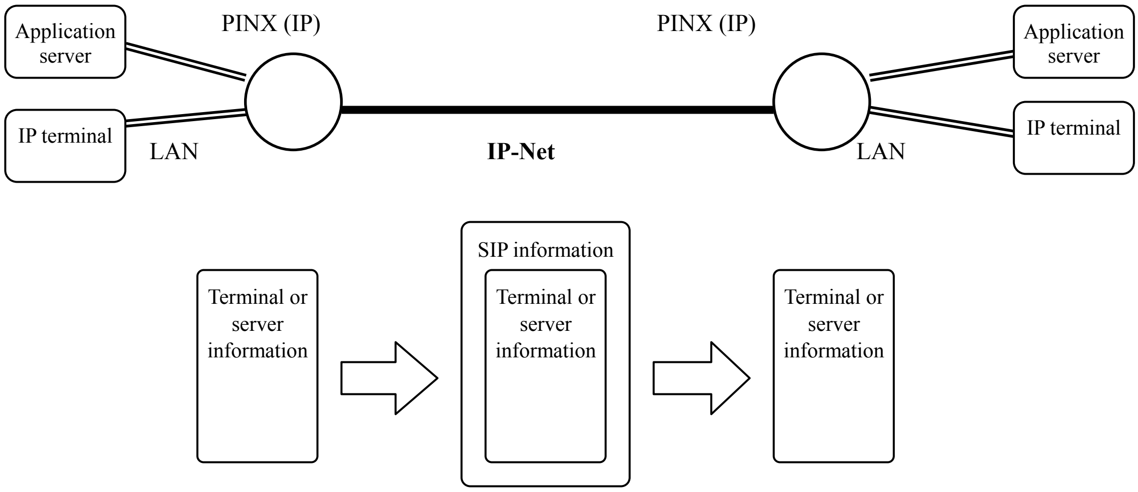


Figure 4-2 Flow of signalling information in the user-specific SIP protocol

5. General Principle

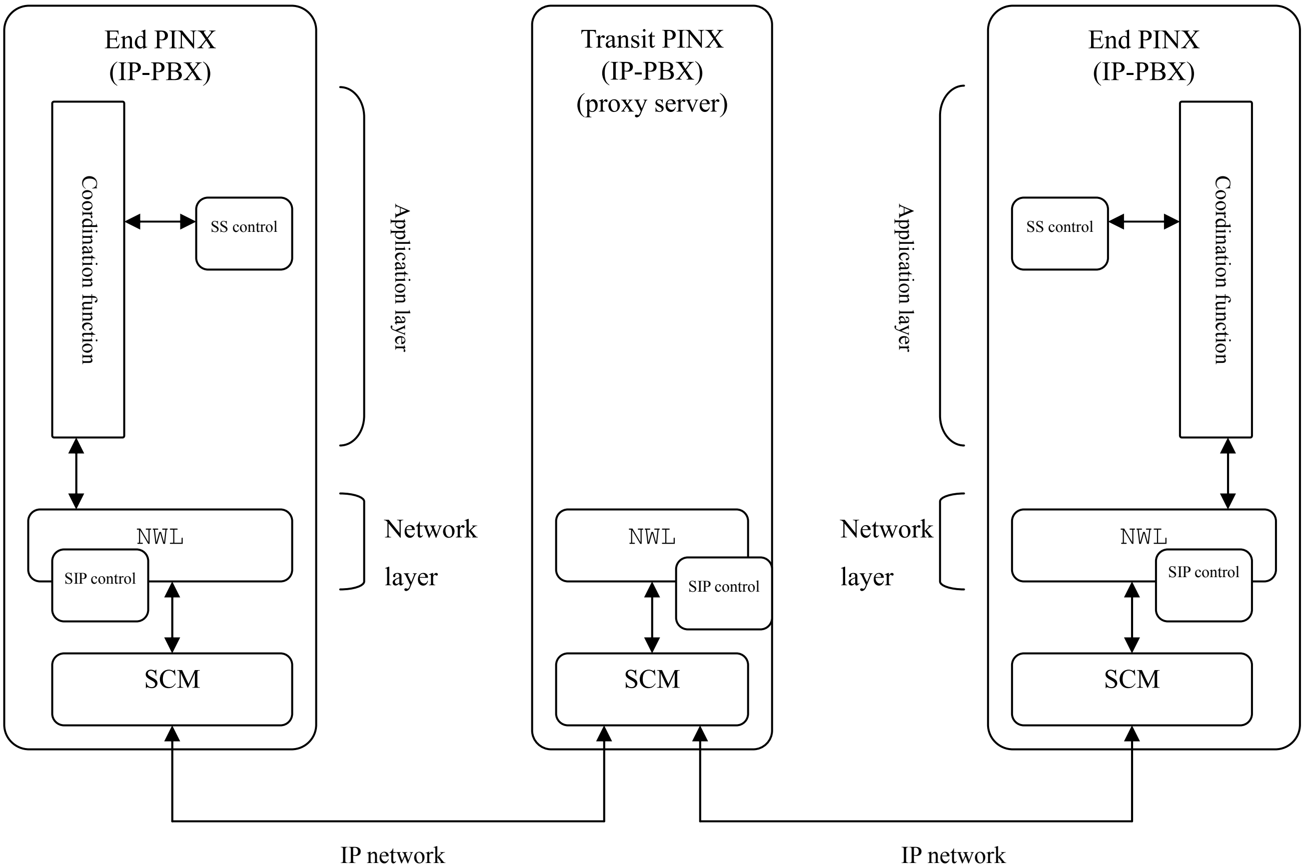
User-specific protocol information, as defined in this standard, provides a method of exchanging the signal information area for the control of supplementary services specific to each enterprise on an IP network.

It is not that this information controls supplementary services by itself, but that user-specific information is provided for generic purposes from an SS control entity.

This user-specific protocol information operates at the C reference point between two PINXs, together with the UDP protocol for basic call control (RFC3261). This also uses the service of the signalling carriage mechanism (SCM). The user-specific protocol specifies the mechanism for supporting supplementary services related to, or completely independent of, basic calls.

5.1 Protocol model

The configuration diagram below is one that is defined with the conceptual model of the generic purpose protocol. At the highest layer (application layer), the actual supplementary service protocol operates between supplementary service control (SS control) entities at the same layer as the service-specific one. The operation of specific SS control entities is outside the scope of this standard. The SS control entities send the vendor-specific information necessary for supplementary services to the coordination functions and prompt them to edit. The SS control entities send information to the network layer (NWL) via the coordination functions. NWL controls use protocol control services at the network layer to send the received user-specific information from the coordination functions to the SIP protocol control parts and write the information to the user-specific areas found in the SIP-Body parts. The network layer is also responsible for controlling routing functions. After writing, SIP-method information is sent to the SCMs. The SCMs shown here provide a function to send and receive Signalling information at layers 1, 2, and 3.



5.2 Definitions of the fields of user-specific SIP protocol information

5.2.1 Definition of user-specific specification information

5.2.1.1 Framework to use to carry user-specific information in a SIP message

If user-specific information is to be incorporated into a SIP-method, a PINX shall incorporate the user-specific information into the MIME body parts of a SIP request and a response by newly setting the media type to application/Enterprise(ENTERNAT) in accordance with RFC3204. At this time, it is preferable that the user-specific information be restricted to message information not exceeding 1.5 k bytes, including the basic header part. If another MIME body part is contained (for example, SDP), the PINX shall use a multi-part MIME.

The PINX shall incorporate a Content-Disposition header indicating "signal" and "handling=required" as a SIP header (for a single-part MME) or as the MIME header in a body part (for a multi-part MIME).

5.2.1.2 Compatibility between ends in user-specific information

If this user-specific information is to be sent, it is necessary to confirm in advance whether both PINXs accept user-specific information. As for this specification, steps up to negotiations depending on whether user-specific information is provided or not are not defined. If, however, there is a PINX that does not support user-specific information, the PINX may discard a request message in accordance with RFC3261. For example, if SIP UAS does not support user-specific information, and the end PINX does not contain the necessary information, the SIP response code 415 Unsupported Media Type is applied.

5.2.1.3 User-specific specification conditions for transit PINXs

As for the handling of user-specific information in transit PINX (including Proxy servers) in an operating agency, if media type information defined as application/Enterprise (ENTERNAT) is carried, it is necessary to provide a function to transmit it. If, however, there is a transit PINX that does not support user-specific information, the PINX may discard a request message in accordance with RFC3261. For example, if SIP UAS does not support user-specific information, and the end PINX does not contain the necessary information, the SIP response code 415 Unsupported Media Type is applied.

5.2.1.4 Conditions for the body of user-specific information

The contents of user-specific information is outside the scope of this standard.

5.2.1.5 Use in combination with tunneling information

For the use of user-specific information in combination with tunneling information, it is necessary to ensure compatibility with the remote PINX in advance. Basically, however, their media types are different so that operation is possible even if a combined use of services is allowed in the same link.

6. Detailed Headers in the SIP-Method

The following is examples of new headers in the Method.

Session Initiation Protocol
    Request-Line: INVITE sip:XXXXXXXX@domain.ne.jp;user=phone SIP/2.0
        Method: INVITE
        Resent Packet: False
    Message Header
        Via: SIP/2.0/UDP 100.50.20.170:5060;branch=z9hG4bK-a52f26-496736.761
        From: <sip:XXXXXXXX@domain;.ne.jp;user=phone>;tag=14748c89
        To: <sip:XXXXXXXX@domain.ne.jp;user=phone>
        Call-ID: 722df03a@100.50.20.170
        CSeq: 787025002 INVITE
        Contact: <sip:XXXXXXX@100.50.20.170:5060>
            Contact Binding: <sip:XXXXXXX@100.50.20.170:5060>
                URI: <sip:XXXXXXXX@100.50.20.170:5060>
                    SIP contact address: sip:XXXXXXXX@100.50.20.170:5060
        Max-Forwards: 55
        Supported:
        Session-Expires: 180
        P-Preferred-Identity: <tel:XXXXXXXXXX>
        Privacy: none
        Allow: INVITE, ACK, BYE, CANCEL, PRACK
        Content-Type: application/ENTERNAT
        Content-Length: 153
    Message body
        Enterprise Native Protocol
            Enterprise Native Protocol Version(v):1.0

The contents of the subsequent fields are not defined.

Supplementary explanation

Content-Type ENTERNAT shall conform to the applicable TTC standard (private SIP).

Appendix: International Standardization of User-Specific Information Specific to Japan

New media information (ENTERNAT) of Content-Type that is defined in Japan shall first be closed within the private SIP within TTC, and proposals to international standardization bodies shall be considered separately.

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