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ATIS-1000679-2015(R2020)

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

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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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ATIS-1000679.2015(R2020)

[Revision of ATIS-1000679.2013]

American National Standard for Telecommunications

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

Alliance for Telecommunications Industry Solutions

Approved April 14, 2015

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-1000679.2005 include:

1. Removal of interworking between SIP and Parser 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-WLI.
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.

ATIS-1000679.2015(R2020)

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 addition 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, ATIS 1200 G Street NW, Suite 500, Washington, DC 20005.

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Table of Contents

Executive Summary	1
1 Scope	1
2 Normative References	2
3 Definitions	4
4 Abbreviations & Acronyms	5
4.1 General	5
4.2 ISUP Messages	6
4.3 ISUP Parameters & Values	6
5 Methodology	7
5.1 Conventions for Representation of the ISUP PDU	7
5.2 Conventions for Representation of SIP/SDP Information	8
5.3 General Principles	8
5.3.1 Identification of Call, Dialog & Call Control Association	9
5.3.2 Encapsulation of ISUP Information	9
5.4 ISUP Encapsulation - Detailed Procedures	9
5.4.1 Sending of ISUP Information to Adjacent SIP Nodes	9
5.4.2 Receipt of ISUP Information	10
5.4.3 Exclusions/Special Considerations	11
5.5 sip: & sips: URIs	13
6 Incoming Call Interworking from SIP to ISUP at an I-IWU	14
6.1 Sending of Initial Address Message (IAM)	15
6.1.1 INVITE Received Without an SDP Offer	15
6.1.2 INVITE Received With an SDP Offer	16
6.1.3 IAM Parameters & Derived Parameters	16
6.2 Sending of COT	29
6.3 Receipt of ACM	30
6.4 Receipt of CPG	31
6.5 Receipt of Answer Message (ANM)	32
6.6 Confusion Message	32
6.7 Receipt of Circuit (CIC) Query response Message	32
6.8 Receipt of EXM	32
6.9 Receipt of PAM	33
6.10 Through Connection of the Bearer Path	33
6.11 Receipt of Suspend Message (SUS) Network Initiated	33
6.12 Receipt of Resume Message (RES) Network Initiated	34
6.13 Release Procedures at the I-IWU	34
6.13.1 Receipt of BYE/CANCEL	34
6.13.2 Receipt of REL	34
6.13.3 Autonomous Release at I-IWU	37
6.13.4 Receipt of RSC, GRS or CGB	38
7 Outgoing Call Interworking from ISUP to SIP at O-IWU	39
7.1 Sending of INVITE	40
7.1.1 Coding of SDP Media Description Lines from USI	42
7.1.2 Request-URI and To header field	44
7.1.3 P-Asserted-Identity, From, & Privacy Header Fields	46
7.1.4 Hop Counter (Optional) [Max Forwards]	50
7.1.5 Coding of Encapsulated ISUP IAM Parameters in Outgoing INVITE (for SIP-I only)	50
7.1.6 Nature of Connection Indicators Parameter	51
7.1.7 Charge Number Parameter	51
7.1.8 Jurisdiction Information	51

7.1.9	Incoming Trunk Group	51
7.1.10	P-Early Media	51
7.1.11	Interworking for ETS	51
7.2	Receipt of 18X response	53
7.2.1	Receipt of 180 Ringing	53
7.2.2	Receipt of 183 Session Progress	55
7.2.3	Receipt of 182 Queued	55
7.3	Expiry of Timer T_{OIW2} & Sending of Early ACM	55
7.4	Receipt of Circuit (CIC) Query Response Message	56
7.5	Receipt of 200 OK INVITE	56
7.5.1	Setting of Backwards Call Indicators in the ANM message (for no ISUP encapsulation case only)	56
7.6	Through Connection of ISUP Bearer Path	56
7.6.1	Tone and Announcement (backward)	57
7.7	Release Procedures at the O-IWU	57
7.7.1	Receipt of Forward REL	57
7.7.2	Receipt of Backward BYE	58
7.7.3	Autonomous Release at O-IWU	58
7.7.4	Receipt of RSC, GRC, or CGB (ISUP)	59
7.7.5	Receipt of 3XX, 4XX, 5XX, 6XX Response INVITE	59
8	Timers	63
Annex A:	Interworking for ISDN & Non-ISDN Supplementary services	64
A.1	Interworking of ISDN CLIP/CLIR Supplementary Service to SIP Networks	64
A.1.1	Operation without Encapsulated ISUP	64
A.1.2	SIP-I	64
A.2	Interworking of COLP/COLR Supplementary Service to SIP Networks	64
A.3	Interworking of Direct-Dialing-In (DDI) Supplementary Service to SIP Networks	64
A.4	Interworking of Malicious Call Identification (MCID) Supplementary Service to SIP networks	65
A.5	Interworking of Sub-addressing (SUB) Supplementary Service to SIP Networks	65
A.6	Interworking of Non-ISDN Call Forwarding Busy (CFB)/ Call Forwarding No Reply (CFNR) / Call Forwarding Unconditional (CFU) Supplementary Services to SIP networks	65
A.6.1	General	65
A.6.2	History-Info Header Method	65
A.6.3	Diversion Header Method	76
A.7	Interworking of ISDN Call Deflection (CD) Supplementary Service to SIP Networks	81
A.8	Interworking of ISDN Explicit Call Transfer (ECT) Supplementary Service to SIP Networks	82
A.9	Interworking of ISDN Call Waiting (CW) Supplementary Service to SIP Networks	82
A.10	Interworking of ISDN Call Hold (HOLD) Supplementary Service to SIP Networks	82
A.11	Interworking of Completion of Calls to Busy Subscriber (CCBS) Supplementary Service to SIP networks	84
A.12	Interworking of Completion of Calls on No Reply (CCNR) Supplementary Service to SIP networks	84
A.13	Interworking of Terminal Portability (TP) Supplementary Service to SIP Networks	84
A.14	Interworking of ISDN Conference Calling (CONF) Supplementary Service to SIP Networks	84
A.15	Interworking of Three-Party Service (3PTY) Supplementary Service to SIP Networks	85
A.16	Interworking of Non-ISDN Multi-location Business Group (MBG) Supplementary Service to SIP Networks	85
A.17	Interworking of ISDN Multi-Level Precedence & Preemption (MLPP) Supplementary Service to SIP Networks	85
A.18	Interworking of Global Virtual Network Service (GVNS) Supplementary Service to SIP Networks	85
A.19	Interworking of International Telecommunication Charge Card (ITCC) Supplementary Service to SIP Networks	85
A.20	Interworking of Reverse Charging (REV) Supplementary Service to SIP Networks	85
A.21	Interworking of ISDN User-to-User Signalling (UUS) Supplementary Service to SIP Networks	85
A.22	Interworking of Emergency Calling Service to SIP Networks	86
A.23	Interworking of ISDN Normal Call Transfer Supplementary Service to SIP networks	86
A.24	Interworking of ISDN Calling Name Identification Restriction Supplementary Service to SIP Networks ..	86
A.25	Interworking of ISDN Calling Name Identification Presentation Supplementary Service to SIP Networks	86

A.26 Interworking of ISDN Message Waiting Indicator Control & Notification Supplementary Service to SIP Networks	86
A.27 Interworking of ISDN Call Park Supplementary Service to SIP Networks	86
A.28 Interworking of Non-ISDN Assist Supplementary Service to SIP Networks	87
A.29 Interworking of Non-ISDN Carrier Selection Supplementary Service to SIP Networks	87
A.30 Interworking of Non-ISDN Directory Assistance with Call Completion Supplementary Service to SIP Networks	87
A.31 Interworking of Non-ISDN User Network Interaction Supplementary Service to SIP Networks	87
A.32 Interworking of Non-ISDN Voice Message Waiting Indication Control Supplementary Service to SIP Networks	87
A.33 Interworking of Non-ISDN 950+Call Supplementary Service to SIP Networks.....	87
A.34 Interworking of Interworking with Non-ISDN Private Network Supplementary Service to SIP Networks .	87
Annex B: SIP/SIP-I References	88
B.1 SIP/SIP-I Signaling References and Profile	88
B.1.1 References.....	88
B.2 SIP/SIP-I Media References.....	88
B.2.1 References.....	88
Annex C: Interworking scenarios between SIP and ISUP.....	89
C.1 Scope	89
C.2 Definitions.....	89
C.3 Abbreviations.....	89
C.4 Methodology.....	90
C.5 Interworking of SIP Access to ISUP.....	90
C.5.1 Example Scenarios for Incoming Call Interworking from SIP to ISUP at I-IWU.....	90
C.5.2 Example scenarios for Outgoing Call Interworking from ISUP to SIP at O-IWU.....	92
Annex D: Interworking scenarios between SIP-I and ISUP.....	95
D.1 Scope	95
D.2 Definitions.....	95
D.3 Interworking of ISUP with SIP-I.....	95
D.3.1 Successful Call Set-up Procedures/Call Flow Diagrams for Basic Call Control.....	96
D.3.2 Unsuccessful Call Setup Procedures/Call Flow Diagrams for Basic Call Control.....	104
D.3.3 Release Procedures/Call Flow Diagrams for Basic Call Control.....	106
D.3.4 Suspend/Resume Procedures/Call Flow Diagrams for Basic Call Control.....	107
Annex E: Formal Syntax of the P-Charge-Info Header	109

Table of Figures

Figure 1.1 - Scope of Interworking between SIP and ISUP	2
Figure 5.1 -Overall Composite Signaling Message Flow	7
Figure 6.1 - IWU Composite Signaling Message Flow	15
Figure 7.1 - Example O-IWU Signaling Message Flow	40
Figure C.1 - Example of a call flow or "arrow" diagram.....	89
Figure C.2 - Successful Basic Call Set-Up from SIP to ISUP (SIP Preconditions and Continuity Check Protocol Used).....	90
Figure C.3 - Successful Basic Call Set-Up from SIP to ISUP (SIP Preconditions and Continuity Check Protocol Not Used)	91
Figure C.4 - Unsuccessful Basic Call Set-Up from SIP to ISUP	91
Figure C.5 - Normal Call Release from SIP to ISUP.....	92
Figure C.6 - Successful Basic Call Set-Up from ISUP to SIP (SIP Preconditions and Continuity Check Protocol Used).....	92
Figure C.7 - Successful Basic Call Set-Up from ISUP to SIP (SIP Preconditions and Continuity Check Protocol Not Used)	93
Figure C.8 - Unsuccessful Basic Call Set-Up from ISUP to SIP	93

Figure C.9 - Normal Call Release from ISUP to SIP	94
Figure D.1 - Example of a call flow or "arrow" diagram	95
Figure D.3 - En bloc, early ACM encapsulation	97
Figure D.5 - En bloc, early session description negotiation	99
Figure D.7 - En bloc, backward session description initiation	101
Figure D.9 - En bloc, segmented preconditions for resource reservation	103
Figure D.11 - Backward release during call setup	104
Figure D.13 - Forward release during call setup, early dialog is already established	106
Figure D.15 - Normal release with SUS message encapsulation	107

Table of Tables

Table 5.1 - ISUP Messages for Special Consideration	12
Table 6.1 - Interworked Contents of the Initial Address Message	17
Table 6.2 - Coding of the Calling Party Number	17
Table 6.3 - Coding of USI from SIP: SIP to ISUP	21
Table 6.4 - Mapping of SIP From/To-Asserted-Identity/Privacy headers to ISUP CLI parameters	23
Table 6.5 - Setting of the Network-provided ISUP Calling Party Number parameter with a CLI	24
Table 6.6 - Mapping of P-Asserted-Identity and Privacy Headers to the ISUP Calling Party Number Parameter	24
Table 6.7 - Mapping of SIP From Header Field to ISUP Calling party number parameter	25
Table 6.8 - Mapping of SIP Request-URI to ISUP Generic Address (ported number) parameter	26
Table 6.9 - Mapping from Max-Forwards to Hop Counter	26
Table 6.10 - Message sent to SIP upon receipt of ACM	31
Table 6.11 - Receipt of CPG at the I-IWU	32
Table 6.12 - 183 Session Progress sent to SIP upon receipt of EXM	33
Table 6.13 - INFO sent to SIP upon receipt of PAM	33
Table 6.14 - INFO sent to SIP upon receipt of SUS (SIP-I only)	33
Table 6.15 - of Resume Message (RES) network initiated (SIP-I only)	34
Table 6.16 - Mapping of SIP Reason header fields into Cause Indicators parameter	34
Table 6.17 - Coding of Cause Value if not taken from the Reason header field (except when encapsulated REL received)	34
Table 6.18 - Mapping of Cause Indicators parameter into SIP Reason header fields	35
Table 6.19 - Receipt of the Release message (REL)	36
Table 6.20 - Autonomous Release at I IWU	38
Table 6.21 - Receipt of RSC, GRS, or CGB messages (ISUP)	39
Table 7.1 - Interworked Contents of the INVITE message	41
Table 7.2 - Coding of SDP Media Description Lines from USI: ISUP to SIP	43
Table 7.3 - Mapping of Called party number and FCI Ported number translation indicator (when GAP for ported number is not included) to SIP Request-URI	45
Table 7.4 - Mapping of Generic address (ported) and Called party number (when both are included), and FCI Ported number to SIP Request-URI	45
Table 7.5 - Mapping of Carrier Identification parameter to SIP Request-URI	46
Table 7.6 - ISUP CLI Parameters to SIP Header fields	47
Table 7.7 - Void	48
Table 7.8 - Mapping of Calling Party Number parameter to SIP P-Asserted-Identity header fields	48
Table 7.9 - Mapping of ISUP Calling Party Number parameter to SIP From header fields	48
Table 7.10 - Mapping of ISUP APRIs into SIP Privacy header fields	49
Table 7.11 - Mapping from Hop Counter to Max-Forwards	50
Table 7.12 - Receipt of 18X Response	53
Table 7.13 - Release from SIP side at O IWU	58
Table 7.14 - Autonomous Release at O IWU	58
Table 7.15 - Receipt of RSC, GRS, or CGB messages (ISUP) at O-IWU	59
Table 7.16 - Receipt of 4XX, 5XX, or 6XX at O IWU	61
Table 8.1 - Interworking Timers	63
Table A.1 - Mapping of SIP messages to ISUP messages	66
Table A.2 - Mapping of hi-targeted-to-uri to ISUP Event Information	66
Table A.3 - Mapping of 181 (Call Is Being Forwarded) → ACM	66
Table A.4 - Mapping of 181 (Call Is Being Forwarded) → CPG if ACM was already sent	67

ATIS-1000679.2015(R2020)

Table A.5 - Mapping of 180 (Ringing) → ACM if no 181 (Call Is Being Forwarded) was received before..... 67

Table A.6 - Mapping of 180 (Ringing) → CPG if a 181 (Call Is Being Forwarded) was received before..... 67

Table A.7 - Mapping of IAM to SIP INVITE request 68

Table A.8 - Mapping of SIP to ISUP messages 70

Table A.9 - Mapping of History-Info header field to ISUP Redirecting number 71

Table A.10 - Mapping of History header to ISUP Redirection Information..... 72

Table A.11 - Mapping of History-Info header field to ISUP Original Called number 73

Table A.12 - Mapping of INVITE to IAM 74

Table A.13 - Mapping of ISUP to SIP Messages 74

Table A.14 - Mapping of ACM →181 (Call Is Being Forwarded) response..... 75

Table A.15 - Mapping of ACM →180 (Ringing) response 75

Table A.16 - Mapping of CPG →181 (Call Is Being Forwarded) response..... 76

Table A.17 - Mapping of CPG → 180 (Ringing) response 76

Table A.18 - Mapping of SIP messages to ISUP messages 77

Table A.19 - Mapping of IAM to SIP INVITE request 77

Table A.20 - Mapping of INVITE to IAM 80

Table A.21 - Mapping of ISUP to SIP Messages 81

Table A.22 - A mapping between ISUP and SIP for Call Hold supplementary service..... 83

Table A.23 - Mapping between ISUP and SIP-I for Call Hold supplementary service 84

American National Standard for Telecommunications –

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

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