ATIS STANDARD

ATIS-1000679.2015(R2020)

Interworking between Session Initiation Protocol (SIP) and ISDN User Part

AMERICAN NATIONAL STANDARD FOR TELECOMMUNICATIONS

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ATIS-1000679.2015(R2020), *Interworking Between Session Initiation Protocol (SIP) and ISDN User Part*

Is an American National Standard developed by the **PSTN Transition (PSTN)** Subcommittee under the **ATIS Packet Technologies and Systems Committee (PTSC)**.

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American National Standard for Telecommunications

Interworking between Session Initiation Protocol (SIP) and ISDN User Part

Alliance for Telecommunications Industry Solutions

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American National Standards Institute, Inc.

Abstract

This Standard defines the signaling interworking between the ISDN User Part (ISUP) protocol and SIP in order to support services that can be commonly supported by ISUP and SIP based network domains. The title of this standard has been modified from ATIS-1000679.2004 to reflect the removal of interworking between SIP and Bearer Independent Call Control.

Foreword

The information contained in this foreword is not part of this American National Standard (ANS) and has not been processed in accordance with ANSI's requirements for an ANS. As such, Foreword may contain material that has not been subjected to public review or a consensus process. In addition, it does not contain requirements necessary for conformance to the standard.

The Alliance for Telecommunication Industry Solutions (ATIS) serves the public through improved understanding between providers, customers, and manufacturers. The Packet Technologies and Systems Committee (PTSC) develops and recommends standards and technical reports related to services, architectures, and signaling, in addition to related subjects under consideration in other North American and international standards bodies. PTSC coordinates and develops standards and technical reports relevant to telecommunications networks in the U.S., reviews and prepares contributions on such matters for submission to U.S. ITU-T and U.S. ITU-R Study Groups or other standards organizations, and reviews for acceptability or per contra the positions of other countries in related standards development and takes or recommends appropriate actions.

This document is entitled *Interworking between Session Initiation Protocol (SIP) and ISDN User Part.*

This standard is intended for use in conjunction with ATIS-1000113.2005(R2010), *Signalling System No. 7 (SS7) – Integrated Services Digital Network (ISDN) User Part.*

There are three normative and three informative annexes in this standard. Information contained in a normative annex forms an integral part of this standard. Information contained in an informative annex is not considered part of this standard, but is rather auxiliary to the standard. Similarly, footnotes are not officially part of this standard.

Significant differences from ATIS-1000679.2009 include:

1. Removal of interworking between SIP and Bearer Independent Call Control (ATIS-1000673).
2. Addition of the set of mappings between ISUP Cause Codes and SIP status codes.
3. Clarification of playing ringing tone from the O-IWU.
4. Handling of early media and cut-through of a media path.
5. Change in the population of a received ISUP Nature of Connection Indicators parameter. The parameter is only passed when SIP-I is used. Handling is changed from incrementing the satellite indicator to passing the satellite indicator unchanged.
6. Addition of mappings for the following ISUP parameters:
   a. Carrier Identification
   b. Carrier Selection Information
   c. Calling Party's Category
   d. Operator Services Information
   e. Originating Line Information
   f. Charge Number
   g. Jurisdiction Information.
7. Clarification of the handling of address presentation restriction information.
8. Revision of the handling of the Transit Network Selection parameter.
9. Specification of interworking in support of Emergency Telecommunications Service.
10. Extension of the through-connection procedures to include the receipt of an ISUP Optional Backward Call Indicators (OBCI) parameter with the User Network Interaction indicator field set to "user network interaction, cut through in both directions".
11. Addition of procedures for the handling of information related to the incoming ISUP trunk group.
12. Addition of normative Annex A describing interworking of ISDN CLIP/CLIR supplementary service to SIP networks.
13. Consistent documentation of the handling of the History-Info and Diversion headers when either or both are used.

Significant differences from ATIS-1000679.2013 include:

1. Clarification of Section 6.1.3.2, replacing "SIP Request URI" with "SIP P-Asserted Identity header field".
2. Clarification of sections A.26 and A.32 required by the fact that the MWI service and Voice Message Waiting Indication Control service, respectively, do not make use of ISUP.

3. Addition into several sections of normative Annex A of the rationale for discontinuing aspects of the service being discussed that are not supported across the IWU.
4. Interworking for the Generic Address parameter for supplementary user provided calling address is removed because this use of the parameter is not applicable for the ANSI ISUP national network.

Future control of this document will reside with the Packet Technologies and Systems Committee (PTSC). This control of additions to the specification, such as protocol evolution, new applications, and operational requirements, will permit compatibility among U.S. networks. Such additions will be incorporated in an orderly manner with due consideration to the ITU-T layered model principles, conventions, and functional boundaries.

Suggestions for improvement of this document are welcome. They should be sent to the Alliance for Telecommunications Industry Solutions, PTSC 1200 G Street NW, Suite 500, Washington, DC 20005.

At the time of consensus on this document, PTSC, which was responsible for its development, had the following leadership:

M. Dolly, PTSC Chair (AT&T)

Viqar Shaikh, PTSC Vice Chair (Applied Communication Sciences)

M. Dolly, PTSC PSTN Chair (AT&T)

Bryan Bethea, PTSC PSTN Vice Chair (Time Warner Cable)

The **PSTN** Subcommittee was responsible for the development of this document.

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Executive Summary

This Standard defines the signaling interworking between the Session Initiation Protocol (SIP) and the ISDN User Part (ISUP) protocol to support services that can be commonly supported by ISUP and SIP based network domains.

1 Scope

This Standard defines the signaling interworking between SIP, with its associated Session Description Protocol (SDP), and the ISDN User Part (ISUP) protocol at an Interworking Unit (IWU). The capabilities of SIP and SDP that are needed to interwork with ISUP are defined in Annex C of this Standard. SIP and SDP are defined by the IETF. ISUP is defined in accordance with ATIS-1000113.2005.

An IWU may be stand-alone or may be combined with an ISUP exchange. It is assumed in this Standard that the initial service requests must be forwarded and/or delivered via a trusted Adjacent SIP Node (ASN) within a SIP network domain. The ASN is viewed as a trusted network entity rather than untrusted user entity, and thus the interface between the IWU and the ASN is a Network-to-Network interface (NNI). Where SIP with Encapsulated ISUP (SIP-I) is used, it is assumed that the remote SIP User Agent can be trusted to receive the ISUP information and is able to process ISUP. Similarly, it is assumed that the ISUP information received from the remote UA can be trusted. Support for SIP interworking at a User-Network Interface (UNI) is not within the scope of this standard. Many security concerns arise if a PSTN/ISDN interconnects with a SIP network (via an IWU) where either some of these assumptions are not valid or the validity of these assumptions cannot be ascertained. In addition, because of the inherently open and distributed nature of IP networks, it should be assumed that PSTN/ISDNs could be susceptible to increased security risks through the interconnection with such networks. Therefore, to reduce such risk, it is highly desirable to follow strong security requirements and guidelines when PSTN/ISDNs are interconnected with SIP networks. RFC 3398 identifies some security issues for SIP-PSTN/ISDN interconnection. This standard takes into account some security aspects including some identified in RFC 3398. RFC 3261 describes various aspects of security for SIP headers and message bodies and various mechanisms to reduce security risks within the SIP network itself. This material should be used as the basis for developing detailed security requirements applicable to an IWU. Such requirements are outside the scope of this standard.

The services that can be supported through the use of the signaling interworking are limited to the services that are supported by SIP and ISUP based network domains. Services that are common to the SIP and ISUP network domains will interwork by using the function of an IWU. The IWU will also handle (through default origination or graceful termination) services or capabilities that do not interwork across domains.

The scope of this Standard is shown in Figure 1.1