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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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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-1000679.2004 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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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

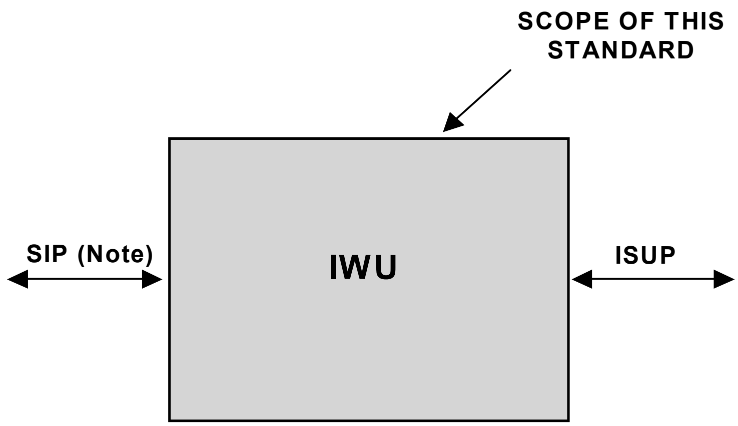

Figure 1.1 - Scope of Interworking between SIP and ISUP

NOTE – The content consists of the SIP headers and message body.

2 Normative References

The following standards and other references contain provisions, which through reference in this text, constitute provisions of this standard. At the time of publication, the editions indicated were valid. All standards and other references are subject to revision; all users of this standard are therefore encouraged to investigate the possibility of applying the most recent edition of the standards and other references listed below.

NOTE – All IETF Standards Track RFCs directly referenced by this Standard are listed in Annex C.

ATIS-1000113.2005(R2010), *Signaling System No. 7 (SS7) - Integrated Service Digital Network User Part (ISUP)*.[1]

ATIS-1000607.2014, *Digital Subscriber Signaling System No. 1 - Layer 3 Signaling Specification for Circuit Switched Bearer Service*.[2]

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ATIS-1000611.1991(R2013), *Signalling System Number 7 (SS7) - Supplementary Services for Non-ISDN Subscribers*.[4]

ATIS-1000613.1991(R2012), *Integrated Services Digital Network (ISDN) - Call Waiting Supplementary Service*.[5]

ATIS-1000616.2014, *Integrated Services Digital Network (ISDN) – Call Hold Supplementary Service*.[6]

ATIS-1000619.1992(R2010), Integrated Services Digital Network (ISDN) - Multi-Level Precedence and Preemption (MLPP) Service Capability. [7]

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ATIS-1000621.2014, *Integrated Services Digital Network (ISDN) – User-to-User Signaling Supplementary Service*. [9]

ATIS-1000622.1999(R2013), *Message Waiting Indicator Control and Notification Supplementary Services and Associated Switching and Signaling Specifications*. [10]

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ATIS-1000625.1993(R2013), *Integrated Services Digital Network (ISDN) - Calling Line Identification Presentation and Restriction Supplementary Services*. [12]

ATIS-1000625.a.1998(R2013), *Supplement to ATIS-1000625.1993(R2008) - Integrated Services Digital Network (ISDN) - Calling Line Identification Presentation and Restriction Supplementary Services, Application of Standard to Wireless PCS Applications*. [13]

ATIS-1000628.2000(R2010), *Emergency Calling Service*.[14]

ATIS-1000628.a.2001(R2010), *ECS-Connection and Ring Back Addendum [Supplement to ATIS-1000628.2000(R2010)]*.[15]

ATIS-1000632.1993(R2014), *ISDN Supplementary Service Normal Call Transfer*. [16]

ATIS-1000639.1995(R2011), *Calling Name Identification Restriction*.[17]

ATIS-1000639.a.2001(R2011), *Supplement to Calling Name Identification Restriction*.[18]

ATIS-1000641.2014, *Calling Name Identification Presentation*.[19]

ATIS-1000641.a.2002(R2012), *Supplement to Calling Name Identification Presentation*.[20]

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ATIS-1000643.1998(R2013), *Integrated Services Digital Network (ISDN) Explicit Call Transfer Supplementary Service*.[22]

ATIS-1000647.1995(R2010), *Integrated Services Digital Network (ISDN) Conference Calling Supplementary Service*.[23]

ATIS-1000647.a.1998(R2010), *Integrated Services Digital Network (ISDN) - Conference Calling Supplementary Service - Operations Across Multiple Interfaces*.[24]

ATIS-1000653.1996(R2010), *Integrated Services Digital Network (ISDN) Call Park Supplementary Service*.[25]

ATIS-1000653.a.1998(R2010), *Integrated Services Digital Network (ISDN) – Call Park Supplementary Service – Generic Procedures for the Control of ISDN Supplementary Services, Clarification for Number Identification*. [26]

ITU-T Recommendation Q.1912.5, *Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control Protocol or ISDN User Part*.[27]

ITU-T Recommendation Q.850, *Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part*.[27]

3GPP TS 24.229, *IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3*, clause 7.2A.12, CPC and OLI tel URI parameter definition.[28]

3 Definitions

For a list of common communications terms and definitions, please visit the *ATIS Telecom Glossary*, which is located at < http://www.atis.org/glossary >.

For ISUP specific terminology, refer to ATIS-1000113.2005, chapter 2. For SIP and SDP specific terminology, refer to RFC 3261 and RFC 2327, respectively. Definitions for additional terminology used in this interworking Standard are as follows:

3.1 Adjacent SIP Node (ASN): A SIP node (e.g., SIP Proxy or Back-to-Back User Agent or the SIP side of an IWU) that has established a direct trust relation (association) with Incoming or Outgoing IWU entities. The SIP Proxy and Back-to-Back User Agent are defined in accordance with RFC 3261.

3.2 Incoming Interworking Unit (I-IWU): This physical entity, which can be combined with an ISUP exchange, terminates incoming calls using SIP and originates outgoing calls using the ISUP protocol.

3.3 Incoming or Outgoing: In this Standard, indicates the direction of a call (not signaling information) with respect to a reference point.

3.4 Incoming SIP or ISUP [Network]: The network from which the incoming calls are received; uses the SIP or ISUP protocol. Without the term "network," simply refers to the protocol.

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[28] This document is available from the Third Generation Partnership Project (3GPP) at < http://www.3gpp.org/specs/specs.htm>.

3.5 Outgoing Interworking Unit (O-IWU): This physical entity, which can be combined with an ISUP exchange, terminates incoming calls using ISUP and originates outgoing calls using SIP.

3.6 Outgoing SIP or ISUP [Network]: The network, to which the outgoing calls are sent, uses SIP or ISUP protocol. Without the term "network," simply refers to the protocol.

3.7 SIP with Encapsulated ISUP (SIP-I): The use of SIP with a message body that encapsulates the ISUP information according to the requirements in this Standard.

3.8 SIP Precondition: Indicates the support of the SIP "precondition procedure" as defined in RFC 3312.

In addition, this Standard makes use of the terms *header field*, *message*, *message body*, *method*, *provisional response*, and *User Agent*, which are defined in RFC 3261, clause 6. It uses the term *payload type* as defined in RFC 3550, and *static* and *dynamic* payload type as defined in that RFC. Finally, it uses the terms *attribute* and *session* as defined in RFC 2327.

Within this document the following terminology is used:

- *Pass to ISUP procedures* describes an operation internal to the IWU; and
- *Send* describes the transmission of a message on the applicable external network interface.

4 Abbreviations & Acronyms

This document uses the following abbreviations and acronyms:

4.1 General

ABNF	Augmented Backus-Naur Form (see RFC 2234)
AMR	Adaptive Multirate (codec)
ASN	Adjacent SIP Node
ATM	Asynchronous Transfer Mode
B2BUA	Back-to-Back User Agent
BNC	Backbone Network Connection
CC	Country Code
CLI	Calling Line Identification
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
FFS	For further study
IANA	Internet Assigned Numbers Authority
IETF	Internet Engineering Task Force
I-IWU	Incoming (to ISUP) Interworking Unit
IPBCP	Internet Protocol Bearer Control Protocol
ISDN	Integrated Services Digital Network
ISN	Interface Serving Node
ISUP	ISDN User Part
IWU	Interworking Unit
MIME	Multi-purpose Internet Mail Extensions
NDC	National Destination Code
NNI	Network To Network Interface
NP	Number Portability
O-IWU	Outgoing (from ISUP) Interworking Unit
PSTN	Public Switched Telephone Network
PT	Payload Type

RFC Request For Comments
RTP Real-Time Transport Protocol
SCCP Signaling Connection Control Part
SDP Session Description Protocol
SIP Session Initiation Protocol
SIP-I SIP with encapsulated ISUP
SN Subscriber Number
UA User Agent
UAC User Agent Client
UAS User Agent Server
UNI User To Network Interface
URI Universal Resource Identifier

4.2 ISUP Messages

ACM Address Complete Message
ANM Answer Message
CGB Circuit Group Blocking
COT Continuity
CPG Call Progress
EXM Exit
GRS Circuit Group Reset
IAM Initial Address Message
PAM Pass Along Message
REL Release
RES Resume
RLC Release Complete
RSC Reset Circuit
SGM Segmentation Message
SUS Suspend

4.3 ISUP Parameters & Values

APRI Address Presentation Restricted Indicator
ATP Access Transport Parameter
BCI Backward Call Indicators
CgPN Calling Party Number
CIC Circuit Identification Code (ISUP)
FCI Forward Call Indicators
GAP Generic Address Parameter
HLC High Layer Compatibility
NOA Nature of Address indicator
NP *"network provided"* (Screening Indicator value)
NPDI NP Database Dip Indicator
USI User Service Information

5 Methodology

Figure 5.1 illustrates an example overall composite signaling message flow for the interworking between SIP and ISUP at the O-IWU and I-IWU. The example overall flow illustrates SIP preconditions and SIP with encapsulated ISUP.

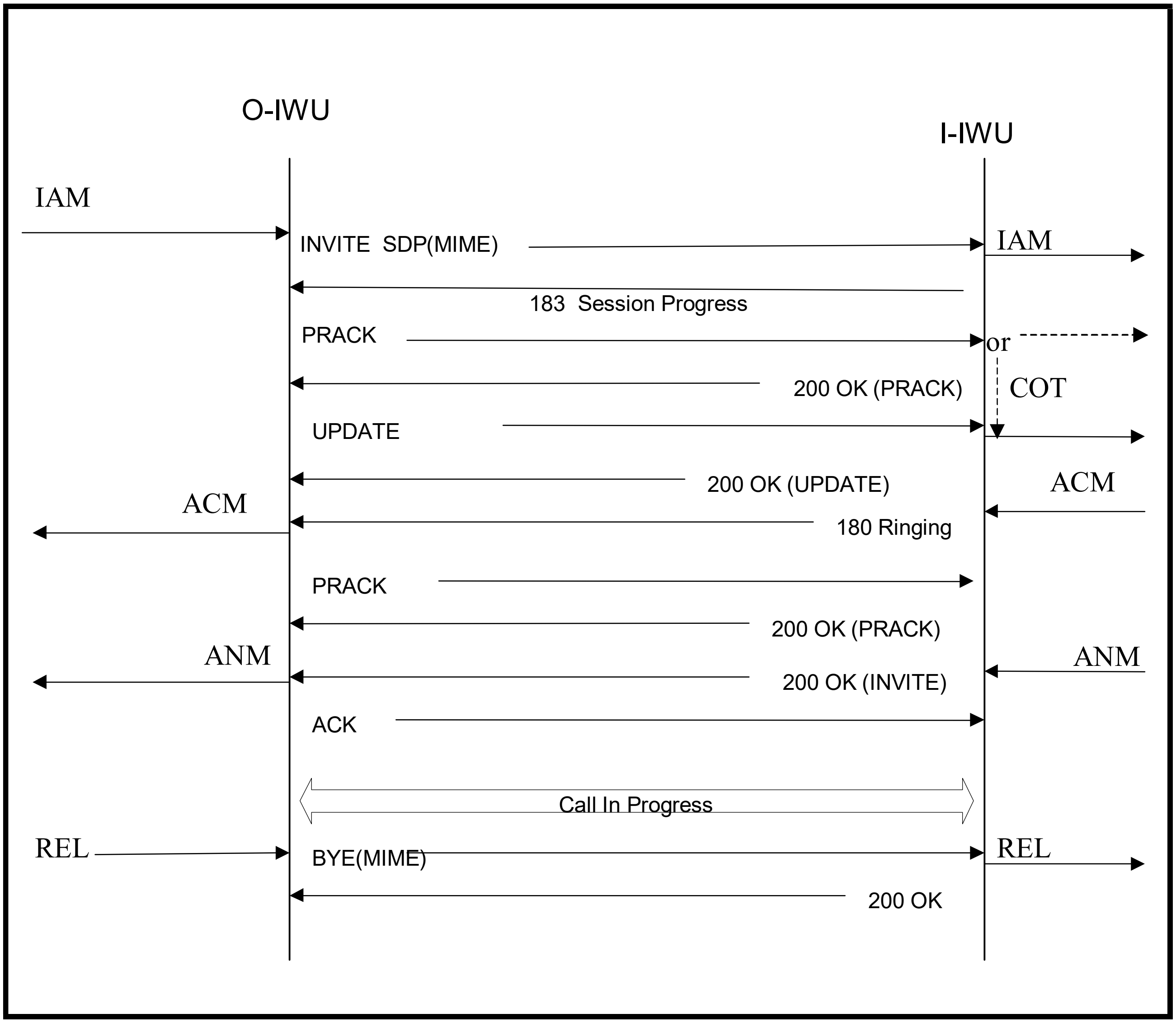


Figure 5.1 -Overall Composite Signaling Message Flow

5.1 Conventions for Representation of the ISUP PDU

1. The first letter of a name for each signaling information element for the following classes of terms is capitalized:
   - Indicators;
   - Parameters;
   - Information elements and
   - Messages.

   Examples: Called Party Number parameter, Initial Address Message.
2. The definition of a parameter value is written in *italics* and is inserted between quotation marks.

Example: Nature of Address value 0000011 – *"national (significant) number."*

5.2 Conventions for Representation of SIP/SDP Information

1. All letters of SIP method names are capitalized.

   Examples: INVITE, INFO.

2. SIP header fields are identified by the unabbreviated header field name as defined in the relevant RFC, including capitalization and enclosed hyphens but excluding the following colon.

   Examples: To, From, Call-ID.

3. Where it is necessary to refer with finer granularity to components of a SIP message, the component concerned is identified by the ABNF rule name used to designate it in the defining RFC (generally from clause 25 of RFC 3261), in plain text without surrounding angle brackets.

   Examples: Request-URI, the userinfo portion of a sip: URI.

4. URI schemes are represented by the lower-case identifier followed by a colon and the abbreviation "URI."

   Examples: sip: URI, tel: URI.

5. SIP provisional and final responses other than 2XX are represented by the status code followed by the normal reason phrase for that status code, with initial letters capitalized.

   Examples: 100 Trying, 484 Address Incomplete.

6. Because of potential ambiguity within a call flow about which request a 200 OK final response answers, 200 OK is always followed by the method name of the request.

   Examples: 200 OK INVITE, 200 OK PRACK.

7. A particular line of an SDP session description is identified by the two initial characters of the line – that is, the line type character followed by "="

   Examples: m=line, a=line.

8. Where it is necessary to refer with finer granularity to components of a session description, the component concerned is identified by its rule name in the ABNF description of the SDP line concerned, delimited with angle brackets.

   Examples: the <media> and <fmt> components of the m= line.

5.3 General Principles

At the SIP interface, the IWU shall act as a UA and shall support the standards noted in Annex C. Interworking with forking in the SIP network is not specified in this document and is for further study. The ISUP interface shall support the protocol as defined in ATIS-1000113.2005.

The following rules apply to the handling of unrecognized ISUP information:

1. For the case in which no ISUP encapsulation is used, the IWU shall act as a Type A exchange for the purposes of ISUP Compatibility procedures.
2. For SIP-I, for the mapping of ISUP to and from SIP header fields and SDP, the IWU behaves as a Type A exchange. However, when handling ISUP information before encapsulating it or after it has been de-encapsulated, the IWU can act as a Type A or Type B exchange depending on the role (e.g., gateway between operators, transit) the IWU is performing for that particular call.
3. Only the procedures, methods, and elements of information (messages, parameters, indicators, headers, etc.) relevant to interworking are described. Therefore, the procedures, methods, and elements of information that are of local significance (i.e., only relevant to one of the signaling systems: SIP or ISUP), are outside the scope of this Standard, as they cannot be interworked.
4. The IWU combined with an ISUP exchange shall provide interworking between the bearer network connections on the SIP and ISUP network domain sides.
5. Before sending any information on the SIP side, the IWU shall consult its local trust policy to determine if the subsequent node to which the outgoing SIP message is directed is trusted to receive that information. Upon determination that Adjacent SIP Node (ASN) is not trusted to receive that information the IWU shall take appropriate action (e.g., omit the information, provide another value, or release the call) based on local policy.

6. Similarly, before accepting any information on the SIP side, the IWU shall consult its local trust policy to determine if the node from which the incoming SIP message came is trusted to originate or pass on that information. Upon determination that the adjacent SIP node (ASN) is not trusted to provide that information, the IWU shall take appropriate action (e.g., ignore the information, use a default value, or release the call) based on local policy.

5.3.1 Identification of Call, Dialog & Call Control Association

The IWU shall establish a one-to-one relationship between a SIP Dialog and an ISUP call/bearer control instance so that interworked signaling information is associated with the same interworked call/bearer.

5.3.2 Encapsulation of ISUP Information

The following general principles of ISUP encapsulation apply within this Standard:

a) *An IWU receiving a SIP message shall remove the ISUP body from the SIP message.* Any differences between the SIP message (e.g., header fields and SDP) and the ISUP message shall be resolved as defined by the procedures within this document. In all cases, the resultant ISUP information shall be passed to the relevant ISUP procedures.

b) *An IWU receiving an ISUP message shall, if appropriate, encapsulate the ISUP message within the body of the SIP message.* There are some exclusions as to which ISUP messages may be encapsulated within a SIP message. Clause 5.4 gives details of ISUP encapsulation procedures. These detailed procedures include a list of ISUP messages that are not encapsulated within SIP.

In all cases whereby the IWU inspects a SIP message and discovers that there is no encapsulated ISUP, the IWU is required to construct an appropriate ISUP message using the information received within the SIP header fields and SDP body (if present). Clauses 6 and 7 of this Standard provide all the information that the IWU requires to be able to perform this task.

NOTE – The interworking specifications in clauses 6 and 7 are of use whether or not the received SIP message contained encapsulated ISUP. In the case that the received SIP message contains encapsulated ISUP information, it provides any necessary interworking between SIP headers and the relevant ISUP parameters, thus enabling encapsulated ISUP information to be modified/updated (if appropriate) prior to ISUP procedures being applied as detailed in a) above. In the case that the received SIP message does not contain any encapsulated ISUP, it provides the means for the IWU to construct the appropriate ISUP messages purely based on the SIP header (and SDP body) information available.

5.4 ISUP Encapsulation - Detailed Procedures

This clause is relevant only when the ISUP information is encapsulated (or is expected to be encapsulated) in SIP messages sent or received on the SIP interface of an IWU. This clause builds on the general principles of ISUP encapsulation outlined in 5.3.2.

5.4.1 Sending of ISUP Information to Adjacent SIP Nodes

5.4.1.1 Introduction

The O-IWU shall apply any interworking procedures detailed within clause 7 affecting parameters within the ISUP, and then proceed to encapsulate any ISUP information received (with the exception of the excluded messages detailed in 5.4.3) in a relevant SIP message (see 5.4.1.3). Setting of header fields relating to the handling of the ISUP body is specified in 5.4.1.2.

Similarly, an I-IWU receiving ISUP information which is not excluded from encapsulation (see 5.4.3) shall apply any interworking procedures detailed in clause 6 affecting the ISUP and then encapsulate the ISUP output in a relevant SIP message (see 5.4.1.3). Setting of header fields relating to the handling of the ISUP body is specified in 5.4.1.2.

5.4.1.2 Header Fields for ISUP MIME Bodies

For the purpose of this specification the "Content-Type" header field associated with the ISUP MIME body shall be supplied as follows:

```
Content-Type: application/ISUP; version= ansi00
```

NOTE – "ansi00" refers to ISUP as defined in ATIS-1000113.2005. However, no action is taken at the IWU based on the "version" parameter.

The Content-Disposition header field associated with the ISUP MIME body shall be set as follows:

```
Content-Disposition: signal; handling = required
```

5.4.1.3 Determination of Trust Policy

Prior to sending out the INVITE message with encapsulated ISUP, the O-IWU shall consult its trust policy to determine whether the ISUP information can be passed to the ASN. If the ASN is trusted to receive ISUP information, the O-IWU shall apply any interworking procedures detailed within clause 7 affecting parameters within the ISUP, and then proceed to encapsulate any ISUP information received (with the exception of the excluded messages detailed in 5.4.3) in a relevant SIP message (see 5.4.1.3). Setting of header fields relating to the handling of the ISUP body is specified in 5.4.1.2.

Similarly, an I-IWU receiving ISUP information which is not excluded from encapsulation (see 5.4.3) shall determine if the node to which the SIP message is being sent is trusted to receive ISUP information. If so, it shall apply any interworking procedures detailed in clause 6 affecting the ISUP, and then encapsulate the ISUP output in a relevant SIP message (see 5.4.1.3). Setting of header fields relating to the handling of the ISUP body is specified in 5.4.1.2.

5.4.1.4 Determination of Which SIP Message to Use to Encapsulate the ISUP Message

For basic call setup, the SIP message used to encapsulate the ISUP message is the SIP message that was first triggered to be sent from the IWU, as a result of the interworking specified within the main body of this Standard and any ISUP specific annexes.

As an example, this means that an ISUP IAM received in clause 7.1 (B) will be encapsulated within the INVITE message that is sent out immediately from the O-IWU.

For other messages, see 5.4.3.

5.4.2 Receipt of ISUP Information

5.4.2.1 De-encapsulation of ISUP Information

On receipt of a SIP message containing encapsulated ISUP, the IWU shall de-encapsulate the ISUP message from the SIP message body. The ISUP message then goes through a number of stages of additional processing before being sent into the ISUP network. This processing is specified in 5.4.2.

5.4.2.1.1 Alignment of SIP Headers and ISUP Body Contents

On receipt of a SIP message containing encapsulated ISUP and prior to following the procedures of 5.4.2.1.3, the IWU shall use the procedures outlined in this Standard for interworking from SIP information to ISUP parameters to align any parameters in the ISUP message that are in conflict with SIP header fields (e.g., due to service invocation within the SIP network) when the SIP message reaches the I-IWU. The alignment rules regarding which header overrides which ISUP parameter and vice versa will depend on application/service related aspects.

Where a default value is defined to be set in the subclauses of clauses 6 and 7, this shall apply to the cases of no ISUP encapsulation as described. For SIP-I, the ISUP field shall be derived from the encapsulated ISUP mime body.

Where a SIP header mapping to ISUP field(s) is defined (for example, the mapping of Request-URI to Called Party Number in 6.1.3.1), then the SIP header should be given precedence over the encapsulated ISUP value in the alignment process unless otherwise stated.

For example, on receipt of an encapsulated IAM, the procedures in clause 6 relating to the population of the Called party number from the Request-URI would be used to re-write the Called Party number parameters/fields in the de-encapsulated ISUP message.

Once all alignment of these ISUP parameters (and any related fields) has completed, the I-IWU shall then pass the resulting ISUP message onto the ISUP procedures (see 5.4.2.1.3).

5.4.2.1.2 Setting of ISUP Parameters by IWU

Prior to following the procedures outlined in 5.4.2.1.3, the IWU will follow any procedures outlined within clause 6 (for the I-IWU) or clause 7 (in the case of the O-IWU), with respect to setting any parameters in the de-encapsulated ISUP message that are required to be autonomously set by the IWU in order to facilitate the interworking.

For example, in a call from SIP to ISUP where the SIP network is using preconditions, the "Continuity" indicator is set by the I-IWU to withhold session set up in the ISUP network until preconditions are met in the SIP network.

5.4.2.1.3 Passing Resulting ISUP Message to ISUP Procedures/Sending of ISUP Message

On receipt of an ISUP message resulting from the actions taken in 5.4.2.1.1 and 5.4.2.1.2, the ISUP message shall be passed to the relevant ISUP procedures. The ISUP message (if any) that results from this step is the message which is sent on the outgoing ISUP interface.

5.4.3 Exclusions/Special Considerations

The ISUP messages listed in Table 5.1 are either not encapsulated within any SIP message or receive special treatment with regards to ISUP encapsulation.

Table 5.1 - ISUP Messages for Special Consideration

ISUP message	Reference
Reset Circuit	5.4.3.1 (Note 1)
Circuit Group Blocking	5.4.3.1
Circuit Group Blocking Acknowledgement	5.4.3.1
Release Complete	5.4.3.5
Group Reset	5.4.3.1
Circuit Group Reset Acknowledgement	5.4.3.1
Confusion	5.4.3.1 or 5.4.3.2 (Note 2)
Forward Transfer	5.4.3.2
Suspend	5.4.3.2
Resume	5.4.3.2
Blocking	5.4.3.1
Blocking Acknowledgement	5.4.3.1
Continuity Check Request	5.4.3.1
Continuity	5.4.3.1
Unblocking	5.4.3.1
Unblocking Acknowledgement	5.4.3.1
Circuit Group Unblocking	5.4.3.1
Circuit Group Unblocking Acknowledgement	5.4.3.1
Circuit Query	5.4.3.1
Circuit Query Response	5.4.3.1
Circuit Reservation	5.4.3.1
Circuit Reservation Acknowledgement	5.4.3.1
Circuit Validation Response	5.4.3.1
Circuit Validation Test	5.4.3.1
Exit	5.4.3.4
Information	5.4.3.2
Information Request	5.4.3.2
Loop Back Acknowledgement	5.4.3.1
Pass Along	5.4.3.2
Facility	5.4.3.2
Segmentation	5.4.3.3
Application Transport	5.4.3.2
Pre-Release information	5.4.3.2
Unequipped Circuit Identification Code	5.4.3.1
NOTE – 1 – Where the ISUP procedures would send reset circuit (RSC) to an ISUP exchange, the IWU shall send an encapsulated REL with release cause 31 (Normal, unspecified). 2 – These messages are either locally terminated or sent transparently depending on whether they are destined for the IWU or for another exchange.	

5.4.3.1 ISUP Side Procedures Only

These messages are not encapsulated within SIP messages since they relate to procedures that are relevant only for the ISUP side of the call. Typically, these messages are related to maintenance of ISUP circuits. If these ISUP messages are received encapsulated within SIP messages, the ISUP information shall be discarded.

5.4.3.2 Transparent Messages

In these cases, the ISUP message is transported through the SIP network encapsulated in the following SIP messages:

a) *183 Session Progress* provisional response, if this is sent by the I-IWU in the backward direction before a confirmed dialog is established.
b) *INFO message* in all other cases.

These messages are deemed important to transport transparently in order to maintain end-to-end service.

5.4.3.3 ISUP Segmentation and ISUP Encapsulation

The Segmentation message itself is not encapsulated within SIP. Instead, the IWU (ISUP side interface) will re-assemble the original message with its segmented part and check the Optional Forward Call Indicators or Optional Backward Call Indicators parameter.

The actions taken by the IWU on the Optional Forward Call Indicators or Optional Backward Call Indicators depend on whether the simple segmentation indicator is the only indicator to be set in the parameter.

If no other indicator is set within the Optional Forward Call Indicators or Optional Backward Call Indicators parameter, the entire parameter is discarded.

If another indicator is set within the Optional Forward Call Indicators or Optional Backward Call Indicators parameter, the IWU shall set the Simple Segmentation Indicator to indicate that no additional information will be sent.

The IWU shall then encapsulate the resulting message within the SIP message body.

5.4.3.4 Handling of Exit Message

If the I-IWU behaves as a gateway exchange, the actions taken on the ISUP side for the setting of the timer ($T_{EXM,d}$) are as described in ATIS-1000113.2005, chapter 4, 2.1.3A.1. At the SIP side, the EXM is encapsulated in the MIME body of either a 183 Session Progress (if this is the first backward 183 SIP message) or an INFO request (if a 183 message had previously been sent) at the expiration of this timer by the I-IWU.

5.4.3.5 Encapsulation of RLC

If a BYE is received containing an encapsulated REL, the 200 OK BYE sent in response shall encapsulate the RLC generated by ISUP procedures.

5.5 sip: & sips: URIs

Wherever this document makes reference to a sip: URI as defined in RFC 3261, the text applies equally to sips: URIs. The difference between the two URI types is of significance only in the SIP network, and does not affect interworking.

6 Incoming Call Interworking from SIP to ISUP at an I-IWU

An Incoming Interworking Unit (I-IWU) entity is used to transport calls originated from a SIP network domain to an ISUP network domain.

The “incoming SIP” refers to the SIP protocol, which is used between the Incoming IWU and the call originating entity (entities) supported in the SIP network domain. Similarly, the “outgoing ISUP” refers to the ISUP protocol supported between the I-IWU and the next-hop entity (entities) in the ISUP network domain.

The I-IWU receives forward and backward signaling information from the incoming SIP and outgoing ISUP sides, respectively. After receiving this signaling information and performing appropriate call/service processing, the I-IWU may signal forward to subsequent ISUP nodes or backward to preceding SIP entities. This clause specifies the signaling interworking requirements for basic call at the I-IWU. This clause is split into sub-clauses based upon the messages sent or received on the outgoing ISUP interface of the I-IWU. Only messages that are generated as a result of interworking to/from the incoming SIP side of the I-IWU are considered in this interworking.

The scope of this clause is based on the key assumptions:

- The I-IWU supports originating basic calls only; and
- Calls originated from the SIP network domain do not require equivalent PSTN/ISDN service interworking.

The service annexes of this document will cover additional interworking specification related to specific PSTN/ISDN services.

The I-IWU shall include a To tag in the first non-100 provisional response in order to establish a dialog as described in section 12 of RFC 3261.

For the SIP-I case, ISUP message segmentation must be handled as described in 5.4.3.3.

Figure 6.1 illustrates an example composite signaling message flow for the interworking between SIP and ISUP at the I-IWU. The composite flow illustrates SIP preconditions and SIP with encapsulated ISUP.

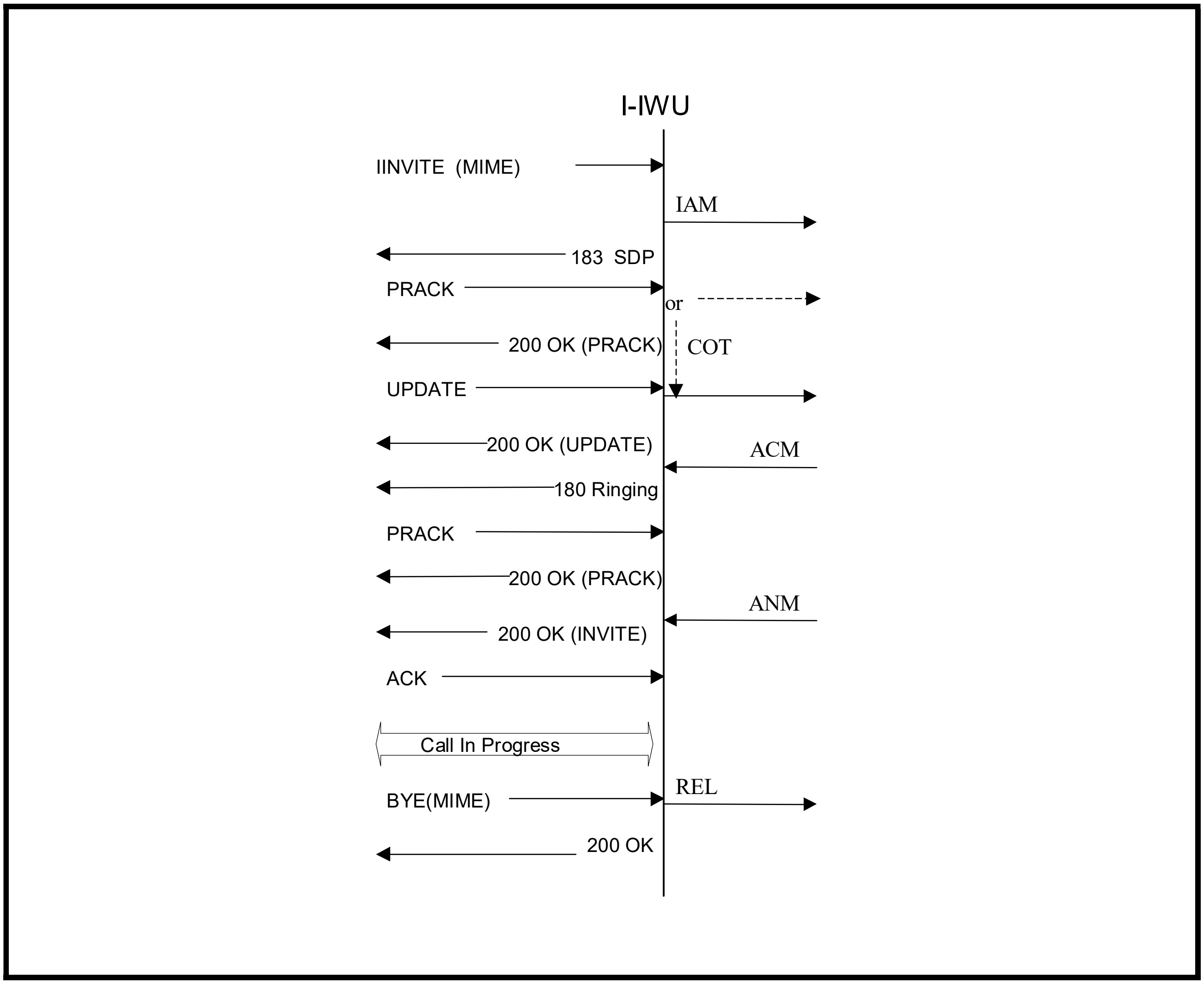


Figure 6.1 - IWU Composite Signaling Message Flow

6.1 Sending of Initial Address Message (IAM)

If an INVITE is received which has enough digits to route to the ISUP network and which cannot be associated with an existing call, the IAM resulting from the "receipt of INVITE" interworking procedures (see 6.1.1 and 6.1.2), or -- in the case of SIP-I -- the de-encapsulated IAM (as updated by the SIP-ISUP interworking procedures within 6.1.3 and associated subclauses) shall be passed to ISUP procedures.

NOTE – If an INVITE is received which does not have enough digits to route to the ISUP network, normal SIP procedures apply and the INVITE is not interworked.

Clauses 6.1.1 and 6.1.2 address the receipt of the INVITE for which an IAM is sent. The procedures for sending of the IAM message depend on whether the INVITE received from the SIP network contains an SDP Offer. See 6.1.1 and 6.1.2.

The IAM parameters are coded according to 6.1.3.

6.1.1 INVITE Received Without an SDP Offer.

Upon receipt of the INVITE without an SDP offer, the I-IWU shall determine if the received INVITE indicates support for reliable provisional responses.

1) *If reliable provisional responses are supported*, the I-IWU shall immediately send an SDP Offer including media description within a 183 Session Progress message.

(a) *If SIP preconditions are not in use*, the I-IWU shall send the IAM upon receipt of the SDP Answer with media description.

(b) *If SIP preconditions are in use*, the I-IWU will send the IAM by continuing on to procedure in 6.1.2 (2) below.

2) *If reliable provisional responses are not supported*, the I-IWU shall immediately send out the IAM.

6.1.2 INVITE Received With an SDP Offer

Upon receipt of an INVITE with an SDP offer:

1) *If SIP preconditions are not in use*, the IAM shall be sent immediately.
2) *If SIP preconditions are in use*, then:
   (a) *If outgoing ISUP signaling on subsequent network supports the use of the continuity check procedure*, the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator in the Nature of Connection Indicators parameter set to "continuity check performed on previous circuit," or "continuity check required on this circuit." The latter setting shall be used if the continuity check is to be performed on the outgoing circuit.
   (b) *If outgoing ISUP signaling on subsequent network does not support the use of the continuity check procedure*, sending of the IAM shall be deferred until all preconditions have been met.

In all cases, 6.1.3 gives specific details related to the population of specific parameters of the IAM. Table 6.1 below gives a summary of parameters within the IAM message that are interworked from the INVITE message, along with a reference to the subclauses of 6.1.3, in which the specific interworking is described.

6.1.3 IAM Parameters & Derived Parameters

Table 6.1 indicates the IAM parameters that interwork from SIP.

Table 6.1 - Interworked Contents of the Initial Address Message

Parameter	Section
Called Party Number	6.1.3.1
Calling Party's Category	6.1.3.2
Nature of Connection indicators	6.1.3.3
Forward Call Indicators	6.1.3.4
User service information	6.1.3.5
Calling Party Number	6.1.3.6.1
Generic Address	6.1.3.6.2
Hop Counter	6.1.3.7
Transit Network Selection	6.1.3.8
Carrier Identification	6.1.3.9
Carrier Selection Information	6.1.3.10
Operator Services Information	6.1.3.11
Originating Line Information	6.1.3.12
Charge Number	6.1.3.13
Jurisdiction Information	6.1.3.14
Generic Digits	6.1.3.15
NOTE – • Whether the precedence parameter is included in the IAM, and -- if included -- its coding is dependent on the information received in the INVITE. This is for further study. • The charge number parameter may be included in the IAM as a network option when ISUP encapsulation is not used.	

6.1.3.1 Called Party Number (Mandatory)

In the Request-URI, a sip: URI with the user=phone parameter and the userinfo part of that URI is an E.164 number encoded as specified by the telephone-subscriber rule of RFC 2806 is required. Support of any other URI schemes in the Request-URI is for further study.

The information contained in the userinfo component of the Request-URI shall be mapped to the called party number parameter of the IAM message. Table 6.2 summarizes this mapping.

If the routing number field is not present in the userinfo component of the Request-URI, the geographical telephone number field contained in the userinfo component of the Request-URI shall be mapped to the Called Party Number parameter of the IAM.

If the routing number field is present in the userinfo component of the Request-URI, the routing number field contained in the userinfo component of the Request-URI shall be mapped to the Called party number parameter of the IAM. For the mapping of the geographical telephone number field contained in the userinfo component of the Request-URI to the Generic Address Parameter of the IAM, refer to clause 6.1.3.6.

Table 6.2 - Coding of the Called Party Number

INVITE→	IAM→	Conditions
Request-URI	**Called Party Number**	
Geographical number in Userinfo	Address Signal	If routing number parameter is not present in the Userinfo
Routing number in Userinfo (following "rn=")	Address Signal	If routing number parameter is present in the Userinfo

If the Contact header in the incoming INVITE includes a “trgp=” parameter, the I-IWU may, depending on the value of the parameter, conclude that the calling party dialed an initial “0” digit or that an operator service is being invoked without an initial dialed “0” digit. In this case, the I-IWU will indicate this by either:

a) Including an ISUP Operator Services Indicator parameter with Information Type encoded as 001 (original access prefix) and Information Value encoded as appropriate from Table 6.2a.

Table 6.2a

ISUP Operator Services Information parameter, Information Value field value	Interpretation	Example Dialing Pattern
0 0 0 1	1+ or 011+	1+10 digits (e.g., from a network-controlled coin phone)
0 0 1 0	0+ or 01+	0+10 digits
0 0 1 1	0-	0

b) Populating the ISUP Called Party Number Nature of Address field as appropriate from Table 6.2b.

Table 6.2b

ISUP Called Party Number Nature of Address field value	Interpretation	Example Dialing Pattern
1 1 1 0 0 0 1	subscriber number, operator requested	1+10 digits (e.g., from a network-controlled coin phone)
1 1 1 0 0 1 0	national number, operator requested	0+NPA-555-1212
1 1 1 0 0 1 1	international number, operator requested	00+10 digits
1 1 1 0 1 0 0	no number present, operator requested	0

In addition, the I-IWU may select the outgoing trunk group based on this information.

6.1.3.2 Calling Party’s Category (Mandatory)

If the Userinfo component of the SIP P-Asserted-Identity header field contains a “cpc=” field, and if the I-IWU needs to send the ISUP Calling Party’s Category (CPC) parameter in the outgoing IAM, then the CPC will be populated as shown in Table 6.2c.

Table 6.2c

SIP cpc field value (note 2)	**ISUP CPC value**	**Interpretation**
"ordinary"	0 0 0 0 1 0 1 0	ordinary calling subscriber
"test"	0 0 0 0 1 1 0 1	test call
"operator"	0 0 0 0 1 0 0 1	national operator
"payphone"	0 0 0 0 0 0 0 0	calling party's category unknown (note 1)
"unknown"	0 0 0 0 0 0 0 0	calling party's category unknown
"mobile-hplmn"	0 0 0 0 0 1 0 1	calling party's category unknown (note 1)
"mobile-vplmn"	0 0 0 0 0 1 1 0	calling party's category unknown (note 1)
"emergency"	1 1 1 0 0 0 0 0	emergency service call
Not provided or unknown value	0 0 0 0 0 0 0 0	calling party's category unknown

NOTE 1 – The cpc field values in this Table follow those given in [TS24.229]. Since there are no procedures specified for U.S. networks for a CPC value of “payphone”, this value is mapped to “calling party's category unknown.”

NOTE 2 – Interworking of received Resource-Priority SIP header for ETS service will override any received SIP "cpc' parameter.

For SIP with encapsulated ISUP (SIP-I), the Calling Party's Category value shall be generated from the Calling Party's Category parameter present in the encapsulated ISUP.

6.1.3.3 Nature of Connection Indicators (Mandatory)

The indicators of the Nature of connection parameters, which are set by the IWU, are as follows:

Table 6.2d

Bits	Indicators in Nature of connection parameter
AB	Satellite indicator
DC	Continuity check indicator (ISUP)
E	Outgoing echo control device

Other Nature of Connection indicators should follow the current ISUP standard.

The following codes should be set by the I-IWU as default in the nature of connection indicators parameter field:

Table 6.2e

Bits	Codes	Meaning	Conditions
AB			
	01	One satellite circuit in the connection	SIP without ISUP encapsulation only
DC (Note)	00	Continuity check not required (ISUP)	Without pending precondition request (all profiles).
	10	Continuity check performed on a previous circuit (ISUP)	With pending precondition request (all profiles).
E	1	Outgoing echo control device included	SIP without ISUP encapsulation only
NOTE – In applying these values, the I-IWU shall ignore the Continuity setting received in an encapsulated IAM. COT is not encapsulated; the I-IWU creates COT as required. See 6.3.			

For SIP with encapsulated ISUP(SIP-I), with the exception of Continuity Check Indicator , which receives a special treatment in clause 6, the Nature of Connection Indicators should be generated by the I-IWU using the Nature of Connection Indicators received in the encapsulated IAM message.

6.1.3.4 Forward Call Indicators (Mandatory)

The indicators of the FCI parameters, which are set by the IWU, are as follows:

Table 6.2f

Bits	Indicators in FCI parameter
D	Interworking indicator
F	ISUP indicator
HG	ISUP preference indicator
I	ISDN access indicator
M	Ported number translation indicator

Other FCI indicators should follow the current ISUP standard.

The appropriate values of the FCI are determined based on analysis of various parameters (signaling, internal states or configurations) at the IWU.

Below are the default values for the following indicators should be set by the I-IWU as default in the FCI parameter:

Table 6.2g

Bits	Codes	Meaning
D	1	Interworking encountered.
F	0	ISDN user part not used all the way.
HG	01	ISDN user part not required all the way
I	0	Originating access non-ISDN

The value of the M bit is set depending on whether an NP Database Dip Indicator (npdi) parameter is present in the userinfo component of the Request-URI, as shown below:

Table 6.2h

Bits	Codes	Meaning	Conditions
M	0	Number not translated	NP dip not performed (npdi not present)
M	1	Number translated	NP dip performed (npdi present)

For the SIP-I case, the Forward Call Indicators shall be generated by the I-IWU using the Forward Call Indicators present within the received encapsulated ISUP message.

6.1.3.5 User Service Information (Mandatory) & Higher Layer Compatibility Information Element Within Access Transport Parameter (optional)

As a network option, when ISUP encapsulation is not used, either:

1) The USI parameter is set to 3.1 kHz audio and transcoding is applied when required (e.g., for 3GPP networks); or
2) If SDP is received from the remote peer before the IAM is sent and if transcoding is not supported at the I-IWU, then the User Service Information (USI) parameter shall be derived from SDP as described in 6.1.3.5.1. Otherwise they shall be set in accordance with local policy.

For SIP-I, the USI and HLC shall be taken from the encapsulated ISUP.

6.1.3.5.1 Transcoding Not Available at the I-IWU (Network Option)

The SDP Media Description Part received by the I-IWU should indicate only one media stream.

Only the “m=”, “b=”, and “a=” lines of the SDP Media Description Part are considered to interwork with the IAM USI and HLC parameters.

The first sub-field (i.e., <media>) of “m=” line will indicate one of the currently defined values: “audio”, “video”, “application”, “data”, “image”, or “control”.

Further studies are needed if <media> of the “m=” line is “video”, “application”, or “control”.

If the round-up bandwidth for <media> equal to audio is 64 kbps or the “b=” line is absent, then USI should be set to “3.1 kHz”, and the <transport> and <fmt-list> are evaluated to determine whether User information layer 1 protocol indicator of USI parameter should be set to “G.711 μ–law” or “G.711 A-law”.

Table 6.3 - Coding of USI from SDP: SIP to ISUP

m= line			b= line	a= line	USI parameter				HLC parameter
<media>	**<transport>**	**<fmt-list>**	**<modifier>:<bandwidth-value>** NOTE – <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.	**rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>**	**Information Transfer Rate**	**Rate Multiplier**	**Information Transport Capability**	**User information layer 1 protocol indicator**	**High layer characteristics identification**
audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A	64 kbit/s		3.1 kHz audio	G.711 □-law	(Note 3)
audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMU/8000	64 kbit/s		3.1kHz audio	G.711 □-law	(Note 3)
audio	RTP/AVP	9	AS:64 kbit/s	rtpmap:9 G.722/8000	64 kbit/s		Unrestricted digital inf. w/tones/ann.		
audio	RTP/AVP	Dynamic PT	AS:64 kbit/s	rtpmap:<dynamic-PT> CLEARMODE/8000 (Note-2)	64 kbit/s		Unrestricted digital information		
Image	udptl	t38	N/A or up to 64 kbit/s	Based on T.38	64 kbit/s		3.1kHz audio		Facsímile Group 2/3
Image	tcptl	t38	N/A or up to 64 kbit/s	Based on T.38	64 kbit/s		3.1kHz audio		Facsímile Group 2/3
audio	RTP/AVP	Dynamic PT	384 kbit/s	rtpmap;dynamic-PT>CLEARMODE/8000 (Note-2)	384 kbit/s		Unrestricted digital information		
audio	RTP/AVP	Dynamic PT	1472 kbit/s	rtpmap;dynamic-PT>CLEARMODE/8000 (Note-2)	1472 kbit/s		Unrestricted digital information		
audio	RTP/AVP	Dynamic PT	1536 kbit/s	rtpmap;dynamic-PT>CLEARMODE/8000 (Note-2)	1536 kbit/s		Unrestricted digital information		

NOTE –
1 - In this table, the codec G.711 is used only as an example. Other codecs are possible.
2 - CLEARMODE has not yet been standardized, and its usage is for further study.
3 - HLC normally absent in this case. It is possible for HLC to be present with the value "Telephony", although ATIS-1000607.2000, clause 4.5.5, indicates that this would normally be accompanied by a value of "Speech" for the Information Transfer Capability element.

6.1.3.6 ISUP Calling Line Identification (CLI) Parameters

Table 6.4 summarizes the cases for mapping from the SIP INVITE header fields to the ISUP CLI parameters. Table 6.5 provides details when the Calling Party Number is given a network provided value. Table 6.6 provides details for Calling Party Number mapping in other cases. Finally, Table 6.7 provides details for mapping from the SIP From header field to the ISUP Calling Party Number.

- *For SIP-I*: If the address within the Calling Party Number or Generic Address after application of the mapping in this clause and processing by ISUP procedures is the same as the respective value contained in the encapsulated ISUP; no additional interworking is needed for that parameter beyond use of ISUP encapsulation. The contrary case is treated in the same way as for when ISUP encapsulation is not used.

  Should any discrepancy occur in privacy settings during the alignment process, the strongest privacy shall prevail.

Table 6.4 - Mapping of SIP From/P-Asserted-Identity/Privacy headers to ISUP CLI parameters

Has a "P-Asserted-Identity" header field containing a URI (Note 2) with an identity in the format "+CC"+"NDC"+"SN" been received?	**Has a "From" header field (Note 3) containing a URI with an identity in the format "+CC"+"NDC"+"SN" been received?**	**Calling Party Number parameter Address signals**	**Calling Party Number parameter APRI**
No	No	Network option to either include a network provided E.164 number (See Table 6.5) or omit the Address Signals	If a Privacy header field was received set APRI as indicated in Table 6.6 otherwise, Network option to set APRI to "presentation restricted" or "presentation allowed"
No	Yes	Derive from the From header field. (See Table 6.7)	If a Privacy header field was received set APRI as indicated in Table 6.6 otherwise, Network option to set APRI to "presentation restricted" or "presentation allowed"
Yes	No	Derive from P-Asserted-Identity (See Table 6.6)	APRI = "presentation restricted" or "presentation allowed" depending on SIP Privacy header. (See Table 6.6)
Yes	Yes	Derived from P-Asserted-Identity (See Table 6.6)	APRI = "presentation restricted" or "presentation allowed" depending on SIP Privacy header. (See Table 6.6)

6.1.3.6.1 Calling Party Number

Table 6.5 - Setting of the Network-provided ISUP Calling Party Number parameter with a CLI

ISUP CgPN Parameter field	Value
Screening Indicator	*“network provided”*
Number Plan Indicator	ISDN/Telephony (E.164)
Address Presentation Restricted Indicator	Presentation allowed/restricted (See Table 6.4)
Nature of Address Indicator	If next ISUP node is located in the same country set to “National (Significant) number“ else set to “International number”
Address signals	If NOA is *“national (significant) number”* no country code should be included. If NOA is “*international number”,* then the country code of the network-provided number should be included.

Table 6.6 - Mapping of P-Asserted-Identity and Privacy Headers to the ISUP Calling Party Number Parameter

Source SIP header field and component	Source component value	Calling Party Number parameter field	Derived value of parameter field
-	-	Numbering Plan Indicator	"*ISDN (Telephony) numbering plan (Recommendation E.164)*"
P-Asserted-Identity, appropriate global number portion of the URI, assumed to be in form "+" CC+NDC+SN (Note 1)	CC	Nature of Address Indicator	If CC is equal to the country code of the country where I-IWU is located AND the next ISUP node is located in the same country then set to *“national (significant) number”* else set to “*international number*”
Privacy, priv-value component (Note 2)	Privacy header field absent	Address Presentation Restricted Indicator (APRI)	"*presentation allowed*"
	"*none*"		"*presentation allowed*"
	"*header*"		"*presentation restricted*"
	"*user*"		"*presentation restricted*"
	"*id*"		"*presentation restricted*"
-	-	Screening Indicator	"*network provided*"
P-Asserted-Identity, appropriate global number portion of the URI, assumed to be in form "+" CC+NDC+SN (Note 1)	CC, NDC, SN	Address Signals	If NOA is “*national (significant) number*” then set to NDC + SN. If NOA is “*international number*” then set to CC+NDC+SN

NOTE –

1 - It is possible that the P-Asserted-Identity header field includes both a tel: URI and a sip: URI. The handling of this case is for further study.

2 - It is possible to receive multiple priv-values, one of which is "*none*", the other "*id*", “header”, or “user”. In this case, APRI shall be set to "*presentation restricted*".

Table 6.7 - Mapping of SIP From Header Field to ISUP Calling party number parameter

Source SIP header field and component	Source component value	Call party number parameter field	Derived value of parameter field
–	–	Type of Address	“supplemental user provided calling address – not screened"
From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	CC	Nature of Address Indicator	If CC is equal to the country code of the country where I-IWU is located AND the next ISUP node is located in the same country then set to *“national (significant) number”* else set to “*international number*”
–	–	Numbering Plan Indicator	"*ISDN (Telephony) numbering plan (Recommendation E.164)*"
Privacy, priv-value component (Note 1)	Privacy header field absent	Address Presentation Restricted Indicator (APRI)	”presentation allowed”
	“none”		“presentation allowed”
	“header”		“presentation restricted”
	“user”		“presentation restricted”
	“id”		“presentation restricted”
	-	Screening Indicator	“user provided, not screened”
From, userinfo component assumed to be in form "+" CC+NDC+SN	CC, NDC, SN	Address Signals	If NOA is “*national (significant) number*” then set to NDC + SN. If NOA is “*international numbe*r” then set to CC+NDC+SN
NOTE – 1 – It is possible to receive multiple priv-values, one of which is “none”, the other “header”, “user” or “id”. In this case, APRI shall be set to “presentation restricted”.			

6.1.3.6.2 Generic Address

In the presence of the routing number and npdi fields in the userinfo component of the Request-URI, the geographical telephone number field contained in the userinfo component of the Request-URI shall be mapped to the GAP of the IAM. The coding of the GAP set by the I-IWU is as specified in the current ISUP Standards and Table 6.8 below.

Table 6.8 - Mapping of SIP Request-URI to ISUP Generic Address (ported number) parameter

Source SIP header field and component	Source component value	Generic Address parameter field	Derived value of parameter field
–	–	Type of Address	"ported number "
- "+" CC+NDC+SN	-	Nature of Address Indicator	*“national (significant) number”*
–	–	Numbering Plan Indicator	"*ISDN (Telephony) numbering plan (Recommendation E.164)*"
Geographical number in Userinfo	"+CC""NDC""SN" from the URI	Address signal	Set to the “NDC” + “SN”
NOTE – The "Geographical number" in the Userinfo refers to the initial telephone number (immediately following "sip:") in the Request-URI.			

6.1.3.7 Hop counter

For the SIP-I case, the I-IWU acting as an independent exchange shall perform the normal ISUP Hop Counter procedure using the Hop Counter taken from the encapsulated IAM, if the Hop Counter parameter is available. The procedure applicable to the no ISUP encapsulation case shall also be used for SIP-I, if no Hop Counter parameter is received in the encapsulated IAM and the succeeding network supports the Hop Counter procedure.

For the no ISUP encapsulation case, the I-IWU shall derive the Hop Counter parameter value from the Max-Forwards header field value by applying a factor to the latter as shown in Table 6.9, where the factor is constructed according to the following principles:

- Hop Counter for a given message should never increase and should decrease by at least 1 with each successive visit to an IWU, regardless of intervening interworking, and similarly for Max-Forwards in the SIP domain.
- The initial and successively mapped values of Max-Forwards should be large enough to accommodate the maximum number of hops that might be expected of a validly routed call.

Table 6.9 - Mapping from Max-Forwards to Hop Counter

Max-Forwards value	Hop Counter value
X	Y = Integer part of (X / Factor)

NOTE – The preceding rules imply that the mapping from Max-Forwards to Hop Counter will take account of the topology of the networks traversed. Since call routing and thus the number of hops taken will depend on the origin and destination of the call, the mapping factor used to derive Hop Counter from Max-Forwards should be similarly dependent on call origin and destination. Moreover, when call routing crosses administrative boundaries, the operator of the I-IWU will coordinate with adjacent administrations to provide a mapping at the I-IWU which is consistent with the initial settings or mapping factors used in the adjacent networks.

In summary, the factor used to map from Max-Forwards to Hop Counter for a given call will depend on call origin and call destination, and will be provisioned at the I-IWU based on network topology, trust domain rules, and bilateral agreement.

6.1.3.8 Transit Network Selection

If the host portion of the Request-URI includes a domain name that is not that of the I-IWU and is not that of the SS7 network at the I-IWU, then the I-IWU will map that domain name to the corresponding four-digit carrier identification code and populate the Transit Network Selection parameter with that four-digit value. Any TNS parameter received as part of an encapsulated ISUP IAM will be ignored.

6.1.3.9 Carrier Identification

If the Userinfo component of the SIP Request URI contains a “cic=” field, and if the I-IWU needs to send the ISUP Carrier Information parameter (CIP) in the outgoing IAM, then the CIP will be populated using the carrier identification code from the “cic=” field, not including any country code present in the “cic=” field.

6.1.3.10 Carrier Selection Information

If the I-IWU needs to send the ISUP Carrier Selection Information parameter (CSI) along with the Carrier Information parameter (CIP) in the outgoing IAM, then:

- *If ISUP encapsulation is used (i.e., for the SIP-I case) and a CSI value is included,* then the included CSI value shall be included in the outgoing IAM.
- *If ISUP encapsulation is used (i.e., for the SIP-I case) and no CSI value is included*, then the outgoing IAM shall include a CSI value of “0 0 0 0 0 0 0 0 no indication”

If ISUP encapsulation is not used, then the outgoing IAM shall include a CSI value of “0 0 0 0 0 0 0 1 selected carrier identification presubscribed and not input by calling party”.

6.1.3.11 Operator Services Information

If the digits of the Request URI include a leading “0” or “1” followed by additional digits, then the I-IWU will populate the Operator Services Information parameter in the outgoing IAM with Information Value Type = 001 (original access prefix) and Information Value = 0001 (1+ or 011+) or 0010 (0+ or 01+), as appropriate.

If the digits of the Request URI include only the digits “0” or “00”, then the I-IWU will populate the Operator Services Information parameter in the outgoing IAM with Information Value Type = 001 (original access prefix) and Information Value = 0011 (0-).

If the original dialed digits in the SIP History Info header or the Diversion header begin with a leading “0” or “1”, then the I-IWU will populate the Operator Services Information parameter in the outgoing IAM as indicated in the previous two paragraphs.

If the I-IWU identifies the host portion of the Request-URI as a domain name associated with a particular operator service that is accessed by a customer dialing pattern with a leading “0”, or “1”, then the I-IWU will populate the Operator Services Information parameter in the outgoing IAM with Information Value Type = 001 (original access prefix) and the appropriate Information Value.

This mapping is shown in Table 6.9a.

Table 6.9a

Dialed Digits (received or derived)	Information Value Type 001 Information Value
1+additional digits	0001
011+additional digits	0001
0+additional digits	0010
00+additional digits	0010
0 or 00 (only)	0011

6.1.3.12 Originating Line Information

If the Userinfo component of the P-Asserted-Identity header, or alternatively the From header if not present in the P-Asserted-Identity header, contains an “oli=” field, then the I-IWU will populate the Originating Line Information parameter in the outgoing IAM with the two-digit value from the “oli=” field. The interpretation of the values will be that given by the North American Numbering Plan Administrator [ref. http://www.nanpa.com/number_resource_info/ani_ii_assignments.html]

The “oli=” field is specified in 3GPP TS 24.229, Clause 7.2A.12.

6.1.3.13 Charge Number

If the incoming INVITE includes a P-Charge-Info header, then the I-IWU will include a Charge Number parameter in the outgoing IAM.

Thus, for example, a received P-Charge-Info header of:

P-Charge-Info: <SIP: +17326996201;user=phone>@domain

would be mapped to a Charge Number parameter of:

Odd/even indicator: Even

Nature of Address indicator: ANI of the calling party; national number (hex 0x03)

Numbering Plan: ISDN Numbering Plan

Address Information: 7326996201.

6.1.3.14 Jurisdiction Information

If the incoming INVITE includes an “rn” parameter in a From, P-Asserted-Identity, History-Info, or Diversion header, then the I-IWU must populate the Jurisdiction Information parameter with the six most significant digits in the “rn” parameter after removal of the "+CC" when present and CC equals "1". When the "+CC" is not present, but the "rn-context" reflects the North American numbering plan, the six most significant digits in the "rn" parameter are used. If the "+CC" or "rn-context" does not reflect the North American number plan, then the Jurisdiction Information parameter may be populated based on local policy. The "rn" parameter in the From header shall only be used based on local trust policy. The "rn" parameter in the P-Asserted-Identity, when present, takes precedence over the parameter should it appear in the From header. The parameter in the History-Info header or Diversion header entry associated with the redirecting parameter shall take precedence over the parameter in either the From header or P-Asserted-Identity header. If no "rn" parameter appears in the History-Info header or the Diversion header entry associated with the redirecting number, then no Jurisdiction Information parameter shall be provided in the ISUP IAM message.

6.1.3.15 Generic Digits

If the incoming INVITE includes a Geolocation header that contains a location URI consisting of digits and a Geolocation-Routing header set to “no” (or no Geolocation-Routing header is included in the INVITE), then the I-IWU shall populate the Generic Digits parameter in the ISUP IAM as follows:

- Type of Digits: “location identification number”,
- Encoding Scheme: “BCD even,” or “BCD odd”, as appropriate, and
- Digits: the digits (removing the Country Code, if present) from the userinfo portion of the location URI.

Note that this mapping is not symmetric with the mapping of the ISUP Generic Digits parameter to the SIP From header in Section 7.1.3.

6.1.3.16 Interworking for ETS

Table 6.9b - SIP to ISUP Interworking in Support of ETS

SIP	ISUP
R-URI/To: = Destination Number This is a normal (non-ETS) call	CdPN = Destination Number CPC not equal to NS/EP Call MTP PRIORITY = 0 No PRECEDENCE PARAMETER
R-URI/To: = Destination Number RPH [ets.x] If from trusted source. See <VoIP-PSTNGW-SIP-01700> for non-	CdPN = Destination Number CPC = NS/EP Call, MTP PRIORITY = 1 No PRECEDENCE PARAMETER

SIP	ISUP
trusted source	
R-URI/To: = Destination Number RPH [ets.x, wps.y] If from trusted source. See <VoIP-PSTNGW-SIP-01700> for non-trusted source	CdPN = Destination Number CPC = NS/EP Call, MTP PRIORITY = 1 PRECEDENCE PARAMETER = Look Ahead for Busy = 10 Precedence Level = y (0 – 4) Network Identity = 0100 MLPP Service Domain = z (H'40024B' - H'40024F')
R-URI/To: = ETS-DN No RPH or ets.DF	CdPN = ETS-DN CPC = NS/EP Call, MTP PRIORITY = 1 No PRECEDENCE PARAMETER
R-URI/To: = ETS-DN RPH [ets.x, wps.y]	CdPN = ETS-DN CPC = NS/EP Call, MTP PRIORITY = 1 PRECEDENCE PARAMETER = Look Ahead for Busy = 10 Precedence Level = y (0 – 4) Network Identity = 0100 MLPP Service Domain = z (H'40024B' - H'40024F')
R-URI/To: = ETS-DN RPH [wps.y] This is an errored header	Request is rejected using 400 response with 417. See <VoIP-GEN-SIP-01800>
R-URI/To: = Destination Number RPH [wps.y] This is an errored header	Request is rejected using 400 response with 417. See <VoIP-GEN-SIP-01800>
The default (DF) priority level is provisioned at the PSTN Gateway and its value is determined by policy. The default priority level is used until the call can be authenticated. The NS/EP Call value for CPC is H'E2' Note that if y = 0 then z = H'40024B', and if y = 4 then z = H'40024F'.	

6.2 Sending of COT

When the I-IWU determines that all the preconditions on the incoming SIP side have been met and any continuity tests on the outgoing ISUP side have been successfully completed, the I-IWU shall send the COT message with the Continuity Indicator in the COT message set to "Continuity check successful."

6.3 Receipt of ACM

Table 6.10 provides a summary of how the ACM message is interworked to the SIP side by an I-IWU.

On receipt of the ACM, the backward SIP response sent on the incoming side of the I-IWU depends upon the value of the Called Party's Status Indicator in the Backwards Call Indicator (BCI) parameter of the ACM.

(A) *If the BCI (called party status indicator) = "subscriber free"*, then:

- *If no ISUP encapsulation is used*, then the 180 Ringing SIP response is sent from the I-IWU. If the I-IWU supports the P-Early-Media header as a network option and if the INVITE request included the P-Early-Media header, the I-IWU shall include in the SIP 180 Ringing response a P-Early-Media header authorizing early media, except when
  - o the I-IWU has already sent a reliable provisional response including a P-Early-Media header, as defined in IETF RFC 5009, and
  - o the most recently sent P-Early-Media header authorization matches that which would be sent.
- If the received ACM contained a Backward call indicators parameter with the Interworking indicator set to “interworking encountered”, or the Optional backward call indicators parameter with the User-Network interaction indicator set to "user-network interaction occurs, cut through in both directions", then the P-Early-Media header shall authorize backward and forward early media (i.e., "sendrecv"), otherwise the P-Early-Media header shall only authorize backward early media (i.e., "sendonly").
- *If ISUP encapsulation is used (i.e., for the SIP-I case)*, a 180 Ringing SIP response is sent from the I-IWU. The ACM is encapsulated within this response.

(B) *If the BCI (called party status indicator) = "in-band information or an appropriate pattern is now available"* then:

- *If ISUP encapsulation is not used and ISUP encapsulation is not used,* then if the I-IWU supports the P-Early-Media header as a network option and the INVITE request included the P-Early-Media header, the I-IWU shall send a SIP 183 Session Progress response with a P-Early-Media header authorizing *early* media, except when:
  - o the I-IWU has already sent a reliable provisional response including a P-Early-Media header, as defined in IETF RFC 5009, and
  - o the most recently sent P-Early-Media header authorization matches that which would be sent.

  If the received ACM contained a Backward call indicators parameter with the Interworking indicator set to “interworking encountered”, or the Optional backward call indicators parameter with the User-Network interaction indicator set to "user-network interaction occurs, cut through in both directions", then the P-Early-Media header shall authorize backward and forward early media (i.e., "sendrecv"), otherwise the P-Early-Media header shall only authorize backward early media (i.e., "sendonly").
- *As a network option, the O-IWU may return a 183 containing SDP.*
- *If ISUP encapsulation is not used,* and the P-Early-Media header is not supported or it is supported but the received INVITE request did not include the P-Early-Media header, or the network option to return a 183 was not performed, then the ACM is not interworked.

(C) *BCI (called party status indicator) = "no indication"*, then

- *If ISUP encapsulation is not used*, the ACM is not interworked;

  NOTE – A backward path is available as soon as the IAM is sent and appropriate SDP is received from the calling end.
- *In the case of SIP-I*, a 183 Session Progress response is sent from the I-IWU (see Table 6.10). The ACM is encapsulated within this response.

NOTE – ACM with cause parameter is not interworked (except for SIP-I operation). Protection against indefinite prolongation of the call is provided by timers specified in ATIS-1000113.2005.

Table 6.10 - Message sent to SIP upon receipt of ACM

←Message sent to SIP	←ACM Backward call indicators parameter Called party's status indicator	
183 Session Progress	00	*No indication*
180 Ringing	01	*Subscriber free*

6.4 Receipt of CPG

On receipt of the CPG, the backward SIP response sent on the incoming side of the I-IWU depends upon the value of the Event Information parameter of the CPG.

(A) If the Event information = "alerting", then:

- *If no ISUP encapsulation is used,* then the 180 Ringing SIP response is sent from the I-IWU. If the I-IWU supports the P-Early-Media header as a network option and if the INVITE request included the P-Early-Media header, the I-IWU shall include in the SIP 180 Ringing response a P-Early-Media header authorizing early media, except when
- *If no ISUP encapsulation is used*, then the 180 Ringing SIP response is sent from the I-IWU. If the I-IWU supports the P-Early-Media header as a network option and if the INVITE request included the P-Early-Media header, the I-IWU shall include in the SIP 180 Ringing response a P-Early-Media header authorizing early media, except when
    - the I-IWU has already sent a reliable provisional response including a P-Early-Media header, as defined in IETF RFC 5009, and
    - the most recently sent P-Early-Media header authorization matches that which would be sent.

If the received CPG contained a Backward call indicators parameter with the Interworking indicator set to “interworking encountered”, or the Optional backward call indicators parameter with the User-Network interaction indicator set to "user-network interaction occurs, cut through in both directions", then the P-Early-Media header shall authorize backward and forward early media (i.e., "sendrecv"), otherwise the P-Early-Media header shall only authorize backward early media (i.e., "sendonly").

- *If ISUP encapsulation is used (i.e., for the SIP-I case)*, a 180 Ringing SIP response is sent from the I-IWU. The ACM is encapsulated within this response.

(B) Event information = "in-band information or an appropriate pattern is now available" then:

- *If ISUP encapsulation is not used and ISUP encapsulation is not used then if* the I-IWU supports the P-Early-Media header as a network option and the INVITE request included the P-Early-Media header, the I-IWU shall send a SIP 183 Session Progress response with a P-Early-Media header authorizing early media, except when
    - the I-IWU has already sent a reliable provisional response including a P-Early-Media header, as defined in IETF RFC 5009, and
    - the most recently sent P-Early-Media header authorization matches that which would be sent.

If the received CPG contained a Backward call indicators parameter with the Interworking indicator set to “interworking encountered”, or the Optional backward call indicators parameter with the User-Network interaction indicator set to "user-network interaction occurs, cut through in both directions", then the P-Early-Media header shall authorize backward and forward early media (i.e., "sendrecv"), otherwise the P-Early-Media header shall only authorize backward early media (i.e., "sendonly").

- *As a network option, the O-IWU may return a 183 containing SDP.*

If ISUP encapsulation is not used, and either the P-Early-Media header is not supported or it is supported but the received INVITE request did not include the P-Early-Media header, or the network option to return a 183 was not performed then the CPG is not interworked.

For the SIP-I case, on receipt of a CPG message, either a 180 Ringing or 183 Session Progress SIP response shall be sent from the SIP side of the I-IWU as shown in Table 6.11. This response shall encapsulate the CPG message.

Table 6.11 - Receipt of CPG at the I-IWU

←Message sent to the SIP	←CPG Event information parameter Event indicator
180 Ringing	000 0001 (*alerting*)
183 Session Progress in case of SIP-I; otherwise not interworked.	000 0010 (*progress*) or 000 0011 (*in-band information or an appropriate pattern is now available*)

6.5 Receipt of Answer Message (ANM)

On receipt of ISUP ANM, the IWU shall indicate to the SIP protocol to send a 200 OK INVITE to the UAC. If no offer was received in the initial INVITE, and reliable provisional responses were not supported, the 200 OK INVITE shall include an SDP offer consistent with the USI used on the ISUP side. When SIP-I is used, the Answer message is encapsulated in a 200 OK INVITE final response.

6.6 Confusion Message

The Confusion message shall be discarded by the I-IWU unless encapsulation is used. The message may be encapsulated and passed to the SIP side by the IWU in specific cases (see ATIS-1000113.2005, chapter 4, clause 2.9.5.4.2).

6.7 Receipt of Circuit (CIC) Query response Message

After sending a Circuit (CIC) Query Message, the I-IWU expects to receive a Circuit (CIC) Query Response Message. The I-IWU shall process the Circuit (CIC) Query Response Message as described in ATIS-1000113.2005, chapter 4, clause 2.8.2A. If the ISUP procedures result into release of a call, appropriate actions shall be taken on the SIP side. In the case that ISUP encapsulation is used, the Circuit (CIC) Query Response Message shall not be encapsulated.

6.8 Receipt of EXM

The actions taken on the ISUP side upon receipt of the Exit Message (EXM) are described in ATIS-1000113.2005, chapter 4, clause 2.1.3.A.2

If the I-IWU does not encapsulate ISUP messages into SIP messages, no action is taken upon receipt of an EXM; the EXM is ignored.

If the I-IWU does encapsulate ISUP messages into SIP messages, the EXM is encapsulated in the MIME body of a 183 Session Progress (if this is the first backward 183 SIP message) or an INFO request (if a backward 183 SIP message had previously been sent).

See Table 6.12 below.

Table 6.12 - 183 Session Progress sent to SIP upon receipt of EXM

←Message sent to SIP	←Message Received from ISUP	Conditions
183 Session Progress	EXM	If a 183 Session Progress had not been previously sent
INFO	EXM	If a 183 Session Progress had been previously sent

6.9 Receipt of PAM

The actions taken on the ISUP side upon receipt of the Pass Along Message (PAM) are based on the contents of the PAM.

In the case of SIP-I, the PAM is encapsulated in the MIME body of an INFO request, see Table 6.13 below.

Table 6.13 - INFO sent to SIP upon receipt of PAM

←Message sent to SIP	←Message Received from ISUP
INFO	PAM

6.10 Through Connection of the Bearer Path

Through connection of bearer path is applicable only to I-IWUs that are capable of bearer control.[29]

Through connection at the I-IWU shall follow the ATIS-1000113.2005 through connection procedures for the originating exchange, unless the ACM or CPG message includes an Optional Backward Call Indicators (OBCI) parameter with the User Network Interaction indicator field set to "user network interaction, cut through in both directions", then the I-IWU shall follow the through connect the transmission path in both directions.

As a network option, the I-IWU may follow the through connect the transmission path in both directions upon receiving indication that inband information may be available, if not already done. For the SIP-I case, the I-IWU shall follow the through connection procedures in ATIS-1000113.2005 for the transit exchange.

If the ACM includes an Optional Backward Call Indicators (OBCI) parameter with the User Network Interaction indicator field set to "user network interaction, cut through in both directions", then the I-IWU shall follow the through connection procedures in ATIS-1000113.2005 for the transit exchange.

For the SIP-I case, the I-IWU shall follow the through connection procedures in ATIS-1000113.2005 for the transit exchange.

6.11 Receipt of Suspend Message (SUS) Network Initiated

If the I-IWU is the controlling exchange for the Suspend procedure, the actions taken on the ISUP side upon receipt of the suspend message (SUS) are described ATIS-1000113.2005, clause 2.5.1.3. For the no ISUP encapsulation case, SUS is not interworked.

In the SIP-I case, the SUS is encapsulated in the MIME body of an INFO request. This is summarized in Table 6.14.

Table 6.14 - INFO sent to SIP upon receipt of SUS (SIP-I only)

←Message sent to SIP	←Message Received from ISUP
INFO	SUS

[29] ITU-T Q.1912.5 refers to these types of gateways as either *Type 1* (no ISUP encapsulation) or *Type 3* (with ISUP encapsulation).

6.12 Receipt of Resume Message (RES) Network Initiated

If the I-IWU is the controlling exchange for the Resume procedure, the actions taken on the ISUP side upon receipt of the resume message (RES) are described in ATIS-1000113.2005, clause 2.4.2c. RES is not interworked and no action is taken on the SIP side.

In the SIP-I case, the I-IWU shall encapsulate the RES in an INFO method. This is summarized in Table 6.15.

Table 6.15 -of Resume Message (RES) network initiated (SIP-I only)

←Message sent to SIP	←Message Received from ISUP
INFO	RES

6.13 Release Procedures at the I-IWU

6.13.1 Receipt of BYE/CANCEL

On receipt of SIP BYE or CANCEL, the I-IWU shall send an ISUP REL to the ISUP side.

In the case of SIP-I, the encapsulated REL received in a BYE message shall be passed to ISUP procedures without modification. A received CANCEL message shall be treated as described for the no ISUP encapsulation case below.

For the no ISUP encapsulation case, if the Reason header field is included in the BYE or CANCEL, then the cause value may be mapped to the ISUP cause value field in the ISUP REL as shown in Table 6.16, depending on local policy. Table 6.17 shows the coding of the cause value in the REL if it is not available from the Reason header field. In all cases, the Location Field shall be set to "*network beyond interworking point*".

Table 6.16 - Mapping of SIP Reason header fields into Cause Indicators parameter

component of SIP Reason header field	component value	ISUP Parameter / field	Value
protocol	"*Q.850*"	Coding standard	*ITU-T Standard*
protocol	"*ANSI*"	Coding standard	*ANSI Standard*
protocol-cause	"*cause = XX*" (Note 1)	Cause Value	"*XX*" (Note 1)
-	-	Location	*Network beyond interworking point*
Note 1 – "XX" is the Cause Value as defined in Q.850 or ATIS-1000113.2005 (depending on value of Coding standard).			

Table 6.17 - Coding of Cause Value if not taken from the Reason header field (except when encapsulated REL received)

SIP Message →	REL → Cause Indicators parameter
BYE	Cause value No. 16 (normal clearing)
CANCEL	Cause value No. 31 (normal unspecified)

6.13.2 Receipt of REL

On receipt of an ISUP REL, the I-IWU immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side.

Depending on local policy a Reason header field containing the received (Q.850 or ANSI) Cause Value of the REL may be added to the SIP final response or BYE sent as a result of this clause. The mapping of the Cause Indicators parameter to the Reason header is shown in Table 6.18.

Table 6.18 - Mapping of Cause Indicators parameter into SIP Reason header fields

Cause indications parameter field	Value of parameter field	component of SIP Reason header field	component value
Coding standard	*ITU-T Standard*	protocol	"*Q.850*"
Coding standard	*ANSI Standard*	protocol	"ANSI"
Cause Value	"*XX*" (Note 1)	protocol-cause	"*cause= XX*" (Note 1)
-	-	reason-text	Should be filled with the definition text as stated in Q.850 or ATIS-1000113.2005 (Note 2)
NOTE – 1 – "XX" is the Cause Value as defined in Q.850 or ATIS-1000113.2005. 2 – Due to the fact that the Cause Indications parameter does not include the definition text as defined in Q.850 or ATIS-1000113.2005, this is based on provisioning in the O-IWU.			

On receipt of REL before receiving ANM, the I-IWU shall send the appropriate SIP status-code in a final response to the SIP peer. See Table 6.19 for the mapping from ISUP cause code to SIP status code. ISUP cause codes not appearing in Table 6.19 shall have the same mapping as the appropriate ATIS-1000113.2005 class defaults.

For the SIP-I, the appropriate SIP status code of the SIP response that encapsulates the REL message should be same as the default mapping shown in Table 6.19 for the no ISUP encapsulation case.

Table 6.19 - Receipt of the Release message (REL)

←SIP Message	← REL Cause Indicators parameter (Note 5)
Cause values with Coding Standard field set to 00 (ITU-T standard) (Note 2)	
404 Not Found	Cause value No. 1 (unallocated (unassigned) number)
604 Does not exist anywhere	Cause value No 2 (no route to network)
604 Does not exist anywhere	Cause value No 3 (no route to destination)
500 Server Internal Error	Cause value No. 4 (Send special information tone)
No mapping (No procedure specified for this cause value in U.S. networks)	Cause value No. 5 (Misdialed trunk prefix) (No procedures specified for this cause value in U.S. networks)
500 Server Internal Error (SIP-I only)	Cause value No. 8 (Preemption)
500 Server Internal Error (SIP-I only)	Cause value No. 9 (Preemption-circuit reserved for reuse)
486 Busy Here	Cause value No. 17 (user busy)
480 Temporarily unavailable	Cause value No 18 (no user responding)
480 Temporarily unavailable	Cause value No 19 (no answer from the user)
480 Temporarily unavailable	Cause value No. 20 (subscriber absent)
603 Decline IF location field is set to user ELSE 403 Forbidden	Cause value No 21 (call rejected)
410 Gone	Cause value No 22 (number changed)
No mapping (due to redirection procedures)	Cause value No. 23 (redirect to a new destination)
502 Bad Gateway	Cause value No. 27 (destination out of order)
484 Address Incomplete	Cause value No. 28 invalid number format (address incomplete)
501 (Not Implemented)	Cause value No 29 (facility rejected)
480 Temporarily unavailable	Cause value No 31 (normal unspecified) (Class default) (Note 3)
503 Service Unavailable (Note 4)	Cause value No. 34 (no circuit/channel available)
500 Server Internal error	Cause value No 38 (Network out of order)
503 Service Unavailable (Note 4)	Cause value No 41 (Temporary failure)
503 Service Unavailable (Note 4)	Cause value No 42 (Switching equipment congestion)
500 Server Internal error	Cause value No 43 (Access information discarded)
503 Service Unavailable (Note 4)	Cause value No 44 (Requested channel not available)
500 Server Internal error (SIP-I only)	Cause value No 46 (Precedence call blocked)
503 Service Unavailable (Note 4)	Cause value No 47 (Resource unavailable, unspecified) (class default)
488 Not acceptable here	Cause value No 50 (requested facility not subscribed)
No mapping (No procedure specified for this cause value in U.S. networks)	Cause value No. 53 (outgoing calls barred within CUG) (No procedure specified for U.S. networks)
No mapping (No procedure specified for this cause value in U.S. networks)	Cause value No. 55 (incoming calls barred within CUG) (No procedure specified for U.S. networks)
603 Decline	Cause value No 57 (bearer capability not authorized)
503 Service Unavailable (Note 4)	Cause value No 58 (bearer capability not presently available)
500 Server Internal Error	Cause value No. 62 (inconsistency in designated outgoing access information and subscriber class)
501 (Not Implemented)	Cause value No 63 (service option not available, unspecified) (Class default)
500 Server Internal error	Cause value No 65 (Bearer capability not implemented)
501 Not Implemented	Cause value No 69 (Requested facility not implemented)
501 Not Implemented	Cause value No 70 (Only restricted digital information capability is available)
501 Not Implemented	Cause value No 79 (Service or option not implemented, unspecified) (class default)

←SIP Message	← REL Cause Indicators parameter (Note 5)
No mapping (No procedure specified for this cause value in U.S. networks)	Cause value No. 87 (user not member of CUG) (No procedure specified for U.S. networks)
606 Not Acceptable	Cause value No 88 (incompatible destination)
No mapping (No procedure specified for this cause value in U.S. networks)	Cause value No. 90 (non-existent CUG) (No procedure specified for U.S. networks)
500 Server Internal error	Cause value No 91 (invalid transit network selection)
501 Not Implemented	Cause value No 95 (invalid message) (Class default)
501 Not Implemented	Cause value No 97 (Message type non-existent or not implemented)
501 Not Implemented	Cause value No 99 (information element/parameter non-existent or not implemented))
504 Server timeout	Cause value No. 102 (recovery on timer expiry)
501 Not Implemented	Cause value No 103 (Parameter non-existent or not implemented, passed on)
501 Not Implemented	Cause value No 110 (Message with unrecognized parameter, discarded)
400 Bad Request	Cause value No. 111 (protocol error, unspecified) (Class default)
500 Server Internal error	Cause value No. 127 (interworking unspecified) (Class default)
Cause values with Coding Standard field set to 01 (ANSI standard) (Note 2)	
404 Not Found	Cause value No. 23 (unallocated destination number)
500 Server Internal Error	Cause value No. 24 (unknown business group)
500 Server Internal Error	Cause value No. 25 (exchange routing error)
404 Not Found (Note 1)	Cause value No. 26 (misrouted call to a ported number)
No mapping. (No procedure specified for this cause value in U.S. networks)	Cause value No. 27 (Number Portability (NP) Query on Release (QoR) – number not found) (No procedures specified for this cause value in U.S. networks)
500 Server Internal Error (SIP-I only)	Cause value No. 45 (preemption)
500 Server Internal Error (SIP-I only)	Cause value No. 46 (precedence call blocked)
500 Server Internal Error	Cause value No. 51 (call type incompatible with service request)
No mapping (No procedure specified for this cause value in U.S. networks)	Cause value No. 54 (call blocked due to group restrictions)
NOTE – 1 – This may impact operations and maintenance. 2 – The Coding Standard field in the Cause Indicators parameter in the received REL message may be set to either "ITU-T Standard" or "ANSI Standard." This table is separated into two sections pertaining to each of these values of the Coding Standard field. 3 – Although cause value 31 is not formally in class 0, it is used as the default value for both class 0 and class 1. 4 – No Retry-After header field shall be included. 5 – For Cause Value No. 16, see Section 6.13.1.	

On receipt of REL after receiving ANM, the I-IWU shall send BYE. For SIP-I, this BYE message shall encapsulate the received REL message.

6.13.3 Autonomous Release at I-IWU

Table 6.20 shows the trigger events at the IWU and the release initiated by the IWU when the call is traversing from SIP to ISUP.

If an automatic repeat attempt initiated by the I-IWU is not successful (because the call is not routable), the I-IWU shall send a 480 Temporarily Unavailable response to the SIP side. No actions on the ISUP side are required.

If, after answer, ISUP procedures result in autonomous REL from the IWU, then a BYE shall be sent on the SIP side.

If the I-IWU receives unrecognized backward ISUP signaling information and determines that the call needs to be released based on the coding, the I-IWU shall send a 500 Internal Server Error response on the SIP side. Depending on local policy a Reason header field containing the (Q.850 or ANSI) Cause Value of the REL message sent by the I-IWU may be added to the SIP Message (BYE or final response) sent by the SIP side of the I-IWU.

For SIP-I, depending on the trigger event, a BYE or the appropriate SIP status code of the SIP response that encapsulates the REL message should be the same as the default mapping shown in Table 6.20 for the no ISUP encapsulation case.

Table 6.20 - Autonomous Release at I IWU

← SIP	Trigger event	REL → cause parameter
484 Address Incomplete	Determination that insufficient digits received See 6.2.2.	Not applicable.
480 Temporarily Unavailable	Congestion at the IWU.	Not applicable.
BYE	ISUP procedures result in release after answer	According to ISUP procedures
500 Server Internal Error	Call release due to the ISUP compatibility procedure (Note 1)	According to ISUP procedures.
484 Address Incomplete	Call release due to expiry of T7 within the ISUP procedures	According to ISUP procedures.
480 Temporarily Unavailable.	Other ISUP procedures result in release before answer.	According to ISUP procedures.
NOTE – IWU receives unrecognized ISUP signaling information and determines that the call needs to be released based on the coding of the compatibility indicators, refer to ATIS-1000113.2005.		

6.13.4 Receipt of RSC, GRS or CGB

Table 6.21 shows the message sent by the IWU upon receipt of an ISUP RSC message, GRS message, or CGB message with the Circuit Group Supervision Message Type Indicator coded as “hardware failure oriented”, when at least one backward ISUP message relating to the call has already been received. The IWU sends BYE if it has already received an ACK for the 200 OK INVITE it had sent. If it has sent 200 OK INVITE but has not yet received an ACK for the 200 OK INVITE, then the IWU shall wait until it receives the ACK for the 200 OK INVITE before sending the BYE. Otherwise, it sends 500 Server Internal Error. On receipt of a GRS or CGB message, one SIP message is sent for each call association. Therefore, multiple SIP messages may be sent on receipt of a single GRS or CGB message.

In the case that ISUP encapsulation is being used, the SIP BYE or 500 Server Internal Error message shall encapsulate the REL generated by ISUP procedures, rather than the RSC, GRS, or CGB message that caused it to be generated.

Table 6.21 - Receipt of RSC, GRS, or CGB messages (ISUP)

← SIP	← Message received from ISUP
500 Server Internal Error or BYE	Reset Circuit message (RSC)
500 Server Internal Error or BYE	Circuit Group Reset message (GRS)
500 Server Internal Error or BYE	Circuit Group Blocking message (CGB) with the Circuit Group Supervision Message Type indicator coded *"hardware failure oriented"*

7 Outgoing Call Interworking from ISUP to SIP at O-IWU

An Outgoing interworking Unit (O-IWU) is used to transport calls from an ISUP network domain to a SIP network domain.

The "outgoing SIP" refers to the SIP protocol, which is used between the Outgoing IWU and the call terminating entity (entities) in the SIP network domain. Similarly, by definition, "incoming ISUP" refers to the ISUP protocol supported between the Outgoing IWU and the preceding entity.

The Outgoing IWU receives forward and backward signaling information from the "incoming ISUP" and "outgoing SIP" sides, respectively. After receiving this signaling information and performing appropriate call/service processing, the Outgoing IWU may signal to subsequent SIP nodes or preceding ISUP entities.

If the address information received from the preceding ISUP exchange is not in the form of an E.164 international public telecommunication number, the O-IWU shall add the country code or the country code and national destination code of the preceding exchange to form the international public telecommunication number.

This clause specifies the signaling interworking requirements for basic call at the outgoing IWU. The chapter is split into clauses based upon the messages sent or received on the outgoing (SIP) interface of the IWU. Only messages that are generated as a result of interworking to/from the incoming ISUP side of the IWU are considered in this interworking. Messages that are generated as a result of a local protocol state machine are not re-described in this specification.

For the SIP-I operation, ISUP message segmentation must be handled as described in 5.4.3.3.

Figure 7.1 illustrates an example signaling message flow for the interworking between SIP and ISUP at the O-IWU. The flow illustrates SIP preconditions and SIP with encapsulated ISUP.

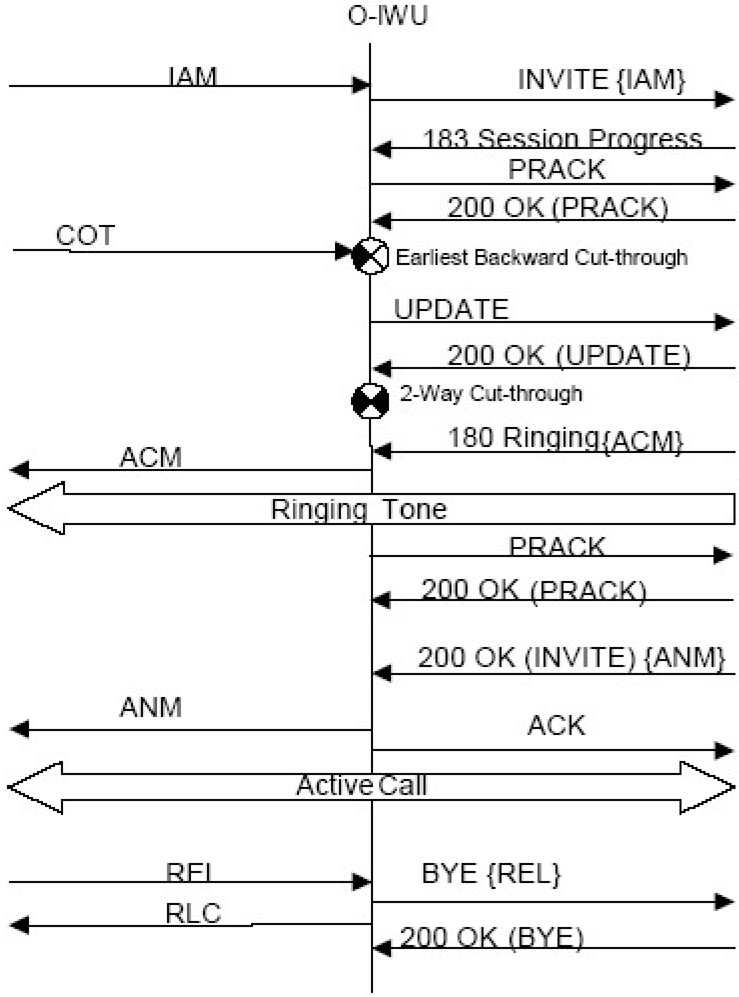


Figure 7.1 - Example O-IWU Signaling Message Flow

7.1 Sending of INVITE

After performing the normal ISUP handling for incoming IAM and choosing to route the call to the SIP network domain, the O-IWU will invoke the outgoing SIP signaling procedures as described in this clause.

The O-IWU will invoke the outgoing SIP signaling procedure using one of the following scenarios. Which scenario is used depends upon whether preconditions are used in the SIP network:

(A) Send INVITE without precondition upon receipt of ISUP IAM.

(B) Send INVITE with precondition upon receipt of ISUP IAM.

Details of the procedures are described in this subclause. Coding of the IAM received and the INVITE sent by the O-IWU are specified in 7.1.1 through 7.1.5.

For SIP-I, the IAM resulting from the application of ISUP procedures and the procedures of this clause is encapsulated in the outgoing INVITE.

Timer (T_{OIW2}) is started when the INVITE is sent.

If timer (T_{OIW2}) expires, an early ACM is sent to the ISUP network. See 7.3.

A) Sending INVITE without Precondition for ISUP IAM

Outgoing SIP procedures apply with the following clarifications and exceptions with regards to when INVITE is to be sent.

INVITE is sent when the ISUP IAM is received and the Continuity Check indicator in the Nature of Connection Indicators parameter in the IAM is set to indicate “continuity check not required”.

Sending of INVITE is delayed if the Continuity Check indicator in the Nature of Connection Indicators parameter in the IAM is set to indicate either “continuity check required on this circuit” or “continuity check performed on previous circuit”. INVITE shall be sent on receipt of the Continuity message with the Continuity Indicators parameter set to “continuity check successful”. INVITE shall not be sent if the Continuity message is received with the Continuity Indicators parameter set to “continuity check failed” or the ISUP timer T8 expires.

B) Sending INVITE with Precondition for ISUP IAM

INVITE with precondition is sent on receipt of ISUP IAM. Incoming ISUP procedures apply, with the following clarifications and exceptions as to when a confirmation of the precondition being met is to be sent.

NOTE – Configured procedures may delay the INVITE until local resources have been reserved on the outgoing bearer path.

The O-IWU should initiate the precondition signaling procedure using the SDP-Offer in the INVITE. The precondition signaling is concluded upon sending (within an SDP Offer-Answer exchange) the confirmation of a precondition being met. The SDP Offer or Answer carrying the confirmation of a precondition being met is sent when both of the following conditions are satisfied.

1. If the Continuity Check indicator in the Nature of Connection Indicators parameter in the incoming IAM is set to indicate either “continuity check required on this circuit” or “continuity check performed on previous circuit”, the Continuity message with the Continuity Indicators parameter set to “continuity check successful” shall be received.
2. The requested preconditions are met in the SIP network.

NOTE – As a network option (e.g., 3GPP), the signaling of “preconditions being met” always occurs within the SDP Offer in the UPDATE message.

CANCEL or BYE (according to the rule in 7.8.1) shall be sent if the Continuity message is received with the Continuity Indicators parameter set to “continuity check failed” or the ISUP timer T8 expires.

REL with Cause Value 47 (resource unavailable, unspecified) shall be sent on the ISUP side of the O-IWU and CANCEL or BYE (according to the rule in 7.7.1) shall be sent on the SIP side if internal resource reservation was unsuccessful. See 7.8.3 for further details.

For both cases of sending INVITE (A and B), Table 7.1 provides a summary of how the header fields within the outgoing INVITE message are populated.

Table 7.1 - Interworked Contents of the INVITE message

IAM (Note 1)→	INVITE→
Called Party Number	Request-URI (See 7.1.2))
	To (See 7.1.2)
Calling Party Number	P-Asserted-Identity (See 7.1.3)
	Privacy (See 7.1.3)
	From (See 7.1.3)
GAP (Supplemental User Provided Calling Address)	From (See 7.1.3)
GAP (Ported Number)	Request-URI (See 7.1.2)
Forward Call Indicators (Ported number translation indicator)	Request-URI (See 7.1.2)
Hop Counter	Max-Forwards (See 7.1.4)
Transit Network Selection	Request-URI (See 7.1.2)
USI	Message Body (application/SDP) (See 7.1.1)

Carrier Identification	Request-URI (See 7.1.2)
Carrier Selection Information	Request-URI (See 7.1.2)
Operator Services Information	Contact (TBD)
Calling Party’s Category	P-Asserted-Identity (See 7.1.3) From (See 7.1.3)
Calling Party’s Category	P-Asserted-Identity (See 7.1.3)
	From (See 7.1.3)
Originating Line Information	1. P-Asserted-Identity (See 7.1.3)
Charge Number	2. P-Charge-Info (See 7.1.7)
Jurisdiction Information	P-Asserted-Identity (See 7.1.8)
	Diversion (See 7.1.8)
	History-Info (See 7.1.8)
ISUP Message	Message Body (application/ISUP) (Note 2)
NOTE – 1 – When the coding of the received CPC is “NS/EP call” or “emergency service call”, or when the Precedence parameter is received, the additional information to be sent in the INVITE is for further study. 2 – SIP-I only. See 5.4.1.3	

7.1.1 Coding of SDP Media Description Lines from USI

The User Service Information parameter of the IAM received by the O-IWU indicates the user-requested bearer service characteristics. USI codes should be mapped to the SDP information. ATIS-1000113.2005, chapter 3 provides an exhaustive listing of the available codes in the USI. In principle, any combination of those codes can be mapped into any SDP information as long as transcoding is available.

The O-IWU, as a network option (e.g., 3GPP) , shall be capable of encoding the SDP for the AMR codec, which is specified in RFC 3267, *RTP payload format and file storage format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) audio codec*.

7.1.1.1 Transcoding Available at the O-IWU

If G.711 encoding may be used then, the O-IWU shall send an SDP Offer with both “G.711 µ-law” and “G.711 A-law” included in the media description and “G.711 µ-law” shall take precedence over “G.711 A-law”.

7.1.1.2 Transcoding Not Available at the O-IWU

Table 7.2 provides the mapping relations from USI codes to SDP media description lines when transcoding is not available at the O-IWU.

Table 7.2 - Coding of SDP Media Description Lines from USI: ISUP to SIP

USI parameter	USI parameter	USI parameter	USI parameter	HLC IE in ATP	m= line			b= line	a= line
Information Transfer Rate	**Rate Multiplier**	**Information Transport Capability**	**User information layer 1 protocol indicator**	**High layer characteristics identification**	**<media>**	**<transport>**	**<fmt-list>**	**<modifier>:<bandwidth-value>**	**rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>**
speech		Speech	G.711 μ-law	"Ignore"	audio	RTP/AVP	0 (and possibly 8) Note 1	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000) Note 1
speech		Speech	G.711 μ-law	"Ignore"	audio	RTP/AVP	Dynamic PT (and possibly a 2nd Dynamic PT)	AS:64	rtpmap:<dynamic-PT> PCMU/8000 (and possibly rtpmap:8 PCMA/8000) Note 1
3.1 kHz audio		3.1 kHz audio	G.711 μ-law	Telephony or "HLC absent"	audio	RTP/AVP	0	AS:64	rtpmap:0 PCMU/8000
3.1 kHz audio		3.1 kHz audio		Facsimile Group 2/3	image	udptl	t38	AS:64	Based on T.38.
3.1 kHz audio		3.1 kHz audio		Facsimile Group 2/3	image	tcptl	t38	AS:64	Based on T.38.
64 kbit/s unrestricted		Unrestricted digital inf. W/tone/ann.	N/A	"Ignore"	audio	RTP/AVP	9	AS:64	Rtpmap:9 G.722/8000
64 kbit/s unrestricted		Unrestricted digital information	N/A	"Ignore"	audio	RTP/AVP	Dynamic PT	AS:64	rtpmap:<dynamic-PT> CLEARMODE/8000
2 × 64 kbit/s unrestricted	2	Unrestricted digital information	N/A	"Ignore"	FFS	FFS	FFS	FFS	FFS
384 kbit/s unrestricted		Unrestricted digital information	N/A	"Ignore"	FFS	FFS	FFS	FFS	FFS

NOTE –

1 – Both PCMA and PCMU required under the conditions stated in 7.1.1.

2 – Since CLEARMODE has not yet been standardized, its use is for further study.

3 - HLC normally absent in this case. It is possible for HLC to be present with the value "Telephony", although ATIS-1000607.2000, clause 4.5.5, indicates that this would normally be accompanied by a value of "Speech" for the Information Transfer Capability element.

7.1.2 Request-URI and To header field

The Called Party Number parameter of the IAM message contains the forward address information to derive the userinfo component of the INVITE Request-URI.

> NOTE – The O-IWU follows existing ISUP procedure to select the outgoing route. If a new Called Party Number is derived for the outgoing route, then the newly derived Called Party Number should be mapped into the userinfo component of the INVITE Request URI.

For the basic call, the address information contained in the Called Party Number parameter is also considered as the identification of the called party. This information is used to derive the addr-spec component of the To: header field.

If the Request-URI or the To header field contains a sip: URI, it shall include the "user=phone" URI parameter.

For a dialed number for which a NP query has not been performed, the following rules shall be used to derive the userinfo component of the INVITE Request-URI (as summarized in Table 7.3):

- The Called party number parameter containing the dialed number shall be mapped to the geographical telephone number field.
- The FCI parameter containing the ported number translation indicator set to number not translated of the IAM shall not be mapped and the NP Database Dip Indicator (npdi) field shall not be included.

For a dialed number that has not been ported and a NP query has been performed, the following rules shall be used to derive the userinfo component of the INVITE Request-URI (as summarized in Table 7.3):

- The Called party number parameter containing the dialed number shall be mapped to the geographical telephone number field.
- The FCI parameter containing the ported number translation indicator set to number translated of the IAM shall be mapped to the NP Database Dip Indicator (npdi) field.

For a dialed number that has been ported and has a routing number associated with it, the following rules shall be used to derive the userinfo component of the INVITE Request-URI (as summarized in Table 7.4):

- The Called party number parameter containing the location routing number shall be mapped to the routing number field.
- The GAP containing the ported number shall be mapped to the geographical telephone number field.
- The FCI parameter containing the ported number translation indicator set to number translated of the IAM shall be mapped to the NP Database Dip Indicator (npdi) field.

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

ISUP Parameter / field	Value	SIP Component	Value
Called party number	Digits	Request-URI	Userinfo
Address signal	Either NCD + SN (national number) or CC + NCD + SN (international number)	Userinfo's geographical number	If national number, prepend +CC to Address signal digits, as in: "+CC" "NCD" "SN". If international number, prepend "+".
Forward Call Indicators	Ported number translation indicator	Userinfo's npdi parameter	If Ported number translation indicator is equal to "1", append ";npdi" to Userinfo.

Table 7. 4 - Mapping of Generic address (ported) and Called party number (when both are included), and FCI Ported number to SIP Request-URI

ISUP Parameter / field	Value	SIP Component	Value
Generic Address Type of number	"ported number"	Request-URI	Userinfo
Address signal	Since NOA is "*national (significant) number*" then the format of the address signals is: NCD + SN	Userinfo's geographical number	Add +CC to Address signal digits, as in: "+CC" "NCD" "SN"
Forward Call Indicators	Ported number translation indicator	Userinfo's npdi parameter	";npdi" is added to Userinfo
Called party number Address signal	Since NOA is "*national (significant) number*" then the format of the address signals is: NCD + SN	Userinfo's routing number	";rn=routing number" is added to Userinfo, with +CC being prefixed to Address signal's NCD+SN

In ANSI ISUP, carrier information is carried in three parameters:

1. The Carrier Identification parameter (CIP) indicates the transit network selected by the originating subscriber. The CIP is passed transparently from network to network.
2. The Carrier Selection Information (CSI) parameter indicates whether the calling user selected the transit network by presubscription or dialed input and if presubscribed whether or not the carrier identification code was also dialed. The CSI is passed transparently from network to network.
3. The Transit Network Selection (TNS) parameter indicates the identity of the transit network. The TNS is used to route the call to a specific network and is The Transit Network Selection parameter is removed before the call is passed to the indicated transit network.

The O-IWU follows the procedures in [RFC 3398] to map any received CIP to the "cic=" field of the Userinfo component of the SIP Request-URI. The "cic=" field is populated with the carrier identification code from the CIP.

The O-IWU does not map any received CSI into SIP. The information is lost unless ISUP encapsulation is used.

The O-IWU maps any received TNS value to a corresponding value for the domain of the Request-URI. Since the intent of the TNS parameter is to force the call setup to transit the named network, the result of this mapping will be either:

1. Delivering the call setup into the named (SIP) network, where the domain of the destination can be determined, or
2. Delivering the call setup to an I-IWU, where the domain will be mapped back into the TNS parameter with its original value.

Table 7.5 summarizes the mappings for the TNS, CIP, and CSI parameters.

Table 7.5 - Mapping of Carrier Identification parameter to SIP Request-URI

ISUP Parameter / field	Value	SIP Component	Value
Carrier Identification parameter (if available) Digits	4 digits, as in YYYY	Request URI Userinfo's carrier ID code	If CIP is available, ";cic=carrier ID code" is added to Userinfo, with +CC being prefixed to the CIP's Digits, as in: "+CC""YYYY"
Transit Network Selection parameter (if available)	4 digits, as in XXXX	Hostport component of Request-URI	If TNS is available, hostport component of Request-URI is derived from the TNS' Digits

7.1.3 P-Asserted-Identity, From, & Privacy Header Fields

The O-IWU maps any received Originating Line Information parameter value to the “oli” parameter in the P-Asserted-Identity header[30].

As a network option, the O-IWU may infer the originating line information from the ISUP trunk group and may use this information to determine the appropriate value of the “oli” parameter in the P-Asserted-Identity header.

The O-IWU maps the “Digits” field of any received Generic Digits parameter with “Type of Digits” field equal to “0 1 1 0 1” (location identification number) to the From header “addr-spec” field.

Table 7.6 provides details for mapping Calling Party Number to the P-Asserted-Identity and From header fields. Table 7.8 provides details for the mapping from Calling Party Number to P-Asserted-Identity, while Table 7.9 provides details for the mapping from Calling Party Number to the From header field. Table 7.10 provides details from the APRI sub-field of Calling Party Number into the Privacy header field. Table 7.10a gives the mapping from the Originating Line Information to the SIP P-Asserted Identity and From headers in the INVITE request. Table 7.10b gives the mapping from the Calling Party’s Category to the SIP P-Asserted Identity and From headers in the INVITE request.

If the From or the P-Asserted-Identity header field contains a sip: URI, it shall include the “user=phone" URI parameter.

[30] If the P-Asserted-Identity header is not supported, the O-IWU may, as a network option, map the value of the received Originating Line Information parameter to the “oli” parameter in the From header.

Table 7.6 - ISUP CLI Parameters to SIP Header fields

Has a Calling Party Number parameter with complete E.164 number, with APRI = "presentation allowed" or "presentation restricted" been received?	P-Asserted-Identity header field	From header field: display-name (optional) and addr-spec	Privacy header field
N	Header field not included	Unavailable@ Hostportion	Header field not included
Y (See Note 1)	If the Screening Indicator is "User provided, screening passed" or "Network provided", then derived from Calling Party Number parameter address signals (See Table 7.8) Otherwise, the P-Asserted-Identity header field is omitted.	display-name may be derived from Calling Party Number (CgPN) if possible. addr-spec is derived from Calling Party Number parameter address signals (See Table 7.9)	If Calling Party Number parameter APRI = "restricted" then priv-value is set to "user". "id" is also included when P-Asserted-Identity header field is provided. For other APRI settings Privacy header is not included or if included, "id" is not included. "user" also may not be included depending of privacy services applied. (See Table 7.10)
NOTE – 1 - A Network Provided CLI in the CgPN parameter may occur on a call from an analogue access line. Therefore in order to allow the "display" of this Network Provided CLI at a SIP UAS it must be mapped into the SIP From header. It is also considered suitable to map into the P-Asserted-Identity header since in this context it is a fully authenticated CLI related exclusively to the calling line, and therefore valid as a User Provided - Passed Screening CLI for this purpose.			

Table 7.7 – Void

Table 7.8 - Mapping of Calling Party Number parameter to SIP P-Asserted-Identity header fields

ISUP Parameter / field	Value	SIP Component	Value
Calling Party Number		P-Asserted-Identity header field	display-name (optional) and addr-spec
Nature of Address Indicator	"national (significant) number"	addr-spec	Add CC (of the country where the IWU is located) to CgPN address signals then map to URI
	"international number"		Map complete CgPN address signals to URI
Address signal	If NOA is "national (significant) number" then the format of the address signals is: NDC + SN If NOA is "international number" then the format of the address signals is: CC + NDC + SN	display-name	display-name may be mapped from Address Signal, if possible and network policy allows it
		addr-spec	"+CC" "NDC" "SN" mapped to the appropriate global number portion of URI scheme used

Table 7.9 - Mapping of ISUP Calling Party Number parameter to SIP From header fields

ISUP Parameter / field	Value	SIP Component	Value
Calling Party Number		From header field	display-name (optional) and addr-spec
Nature of Address Indicator	"national (significant) number"	addr-spec	Add CC (of the country where the IWU is located) to CgPN address signals then map to user portion of URI scheme used
	"international number"		Map complete CgPN address signals to map to user portion of URI scheme used
Address signal	If NOA is "national (significant) number" then the format of the address signals is: NDC + SN If NOA is "international number" then the format of the address signals is: CC + NDC + SN	display-name	Display-name may be mapped from Address Signal, if possible and network policy allows it.
		addr-spec	"+CC" "NDC" "SN" mapped to user portion of URI scheme used

Table 7.10 - Mapping of ISUP APRIs into SIP Privacy header fields

ISUP Parameter / field	Value	SIP Component	Value
Calling Party Number		Privacy header field	priv-value
APRI	“presentation restricted”	Priv-value	“id” and “user” (“id” included only if the P-Asserted-Identity header is included in the SIP INVITE)
	“presentation allowed”	Priv-value	omit Privacy header or Privacy header without “id” and maybe also without “user” if other privacy service is needed)
NOTE – When Calling Party Number parameter is received, P-Asserted-Identity header is always derived from it as in Table 7.8.			

Table 7.10a - Mapping of Originating Line Information parameter to SIP P-Asserted-Identity and From header fields

ISUP Parameter / field	Value	SIP Component	Value
Originating Line Information	binary equivalent of the II digits	P-Asserted-Identity header field	decimal equivalent of received value
Originating Line Information	binary equivalent of the II digits	From header field	decimal equivalent of received value

Table 7.10b - Mapping of Calling Party’s Category parameter to SIP P-Asserted-Identity and From header fields

ISUP Parameter / field	Value	SIP Component	Value
Calling Party’s Category	0000 0000 - unknown	P-Asserted-Identity header field	“unknown”
	0000 1001 - national operator		“operator”
	0000 1101 - ordinary calling subscriber		“ordinary”
	0000 1101 - test call		“test”
	1110 0000 - emergency service call		“emergency”
	1110 0001 - high priority emergency service call		“emergency”
	1110 0000 - National Security and Emergency Preparedness (NS/EP) Call		See section 7.1.11
	Any other value		“unknown”
Calling Party’s Category	0000 0000 - unknown	From header field	“unknown”
	0000 1001- national operator		“operator”
	0000 0010 - ordinary calling subscriber		“ordinary”
	0000 1101 - test call		“test”

ISUP Parameter / field	Value	SIP Component	Value
	1110 0000 - emergency service call		"emergency"
	1110 0001 - high priority emergency service call		"emergency"
	1110 0010 - National Security and Emergency Preparedness (NS/EP) Call		See section 7.1.11
	Any other value		"unknown"

7.1.4 Hop Counter (Optional) [Max Forwards]

For SIP-I, if the Hop Counter parameter is available, then the O-IWU acting as an independent exchange shall perform the normal ISUP Hop Counter procedure as it constructs the outgoing encapsulated IAM.

For the case in which ISUP encapsulation is not supported, the O-IWU shall derive the Max-Forwards header field value from the Hop Counter value when that is available. It shall do so by applying a factor to the Hop Counter value as shown in Table 7.11**Error! Reference source not found.**, where the factor is constructed according to the following principles:

a) Max-Forwards for a given message should never increase, and should decrease by at least 1 with each successive visit to an IWU, regardless of intervening interworking, and similarly for Hop Counter in the ISUP domain.

b) The initial and successively mapped values of Max-Forwards should be large enough to accommodate the maximum number of hops that might be expected of a validly routed call.

The following table shows the principle of the mapping:

Table 7.11 - Mapping from Hop Counter to Max-Forwards

Hop Counter value	Max-Forwards value
X	Y = Integer part of (X * Factor)

NOTE – The preceding rules imply that the mapping between Max-Forwards and Hop Counter will take account of the topology of the networks traversed. Since call routing -- and thus the number of hops taken -- will depend on the origin and destination of the call, the mapping factor used to derive Max-Forwards from Hop Counter should be similarly dependent on call origin and destination. Moreover, when call routing crosses administrative boundaries, the operator of the O-IWU will coordinate with adjacent administrations to provide a mapping at the O-IWU which is consistent with the initial settings or mapping factors used in the adjacent networks.

In summary, the factor used to map from Hop Counter to Max-Forwards for a given call will depend on call origin and call destination, and will be provisioned at the O-IWU based on network topology, trust domain rules, and bilateral agreement.

7.1.5 Coding of Encapsulated ISUP IAM Parameters in Outgoing INVITE (for SIP-I only)

This clause is used to specify coding of certain encapsulated ISUP information based on appropriate ISUP procedures. For computation of certain parameter/indicator values, the IWU is assumed to be an ISDN/PSTN exchange.

7.1.6 Nature of Connection Indicators Parameter

The O-IWU shall pass unchanged any received satellite indicator in the Nature of Connection Indicators parameter if the selected outgoing connection is using SIP-I.

7.1.7 Charge Number Parameter

Table 7.11a gives the mapping from Charge Number or Calling Party Number to the SIP P-Charge-Info header in the INVITE. See Annex E for the syntax of the P-Charge-Info header.

Table 7.11a - Mapping from Charge Number or Calling Party Number to the SIP P-Charge-Info header

Charge Number parameter	Originating Line Information parameter	Digits in P-Charge-Info header field
Y	Any	Digits from Charge Number parameter
N	Any value except 02 (ANI failure)	Digits from Calling Party Number parameter
N	02 (ANI failure)	Not sent

7.1.8 Jurisdiction Information

If the incoming IAM message includes a Jurisdiction Information parameter and no call forwarding has occurred, then the digits from the ISUP JIP shall be mapped to the "rn" parameter of the P-Asserted-Identity after prefixing them with "+1" and four additional filler digits (e.g., "0000") are appended to form a globally significant number. If call forwarding has previously occurred or invoked at the local exchange, then the digits from the ISUP JIP shall be mapped to the "rn" parameter of the History-Info header and/or Diversion header (depending on local policy) URI entry associated with the redirecting number after prefixing the digits with "+1 "and appending four additional filler digits (e.g., "0000") to form a globally significant number.

7.1.9 Incoming Trunk Group

Though not represented in an ISUP parameter, the incoming trunk group may be represented unambiguously in the Contact header in the INVITE by including both the "tgrp" parameter and the "trunk-context" parameter per IETF RFC 4904.

7.1.10 P-Early Media

For a speech call, if the O-IWU supports the P-Early-Media header as a network option, then it shall include the header with a value of "supported" in the initial INVITE request.

7.1.11 Interworking for ETS

The following defines the SIP to ISUP interworking in support of ETS.

Table 7.11b Mapping of ISUP to SIP in Support of ETS

ISUP	SIP
CdPN = Destination Number CPC not equal to NS/EP Call No PRECEDENCE PARAMETER This is a normal (non-ETS) call	R-URI/To: = Destination Number

ISUP	SIP
CdPN = ETS-DN CPC = NS/EP Call No PRECEDENCE PARAMETER	R-URI/To: = ETS-DN RPH [ets.DF]
CdPN = Destination Number CPC = NS/EP Call No PRECEDENCE PARAMETER	R-URI/To: = Destination Number RPH [ets.DF]
CdPN = ETS-DN CPC not equal to NS/EP Call No PRECEDENCE PARAMETER	R-URI/To: = ETS-DN RPH [ets.DF]
CdPN = Destination Number CPC not equal to NS/EP Call PRECEDENCE PARAMETER = Look Ahead for Busy = 10 Precedence Level = y (0 – 4) Network Identity = 0100 MLPP Service Domain = z (H'40024B' - H'40024F')	Request for ETS handling is rejected, and the call is processed as an ordinary call.
CdPN = Destination Number CPC = NS/EP Call PRECEDENCE PARAMETER = Look Ahead for Busy = 10 Precedence Level = y (0 – 4) Network Identity = 0100 MLPP Service Domain = z (H'40024B' - H'40024F')	R-URI/To: = Destination Number RPH [(ets.DF, wps.y) or (ets.y, wps.y)]
CdPN = ETS-DN CPC = NS/EP Call PRECEDENCE PARAMETER = Look Ahead for Busy = 10 Precedence Level = y (0 – 4) Network Identity = 0100 MLPP Service Domain = z (H'40024B' - H'40024F')	R-URI/To: = ETS-DN RPH [(ets.DF, wps.y) or (ets.y, wps.y)]

ISUP	SIP
CdPN = ETS-DN CPC not equal to NS/EP Call PRECEDENCE PARAMETER = Look Ahead for Busy = 10 Precedence Level = y (0 – 4) Network Identity = 0100 MLPP Service Domain = z (H‘40024B’ - H‘40024F’) This is abnormal and shall be logged.	R-URI/To: = ETS-DN RPH [ets.DF, wps.y]
The default (DF) priority level is provisioned at the PSTN Gateway and its value is determined by policy. The default priority level is used until the call can be authenticated. The NS/EP Call value for CPC is H‘E2’ Note that if y = 0 then z = H‘40024B’, and if y = 4 then z = H‘40024F’.	

7.2 Receipt of 18X response

The following table (Table 7.12) provides a summary of the interworking of 18X messages to ISUP messages. For further details please see the reference subclause given in each table row.

Table 7.12 - Receipt of 18X Response

←ISUP message	←18X response
ACM or CPG (Note 1)	180 Ringing
ACM or CPG (Note 2)	183 Session Progress
ACM or CPG (Note 3)	182 Queued
NOTE – 1 – See 7.2.1 2 – See 7.2.2 3 – See 7.2.3	

NOTE – Local ISUP procedures may provide for generation of a backward early ACM (no indication) based upon timer expiry. These procedures operate independently of SIP interworking.

7.2.1 Receipt of 180 Ringing

On receipt of a 180 Ringing message, timer T_{OIW2} (if running) is stopped. If a 180 Ringing is received without any encapsulated ISUP message, the IWU shall send either the ACM or CPG message as determined by ISUP procedures related to whether or not an ACM has previously been sent for this call.

For SIP-I, if 180 Session Progress is received with encapsulated ACM or CPG message, the O-IWU shall determine the appropriate backward ISUP message and parameters based on the encapsulated ISUP message and existing ISUP signaling state. Timer T_{OIW2} shall be stopped (if running).

7.2.1.1 Setting for ACM Backwards Call Indicators (mandatory) (No ISUP encapsulation case only)

The table within this subclause presents the default values of Backward Call Indicator parameters that are set by the O-IWU when ACM is sent. Other Backwards Call Indicator parameters are set according to ISUP procedures.

The indicators of the BCI parameters, which are set by the IWU, are as follows:

Table 7.12a

Bits	Indicators in BCI parameter
DC	Called Party's status indicator
I	Interworking indicator
K	ISDN User part indicator
M	ISDN access indicator

For the case in which ISUP encapsulation is not supported, Called Party's status indicator (Bit DC) is set to "subscriber free".

The following default settings may be used:

Table 7.12b

Parameter	Bits	Codes	Meaning
Interworking Indicator	I	1	interworking encountered
ISDN User part indicator	K	0	ISDN user part not used all the way
ISDN access indicator	M	0	terminating access non-ISDN

Otherwise, the O-IWU may set the appropriate values of other BCI indicators (other than Called Party's status indicator) based on analysis of various information such as signaling, internal states and/or local policies.

7.2.1.2 Settings for Event Information (mandatory) in CPG (no ISUP encapsulation case only)

The table within this subclause presents the default values of the Event Information parameter that are set by the O-IWU when CPG is sent. Other indicators in the Event Information parameters are set according to ISUP procedures.

Table 7.12c

Bits	Indicators in Event Information parameter
G F E D C B A	Event indicator

The following code shall be set by the O-IWU in the Event Information parameter field on receipt of 180 Ringing:

Table 7.12d

Bits	Codes	Meaning
G F E D C B A	0 0 0 0 0 0 1	"alerting"

7.2.1.3 Setting for the Optional Backward Call Indicator in ACM or CPG (no ISUP encapsulation case only)

If the O-IWU support the P-Early-Media header as a network option and the 180 Ringing response included a P-Early-Media header authorizing backward and forward early media (i.e., a value of "sendrecv"), then the User-Network Interaction Indicator shall be set to "user-network interaction occurs, cut through in both directions".

7.2.2 Receipt of 183 Session Progress

If 183 Session Progress is received without any encapsulated ISUP message then if the O-IWU support the P-Early-Media header as a network option and the 183 Ringing response included a P-Early-Media header authorizing early media, then an ACM or CPG message shall be sent indicating that early media may be available. As a network option, an ACM or CPG message may be sent indicating that early media may be available when a 183 is received containing SDP. The T_{OIW2} timer, if running, shall be stopped.

In the case of an ACM message being sent, the Backward Call Indicator parameter shall be included with the Called party's status indicator set to "no indication" and an Optional Backward Call Indicators parameter shall be included with the Inband Information Indicators set to "inband information or an appropriate pattern is now available". If the P-Early-Media authorized backward and forward early media (i.e., a value of "sendrecv") then the Optional Backward Call Indicators parameter Network Interaction Indicator shall be set to "user-network interaction occurs, cut through in both directions".

In the case of a CPG message being sent, the Event Information parameter shall be included set to "inband information or an appropriate pattern is now available". If the P-Early-Media authorized backward and forward early media (i.e., a value of "sendrecv") then the Optional Backward Call Indicators parameter Network Interaction Indicator shall be set to "user-network interaction occurs, cut through in both directions".

Otherwise, no ISUP message is sent backward and ISUP procedures should continue.

For SIP-I, if 183 Session Progress is received with encapsulated ISUP message, the O-IWU shall determine the appropriate backward ISUP message based on the encapsulated ISUP message and existing ISUP signaling state. Timer T_{OIW2} shall be stopped in this case.

7.2.3 Receipt of 182 Queued

For an ETS call/session, if a 182 Queued response message is received, the O-IWU shall send either the ACM or CPG message as determined by ISUP procedures related to whether or not an ACM has previously been sent for this call/session.

The Called party's status indicator (bits DC) of the Backward Call Indicators parameter when ACM is sent is set to “excessive delay”.

The Network Excessive Delay Indicator (bit G) of the Optional Backward Call Indicators parameter when CPG is sent is set to “network excessive delay encountered”.

For procedures associated with the expiry of the timer T_{OIW2} for which the O-IWU has previously received a 182 Queued response message, see Section 7.3.

7.3 Expiry of Timer T_{OIW2} & Sending of Early ACM

When timer T_{OIW2} expires, the O-IWU shall return ACM. In the case that the continuity check is performed (ISUP), the O-IWU shall withhold sending ACM until a successful continuity indication has been received.

For cases in which ISUP encapsulation is not supported, the O-IWU shall return awaiting answer indication (e.g., ringing tone) to the calling party.

The Called Party's status indicator (Bit DC) of BCI parameter is set to "no indication". The other BCI indicators shall be set as described in 7.2.1.1.

Upon expiry of timer T_{OIW2} for which the O-IWU has previously received a 182 Queued response message (e.g., for an ETS call/session), the O-IWU shall not send a ringing tone towards the calling party. No ACM or CPG message shall be generated.

7.4 Receipt of Circuit (CIC) Query Response Message

After sending a Circuit (CIC) Query Message, the O-IWU expects to receive a Circuit (CIC) Query Response Message. The O-IWU shall process the Circuit (CIC) Query Response Message as described in ATIS-1000113.2005, chapter 4, clause 2.8.2A. If the ISUP procedures result into release of a call, appropriate actions shall be taken on the SIP side. In the case that ISUP encapsulation is used, the Circuit (CIC) Query Response Message shall not be encapsulated.

7.5 Receipt of 200 OK INVITE

When the O-IWU receives a 200 OK INVITE for this call, it shall stop Timer T_{OIW2} (if running).

In addition, for the case in which no ISUP encapsulation is supported, the O-IWU shall:

1. Send ANM as determined by ISUP procedures.
2. Stop any existing “awaiting answer indication” (e.g., ringing tone).

For SIP-I, if 200 OK INVITE is received with encapsulated ANM message, the O-IWU shall determine the appropriate backward ISUP message and parameters based on the encapsulated ISUP message and existing ISUP signaling state.

7.5.1 Setting of Backwards Call Indicators in the ANM message (for no ISUP encapsulation case only)

When ANM is sent as the first backward message on the ISUP side, the BCI shall be coded as follows:

- The Called Party's status indicator (Bit DC) of BCI parameter is set to "no indication".
- The other BCI indicators shall be set as described in 7.2.1.1.

7.6 Through Connection of ISUP Bearer Path

Through connection of bearer path is applicable only to I-IWUs that are capable of bearer control.

If ISUP encapsulation is not supported and the O-IWU supports the P-Early-Media header, then if a P-Early-Media header is received authorizing backward early media (i.e., a value of "sendonly"), then through connection in the backward direction shall be performed, if not already done. If a P-Early-Media header is received not authorizing early media (i.e., a value of "inactive"), then through connection shall not be performed or removed if already done. If a P-Early-Media header is received authorizing both backward and forward early media (i.e., a value of "sendrecv"), then through connection in both directions shall be performed by applying the following procedures:

- Through connection of the bearer path shall be completed dependent upon whether or not preconditions are in use on the SIP side of the call.
- The bearer path shall be connected in both directions on completion of the bearer setup on the SIP side. This event is indicated by the receipt of SDP answer acceptable to the O-IWU, and an indication that all mandatory preconditions (if any) have been met.

If the P-Early-Media is not supported, or if it is supported, but the header is not received, then if ISUP encapsulation is not supported, through connection at the O-IWU shall follow the ATIS-1000113.2005, chapter 4 procedures for the destination exchange if this functionality is not available at the ASN. If the ASN supports the ATIS-1000113.2005 procedures for through connection at a destination exchange, the O-IWU shall follow the procedures specified for SIP-I.

If ISUP encapsulation is not supported, then as a network option through connection may follow the procedures specified for SIP-I.

For SIP-I, the following procedures shall apply:

- Through connection of the bearer path shall be completed dependent upon whether or not preconditions are in use on the SIP side of the call.
- The bearer path shall be connected in both directions on completion of the bearer setup on the SIP side. This event is indicated by the receipt of SDP answer acceptable to the O-IWU, and an indication that all mandatory preconditions (if any) have been met.
- The bearer path shall be connected in the forward direction no later than on receipt of 200 OK INVITE.

7.6.1 Tone and announcement (backward)

For the case in which ISUP encapsulation is not supported, the following conditions result in ringing tone being played from the O-IWU:

1. 180 Ringing received AND
2. ISUP procedures indicate that ringing tone can be applied AND
3. No valid P-Early-Media header was received authorizing backward media AND
4. A SDP offer/answer exchange establishing a media path has not completed AND
5. The local arrangements assign the role of destination exchange to the O-IWU rather than the associated SIP entity.

When a SDP offer/answer exchange completes for an early dialog, backward through connection of the bearer path shall be completed; ringing tone shall be removed if being played from the O-IWU.

NOTE –

1. It is possible that ringing tone or a progress announcement is already being played as a result of T_{OIW2} expiry. See 7.3.

2. In the case that the associated SIP entity performs the functions of the destination exchange, other tones or announcements may be received from the SIP network.

7.7 Release Procedures at the O-IWU

7.7.1 Receipt of Forward REL

Upon receipt of an ISUP REL:

1. REL received before INVITE has been sent: no action is required on the SIP side other than to terminate local procedures if any are in progress.
2. REL message received before any response has been received to the INVITE.
3. The O-IWU shall hold the REL message until a SIP response has been received. At that point, it shall take action (4) or (5), below, as appropriate.
4. REL message received at O-IWU before a response has been received which establishes a confirmed dialogue or early dialogue:

   The O-IWU shall send a CANCEL request. If the O-IWU subsequently receives a 200 OK INVITE, then it shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent.
5. REL message received at O-IWU after a response has been received which establishes a confirmed dialogue or early dialogue:

   The O-IWU shall send a BYE request. For cases in which no encapsulation is used, for an early dialog only, CANCEL may be used instead.

For SIP-I, if a BYE message is sent, it shall encapsulate the received REL message.

If the REL message is received after the 200 OK INVITE but before the outgoing side of the O-IWU has sent the ACK, then the O-IWU shall send the ACK before sending a BYE.

Depending on local policy, a Reason header field containing the received (Q.850 or ANSI-specific) cause value of the REL message may be added to the CANCEL or BYE. If the Coding standard field of the Cause indicators parameter is set to "ANSI Standard", the Protocol parameter of the Reason header is set to "ANSI", and if the Coding standard is set to "ITU-T Standard", the Protocol parameter is set to "Q.850". The mapping of the Cause Indicators parameter to the Reason header is shown in Table 19 (see 6.13.2).

7.7.2 Receipt of Backward BYE

On receipt of SIP BYE, the O-IWU shall send an ISUP REL to the ISUP side.

In the case of SIP-I, the encapsulated REL shall be passed to ISUP procedures without modification.

For the no ISUP encapsulation case:

If a Reason header with a Protocol parameter set to either "Q.850" or "ANSI" is included in the BYE, then the appropriate cause value may be mapped to the cause value field in the REL depending on local policy. The mapping of the Reason header to the Cause Indicators parameter is shown in Table 6.16 (see 6.13.1). Table 7.13 shows the coding of the cause value in the REL if it is not available from the Reason header field.

Table 7.13 - Release from SIP side at O IWU

←REL cause parameter	←SIP Message
Cause value No. 16 (normal clearing)	BYE

7.7.3 Autonomous Release at O-IWU

Table 7.14 shows the trigger events at the IWU and the release initiated by the IWU when the call is traversing from ISUP to SIP.

If, after answer, ISUP procedures result in autonomous REL from the IWU then a BYE shall be sent on the SIP side.

Depending on local policy, a Reason header field containing the (Q.850 or ANSI) Cause Value of the REL message sent by the O-IWU may be added to the SIP Message (BYE or CANCEL) to be sent by the SIP side of the O-IWU.

Table 7.14 - Autonomous Release at O IWU

REL ← cause parameter	Trigger event	→ SIP
As determined by ISUP procedure.	COT received with the Continuity Indicators parameter set to “continuity check failed” (ISUP only) or the ISUP timer T8 expires.	Send CANCEL or BYE according to the rule described in 7.7.1.
REL with cause value 47 (resource unavailable, unspecified).	Internal resource reservation unsuccessful	As determined by SIP procedure
As determined by ISUP procedure.	ISUP procedures result in generation of autonomous REL on ISUP side.	CANCEL or BYE, according to the rule described in 7.7.1.
Depending on the SIP release reason.	SIP procedures result in a decision to release the call.	As determined by SIP procedure.

7.7.4 Receipt of RSC, GRS, or CGB (ISUP)

Table 7.15 shows the message sent by the O-IWU upon receipt of an ISUP RSC message, GRS message, or CGB message with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented".

On receipt of a GRS or CGB message, one SIP message is sent for each call association. Therefore, multiple SIP messages may be sent on receipt of a single GRS or CGB message.

The O-IWU shall send CANCEL or BYE according to the rule described in 7.7.1.

Depending on local policy, a Reason header field containing the (Q.850 or ANSI) Cause Value of the REL message sent by the O-IWU may be added to the SIP message (BYE or CANCEL) to be sent by the SIP side of the O-IWU.

In the case that ISUP encapsulation is being used, the RSC, GRS, or CGB ISUP messages shall not be encapsulated within the SIP BYE or CANCEL, but if a BYE request is sent, it shall encapsulate the REL message that would be sent towards a forward ISUP node.

Table 7.15 - Receipt of RSC, GRS, or CGB messages (ISUP) at O-IWU

Message received from ISUP →	SIP →
Reset Circuit Message (RSC)	CANCEL or BYE
Circuit Group Reset Message (GRS)	CANCEL or BYE
Circuit Group Blocking Message (CGB) with the circuit group supervision message type indicator coded *"hardware failure oriented"*	CANCEL or BYE

7.7.5 Receipt of 3XX, 4XX, 5XX, 6XX Response INVITE

Receipt of 3xx Redirection, if received in a response as part of a valid dialog, is handled according to the SIP protocol, resulting in invocation of the local routing function. The remainder of this section applies only to the 4XX, 5XX, and 6XX response cases.

If a Reason header is included in a 4XX, 5XX, 6XX, then the Cause Value of the Reason header should be mapped to the ISUP Cause Value field in the ISUP REL message. The mapping of the Reason header to the Cause Indicators parameter is shown in Table 6.16 (see 6.13.1). Otherwise, the mapping from status code to Cause Value on receipt of a 4XX, 5XX, or 6XX final response to the INVITE on the SIP side is described within Table 7.16.

For SIP-I, if an encapsulated REL is received, it shall be passed to ISUP procedures without modification. In all other cases the procedures in the remainder of this clause apply.

In all cases where SIP itself or subclauses to this section specify additional SIP side behavior related to the receipt of a particular INVITE response, these procedures should be followed in preference to the immediate sending of a REL message to ISUP.

If there are no SIP side procedures associated with this response, the REL shall be sent immediately.

> NOTE – Depending upon the SIP side procedures applied at the O-IWU it is possible that receipt of certain 4xx/5xx/6xx responses to an INVITE may in some cases not result in any REL message being sent to the ISUP network. For example, if a 401 Unauthorized response is received and the O-IWU successfully initiates a new INVITE containing the correct credentials, the call will proceed.

If no further reference is given in the "Remarks" column, then this means that the SIP response is interworked to an ISUP REL message sent on the incoming ISUP side of the O-IWU with the Cause Value indicated within the table. In cases where further reference is indicated, the behavior of the O-IWU is described within the referred-to clause; however the table indicates the "eventual" behavior of the O-IWU in the case that further measures taken on the SIP side of the call (to try to sustain the call) fail resulting in the ISUP half call being released by sending a REL message with the Cause Value indicated.

When the response to the INVITE results in the ISUP REL message with cause 127 "*Interworking*" being sent, then the location should be set to (BI) "*network beyond interworking point*".

Table 7.16 - Receipt of 4XX, 5XX, or 6XX at O IWU

←REL (cause code)	←4XX/5XX/6XX SIP Message	Remarks
111 (Protocol error, unspecified)	400 Bad Request	
127 (Interworking, unspecified)	401 Unauthorised	Note 1
127 (Interworking, unspecified)	402 Payment Required	
No 79 (Service or option not implemented, unspecified)	403 Forbidden	
1 (Unallocated number)	404 Not Found	
127 (Interworking, unspecified)	405 Method Not Allowed	
127 (Interworking, unspecified)	406 Not Acceptable	
127 (Interworking, unspecified)	407 Proxy authentication required	Note 1
102 (Recovery on timer expiry)	408 Request Timeout	
22 (Number changed (without diagnostic))	410 Gone	
127 (Interworking, unspecified)	413 Request Entity too long	Note 1
111 (Protocol error, unspecified)	414 Request-uri too long	Note 1
127 (Interworking, unspecified)	415 Unsupported Media type	Note 1
111 (Protocol error, unspecified)	416 Unsupported URI scheme	Note 1
111 (Protocol error, unspecified)	420 Bad Extension	Note 1
111 (Protocol error, unspecified)	421 Extension required	Note 1
31 (Normal, unspecified)	422 Session Interval Too Small	
127 (Interworking, unspecified)	423 Interval Too Brief	
127 (Interworking, unspecified)	440 Max-Breadth Exceeded	
20 (Subscriber absent)	480 Temporarily Unavailable	
127 (Interworking, unspecified)	481 Call/Transaction does not exist	
127 (Interworking, unspecified)	482 Loop Detected	
25 (Exchange routing error)	483 Too many hops	
28 (Invalid Number format)	484 Address Incomplete	Note 1
1 (Unallocated (unassigned) number)	485 Ambiguous	
17 (User busy)	486 Busy Here	
127 (Interworking, unspecified) or no mapping (Note 3)	487 Request terminated	Note 2
50 (Requested facility not subscribed)	488 Not acceptable here	
No mapping.	491 Request Pending	Note 2
127 (Interworking, unspecified)	493 Undecipherable	
127 (Interworking, unspecified)	500 Server Internal error	
79 (Service or option not implemented, unspecified)	501 Not implemented	
27 (Destination out of order)	502 Bad Gateway	
127 (Interworking, unspecified)	503 Service Unavailable	Note 1
102 (Recovery on timer expiry)	504 Server timeout	
127 (Interworking, unspecified)	505 Version not supported	Note 1
127 (Interworking, unspecified)	513 Message too large	Note 1
127 (Interworking, unspecified)	580 Precondition failure	Note 1
17 (User busy)	600 Busy Everywhere	
21 (Call rejected)	603 Decline	
2 (No route to specified transit network)	604 Does not exist anywhere	
88 (Incompatible destination)	606 Not acceptable	
NOTE – 1 – This response may be handled entirely on the SIP side; if so, it is not interworked. 2 – This response does not terminate a SIP dialog, but only a specific transaction within it.		

←REL (cause code)	←4XX/5XX/6XX SIP Message	Remarks
3 – No mapping if the O-IWU previously issued a CANCEL request for the INVITE.		

7.7.5.1 Special handling of 580 Precondition Failure Received in Response to Either an INVITE or UPDATE

A 580 Precondition failure response may be received as a response either to an INVITE or to an UPDATE request.

7.7.5.1.1 580 Precondition Failure Response to an INVITE.

Release with cause code as indicated in Table 7.16 is sent immediately to the ISUP network.

7.7.5.1.2 580 Precondition Failure Response to An UPDATE Within An Early Dialog

Release with Cause Code '127 Interworking' is sent immediately to the ISUP network. A BYE request is sent for the INVITE transaction within which the UPDATE was sent.

8 Timers

Table 8.1 summarizes the interworking timers introduced in clause 7.

Table 8.1 - Interworking Timers

Symbol	Time-out value	Cause for initiation	Normal termination	At expiry	Reference
T_{OIW2}	4-14 seconds (default of 4 seconds)	Sending of INVITE unless the ACM has already been sent.	On reception of 180 Ringing, 183 Session Progress with encapsulated ACM or 200 OK INVITE	Send early ACM. For the no ISUP encapsulation case, send the awaiting answer indication (e.g., ring tone) or appropriate progress announcement to the calling party.	7.1, 7.2.1, 7.3.1, 7.5.

Annex A: Interworking for ISDN & Non-ISDN supplementary services

(normative)

This Annex describes service interworking of ISDN and non-ISDN supplementary services between SIP and ISUP.

Except where otherwise stated, services in encapsulated ISUP operation use the parameters of the (de)encapsulated ISUP, and no other interworking is required. *Accordingly, the service interworking descriptions below are only for non-encapsulated ISUP operation unless encapsulated ISUP operation is specifically indicated.*

A.1 Interworking of ISDN CLIP/CLIR Supplementary Service to SIP Networks

A.1.1 Operation without Encapsulated ISUP

In accordance with the procedures described within ATIS-1000625.1993(R2008) and ATIS-1000625.a.1998(R2008), the CLIP/CLIR services are only to be interworked between trusted nodes - that is before passing any CLIP/CLIR information over the SIP/ISUP boundary the IWU must satisfy itself that the nodes to which the information is to be sent are trusted.

The interworking between the Calling Party Number and the P-Asserted-Identity header and vice versa used for the CLIP-CLIR service is defined in 6.1.3.6 and 7.1.3. This interworking is essentially the same as for basic call and differs only in that if the CLIR service is invoked the Address Presentation Restriction Indicator (APRI) (in the case of ISUP to SIP calls) or the priv-value of the "calling" Privacy header field (in the case of SIP to ISUP calls) is set to the appropriate "restriction/privacy" value.

In the specific case of ISUP originated calls, use of the CLIP service additionally requires the ability to determine whether the number was network provided or provided by the access signaling system. Due to the possible SIP indication of the P-Asserted-Identity the Screening Indicator is set to *"network provided"* as default. For the CLIP-CLIR service, the mapping of the APRI is described within 6.1.3.6 and 7.1.3.

At the O-IWU the *"presentation restricted"* indication shall be mapped to the Privacy header field with priv-value containing *“id”* and *“header”*.

A.1.2 SIP-I

- *At the O-IWU:* The service shall be supported by encapsulation.
- *At the I-IWU:* If the address within the Calling Party Number after application of the interworking rules in 6.1.3.6 and processing by ISUP procedures is the same as the value contained in the encapsulated ISUP, no additional interworking is needed beyond use of ISUP encapsulation. In the contrary case the Calling Party Sub-address is deleted from the ATP.

A.2 Interworking of COLP/COLR Supplementary Service to SIP Networks

No procedures defined in this version of this document.

A.3 Interworking of Direct-Dialing-In (DDI) Supplementary Service to SIP Networks

No procedures defined in this version of this document.

A.4 Interworking of Malicious Call Identification (MCID) Supplementary Service to SIP networks

No procedures defined in this version of this document.

A.5 Interworking of Sub-addressing (SUB) Supplementary Service to SIP Networks

No procedures defined in this document.

A.6 Interworking of Non-ISDN Call Forwarding Busy (CFB)/ Call Forwarding No Reply (CFNR) / Call Forwarding Unconditional (CFU) Supplementary Services to SIP Networks

A.6.1 General

Call forwarding services between ISUP and SIP are specified. Two methods are defined making use of two different SIP headers: History-Info specified by IETF RFC 4244 and Diversion header documented in IETF RFC 5806. When signaling call forwarding information via SIP messages, the Diversion header and/or the History-Info header may be included. Either the Diversion header or History-Info header may be present in a received SIP message. The information in a received History-Info header will take precedence over the information in a received diversion header when local policy allows the use of both headers.

The interworking procedures for the History-Info method are specified by B.6.1. The interworking procedures for the Diversion header are specified by B.6.2.

A.6.2 History-Info Header Method

A.6.2.1 General

Operation without encapsulated ISUP: Interworking defined by the following subsections shall be applied.

Operation with encapsulated ISUP: Interworking defined by the following subsections shall be applied in addition to the encapsulation of the ISUP message.

A.6.2.2 Interworking at the O-IWU

A.6.2.2.1 General

This subsection describes the optional mapping of Call Forwarding information at the O-IWU to the protocol-cause specified in IETF RFC 3326.

A.6.2.2.2 Interworking SIP to ISUP

Table A.1 - Mapping of SIP messages to ISUP messages

←Message sent to ISUP	←Message Received from SIP	
ACM indicating call forwarding	181 (Call Is Being Forwarded) response	See table A.3
CPG indicating call forwarding (see NOTE)	181 (Call Is Being Forwarded) response	See table A.4
ACM indicating subscriber free	180 (Ringing) response	See table A.5
CPG indicating Alerting (see NOTE)	180 (Ringing) response	See table A.6
ANM	200 (OK) response	
NOTE – A CPG will be sent if an ACM was already sent.		

Table A.2 - Mapping of hi-targeted-to-uri to ISUP Event Information

Source SIP header field and component	Source Component value	Event Information	Derived value of parameter field
		Event indicator	Shall be set to ALERTING if mapped from a 180 (Ringing)
hi-targeted-to-uri; Reason header as defined in IETF RFC 4244 with cause parameter	486		Call forwarded on busy
	408		Call forwarded on no reply
	404		Progress
	302		Call forwarded unconditional
	All other values		Call forwarded unconditional

Table A.3 - Mapping of 181 (Call Is Being Forwarded) → ACM

Source SIP header field and component	Source Component value	ISUP Parameter	Derived value of parameter field
181 (Call Is Being Forwarded)		ACM	
		Backward call indicators – called party's status indicator	No indication
		Notification indicator	Call is forwarded/deflected

Table A.4 - Mapping of 181 (Call Is Being Forwarded) → CPG if ACM was already sent

Source SIP header field and component	Source Component value	ISUP Parameter	Derived value of parameter field
181 (Call Is Being Forwarded) response		CPG	
		Event information	See table A.2
		Notification indicator	Call is forwarded/deflected

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

Source SIP header field and component	Source Component value	ISUP Parameter	Derived value of parameter field
180 (Ringing) response		ACM	
		Backward call indicator – called party's status indicator	Subscriber free
History-Info header with call forwarding entry	If hi-index indicates that there is a call forwarding.	Notification indicator	Call is forwarded/deflected

The mapping described within table Table A.1 can only appear if the communication has already undergone a Call Forwarding in the ISDN/PSTN and the 180 is the first provisional response sent in backward direction.

The IWU can indicate the call diversion in the mapping of 180 (Ringing) to CPG in fact if the response before was a 181 (Call is being forwarded).

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

Source SIP header field and component	ISUP Parameter	Derived value of parameter field
180 (Ringing) response	CPG	
	Event Information – Event indicator	ALERTING
History-Info header with call forwarding entry	Notification indicator	Call is forwarded/deflected

The mapping in table A.6 appears when a 181 previously was mapped to an ACM. Therefore the state machine of the IWU knows that a call forwarding is in progress.

A.6.2.2.3 Interworking ISUP to SIP

For the interworking of 180 (Ringing) response and 200 (OK) response to the regarding ISUP messages and parameters no additional procedures beyond the basic call procedures are needed.

To interwork the redirection number at the O-IWU it can be needed to create placeholder History entries. Such a History entry has to provide a hi-targeted-to-uri with a placeholder value "unknown@unknown.invalid" a Cause parameter and a hi-index as described within Table A.7.

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

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
IAM		INVITE request	
Redirecting number		History-Info header field	hi-targeted-to-uri of the penultimate index entry IF more than 2 Index entries are included ELSE no mapping
Nature of address indicator:	"*national (significant) number*"	hi-targeted-to-uri	Add CC (of the country where the IWU is located) to Redirecting Number signals then map to user portion of URI scheme used. **Addr-spec** "+" CC NDC SN mapped to user portion of URI scheme used
	"international number"		Map complete Redirection number Address Signals to user portion of URI scheme used.
Address Signals	If NOA is "*national (significant) number*" then the format of the Address Signals is: NDC + SN If NOA is "*international number*" then the format of the Address Signals is: CC + NDC + SN	hi-targeted-to-uri	"+" CC NDC SN mapped to userinfo portion of URI scheme used
APRI	"presentation restricted"	Privacy header field of the first hi-targeted-to-uri entry of History-Info header	"history"
	"presentation allowed"		"none"

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
Redirection Information	Redirection counter 1	hi-index	Number of forwardings are shown due to the number of hi-index Entries Index for original called Party Number = 1 Address Signals (CdPN) Number = 1.1
	2		Index for original called Party Number = 1 Index for Redirecting number with Index = 1.1 Address Signals (CdPN) Number = 1.1.1
	N		Index for Original Called Party Number = 1 Placeholder History entry with Index = 1.1 … Fill up … Index for Redirecting Number with = 1+[(N-1)*".1"] Index for Address Signals (CdPN) = 1+N* ".1" (e.g., N=3 → 1.1.1.1)
Redirection Information	Redirecting Reason and Original Redirecting Reason (NOTE 1)	hi-targeted-to-uri; Reason header as defined in IETF RFC 4244 with cause parameter. For a placeholder History entry the value "404" shall be taken (NOTE 2). Cause parameter for redirecting reason will be put in the entry of redirecting number, and cause parameter for original redirecting reason will be put in the entry of original called party number.	Cause parameter value
	unknown/not available		404
	unconditional		302
	User Busy		486
	No reply		408
	Deflection		302

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
Called Party Number	*See* Redirecting number	History-Info header field see hi-targeted-to-uri	URI of the last hi-targeted-to-uri entry of History-Info header field
Original Called Party Number	*See* Redirecting number	History-Info header field see hi-targeted-to-uri	URI of first hi-targeted-to-uri entry of History-Info header field
Original Called Party Number	APRI	Privacy header field of the first hi-targeted-to-uri entry of History-Info header	Priv-value
	"presentation restricted"		"*history*"
	"presentation allowed"		"*none*"

NOTE 1 –

Original Redirecting Reason in ANSI ISUP supports the same values as defined for the Redirecting Reason parameter. In ITU-T ISUP, only the value "unknown/not available" is defined and the remaining values are for national use.

NOTE 2 –

For all History entries except the last one a cause parameter in Reason header as defined in IETF RFC 4244 has to be included.

A.6.2.3 Interworking at the I-IWU

A.6.2.3.1 General

This subsection describes the interworking of the Call Forwarding information at the I-IWU.

A.6.2.3.2 Interworking from SIP to ISUP

Table A.8 - Mapping of SIP to ISUP messages

→Message received from SIP	→Message send to ISUP
INVITE request	IAM

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

Source SIP header field and component	Source Component value	Redirecting number	Derived value of parameter field
latest History-Info header field entry containing a Reason header as defined in IETF RFC 4244 with cause parameter value as listed in the cause parameter row in Table A.2 (Note 1)		Redirecting number	
hi-targeted-to-uri appropriate global number portion of the URI, assumed to be in form "+" CC + NDC + SN	CC	Nature of address indicator	If CC is equal to the country code of the country where IWU is located AND the next ISUP node is located in the same country, then set to "national (significant) number" else set to "international number"
	CC, NDC, SN	Address signals	If NOA is "national (significant) number" then set to NDC + SN. If NOA is "international number" then set to CC + NDC + SN
Privacy header field, priv-value component in History-Info header field as specified in this table(NOTE 2)	"history" or “session” or “header”	APRI	"presentation restricted"
	Privacy header field absent or "none"		"presentation allowed"
NOTE 1 – If it is SIP URI and doesn’t contain “user=phone”, mapping to redirecting number is impossible, therefore no need to generate Redirecting number NOTE 2 – It is possible that an entry of the History-Info header field itself is marked as restricted or the whole History-Info header.			

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

Source SIP header field and component	Source Component value	Redirection Information	Derived value of parameter field
Cause parameter in the last hi-targeted-to-uri containing a Reason header as defined in IETF RFC 4244	**Cause parameter value**	Redirecting Reason	***Redirecting Reason***
	302		Deflection
	486		User busy
	408		No reply
	All other values		unknown
Hi-index		Redirection counter	number of History entries containing a cause-param with value as listed in the cause-param row in this table
Cause parameter in the first hi-targeted-to-uri containing a Reason header as defined in IETF RFC 4244	**Cause parameter value**	Original redirecting reason	**Original redirecting reason**
	302		Deflection
	486		User busy
	408		No reply
	All other values		unknown

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

Source SIP header field and component	Source Component value	Original called number	Derived value of parameter field
		Numbering Plan Indicator	"*ISDN (Telephony) numbering plan (Recommendation E.164)*"
hi-targeted-to-uri of 1st hi-targeted-to-uri containing a Reason header as defined in IETF RFC 4244 with cause parameter; appropriate global number portion of the URI, assumed to be in form "+" CC + NDC + SN(NOTE 1)	CC	Nature of address indicator	If CC is equal to the country code of the country where IWU is located AND the next ISUP node is located in the same country, then set to "national (significant) number" else set to "international number"
	CC, NDC, SN	Address signals	If NOA is "national (significant) number" then set to NDC + SN. If NOA is "international number" then set to CC + NDC + SN
priv-value component in History-Info header field of the History-Info header field entry as defined above in this table (NOTE 2)	"history" or "session" or "header"	APRI	"presentation restricted"
	Privacy header field absent or "none"		"presentation allowed"

NOTE 1 –

If it is SIP URI and doesn't contain "user=phone", mapping to Original Called number is impossible, therefore no need to generate Original Called number

NOTE 2 –

It is possible that an entry of the History-Info header field itself is marked as restricted or the whole History-Info header.

Table A.12 - Mapping of INVITE to IAM

INVITE		IAM	
History-Info header field	See table A.9	Redirecting number	See table A.9
History-Info header field	See table A.10	Redirection Information	See table A.10
Cause parameter in the last hi-targeted-to-uri containing a Reason header as defined in IETF RFC 4244	Cause parameter value	Redirection Information	Redirecting Reason
	486		User busy
	408		No reply
	302		Deflection
	All other values		unknown
History-Info header field	See table A.11	Original Called Number	*See* table A.11

A.6.2.3.3 Interworking from ISUP to SIP

Table A.13 - Mapping of ISUP to SIP Messages

←Message sent to SIP	←Message Received from ISUP	
181 (Being forwarded)	ACM no indication with Notification Information (call is forwarded/deflected)	See table A.9
180 (Ringing)	ACM indicating subscriber free	See table A.10
181 (Being forwarded)	CPG with Event Information indicating call is forwarded or with Notification Information (call is forwarded/deflected)	See table A.11
180 (Ringing)	CPG indicating alerting	See table A.12
200 (OK)	ANM	

A received CPG shall be mapped to a 180 (Ringing) response if the Event Information indicates an Alerting is due to the mapping rule defined within the basic call.

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

ISUP Parameter	Derived value of parameter field	SIP component	Value
Backward Call Indicators – Called party's status indicator	No indication		
Notification indicator	Call is forwarded/deflected		
		History-Info header field with two hi-entries	placeholder address ("unknown@unknown.invalid") for both entries
		Cause parameter in the 2nd latest entry	Cause parameter value
			404

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

ISUP Parameter	Derived value of parameter field	SIP component	Value
Backward Call Indicators – Called party's status indicator	Subscriber free		
Notification indicators	Call is diverting		
		History-Info header	See Table A.9

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

ISUP Parameter	Derived value of parameter field	SIP component	Value
Event Information		History-Info header field with two hi-entries	placeholder address ("unknown@unknown.invalid") for both entries
		History-Info header Cause parameter in the 2nd latest entry	
	Progress and Notification indicator set to "Call is forwarded/deflected"		404
	Call forwarded on busy		486
	Call forwarded on no reply		408
	Call forwarded unconditional		302

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

ISUP Parameter	Derived value of parameter field	SIP component	Value
Event Information	Alerting		
Notification indicator	Call is forwarded/deflected	History-Info header	See table A.9

A.6.3 Diversion Header Method

A.6.3.1 General

Operation without encapsulated ISUP: Interworking defined by the following subsections shall be applied.

Operation with encapsulated ISUP: Interworking defined by the following subsections shall be applied in addition to the encapsulation of the ISUP message.

A.6.3.2 Interworking at the O-IWU

A.6.3.2.1 General

This sub-section describes the optional mapping of Call Forwarding information at the O-IWU.

A.6.3.2.2 Interworking SIP to ISUP

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

←Message sent to ISUP	←Message Received from SIP
ACM - BCI , Called party's status = "No indication" Notification indicator = "Call is forwarded/deflected"	181 (Call Is Being Forwarded) response
CPG – Event information = "Progress" Notification indicator = "Call is forwarded/deflected"	181 (Call Is Being Forwarded) response
ACM – BCI , Called party's status = "Subscriber free" Notification indicator = "Call is forwarded/deflected"	180 (Ringing) response
CPG – Event information = "Alerting" Notification indicator = "Call is forwarded/deflected"	180 (Ringing) response
NOTE – A CPG will be sent if an ACM was already sent.	

A.6.3.2.3 Interworking ISUP to SIP

For the interworking of 18x response and 200 response to the regarding ISUP messages and parameters no additional procedures beyond the basic call procedures are needed.

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

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
Redirecting number		Top Diversion header	Top Diversion header IF Redirection counter is greater than 1 ELSE no mapping
Nature of address indicator:	"*national (significant) number*"	Top Diversion header name-addr	Add CC (of the country where the IWU is located) to Redirecting Number Address Signals then map to user portion of URI scheme used. "+" CC NDC SN mapped to user portion of URI scheme used
	"*international number*"		Map complete Redirection number Address Signals to user portion of URI scheme used.

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
Address Signals	If NOA is "*national (significant) number*" then the format of the Address Signals is: NDC + SN If NOA is "*international number*" then the format of the Address Signals is: CC + NDC + SN	Top Diversion header name-addr	"+" CC NDC SN mapped to userinfo portion of URI scheme used
Redirecting Number	"presentation restricted"	Top Diversion header privacy	"*full*"
	"presentation allowed"		"off" or omitted
Original called party number		Bottom Diversion header	Bottom Diversion header
Nature of address indicator: Address Signals	"*national (significant) number*"	Bottom Diversion header name-addr	Add CC (of the country where the IWU is located) to Original called party number Address Signals then map to user portion of URI scheme used. "+" CC NDC SN mapped to user portion of URI scheme used
	"international number"		Map complete Redirection number Address Signals to user portion of URI scheme used.
	If NOA is "*national (significant) number*" then the format of the Address Signals is: NDC + SN If NOA is "*international number*" then the format of the Address Signals is: CC + NDC + SN		"+" CC NDC SN mapped to userinfo portion of URI scheme used
Original called party number APRI	"presentation restricted"	Bottom Diversion header privacy	"*full*"
	"presentation allowed"		"off" or omitted

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
Redirection Information	Redirection counter 1	Diversion header(s) counter	Only Diversion header counter = 1 (or omitted)
	2		Bottom Diversion header (original called number) counter = 1 (or omitted) Top Diversion header (redirecting number) counter = 1 (or omitted)
	N		Bottom Diversion header (original called number) counter = 1 (or omitted) Top Diversion header (redirecting number) counter = (Redirection counter – 1)
Redirection Information	Original Redirecting Reason	Bottom Diversion header (original called number) reason	Reason parameter value
	unknown		"unknown"
	unconditional		"unconditional"
	user Busy		"user-busy"
	no reply		"no-answer"
	deflection		"deflection"
Redirection Information	Redirecting Reason	Top Diversion header (redirecting number) reason	Reason parameter value
	unknown		"unknown"
	unconditional		"unconditional"
	user Busy		"user-busy"
	no reply		"no-answer"
	deflection		"deflection"

A.6.2.3 Interworking at the I-IWU

A.6.2.3.1 General

This subsection describes the interworking of the Call Forwarding information at the I-IWU.

A.6.3.3.2 Interworking from SIP to ISUP

Table A.20 - Mapping of INVITE to IAM

Source SIP header field and component	Source Component value	ISUP parameter/field	Derived value of parameter field
Top Diversion header (Note 1) (Note 2)		Redirecting number	
Top Diversion header name-addr global number portion of the URI, assumed to be in form "+" CC + NDC + SN	CC	Nature of address indicator	If CC is equal to the country code of the country where IWU is located AND the next ISUP node is located in the same country, then set to "national (significant) number" else set to "international number"
	CC, NDC, SN	Address signals	If NOA is "national (significant) number" then set to NDC + SN. If NOA is "international number" then set to CC + NDC + SN
Top Diversion header privacy field as specified in this table.	"full" or “uri”	APRI	"presentation restricted"
	Other value		"presentation allowed"
Bottom Diversion header (Note 1) (Note 2)		Original called party number	
Bottom Diversion header name-addr global number portion of the URI, assumed to be in form "+" CC + NDC + SN	CC	Nature of address indicator	If CC is equal to the country code of the country where IWU is located AND the next ISUP node is located in the same country, then set to "national (significant) number" else set to "international number"
	CC, NDC, SN	Address signals	If NOA is "national (significant) number" then set to NDC + SN. If NOA is "international number" then set to CC + NDC + SN
Bottom Diversion header privacy field as specified in this table.	"full" or “uri”	APRI	"presentation restricted"
	Other value		"presentation allowed"
Diversion header(s) reason		Redirecting Information	
Reason parameter in the bottom Diversion header	"deflection"	Original Redirecting Reason	deflection
	"user-busy"		user busy
	"no answer"		no reply

Source SIP header field and component	Source Component value	ISUP parameter/field	Derived value of parameter field
	"unconditional"		unconditional
	All other values		unknown
Reason parameter in the top Diversion header	"deflection"	Redirecting Reason	deflection
	"user-busy"		user busy
	"no answer"		no reply
	"unconditional"		unconditional
	All other values		unknown
	"deflection"		deflection
Diversion header counter field(s)		Redirection counter	Sum of all Diversion header counter fields. If a Diversion header does not include a counter, then default a value of 1 for that header.
NOTE 1 – If it is SIP URI and doesn't contain "user=phone", mapping to redirecting number is impossible, therefore no need to generate Redirecting number. NOTE 2 – In the event that there is only a single Diversion header, that header is considered both the top and bottom header in terms of interworking.			

A.6.3.3.3 Interworking from ISUP to SIP

Table A.21 - Mapping of ISUP to SIP Messages

←Message sent to SIP	←Message Received from ISUP
181 (Being forwarded) - no Diversion SIP headers included	ACM BCI = "no indication" with Notification Information = "call is forwarded/deflected"
181 (Being forwarded) -no Diversion SIP headers included	CPG Event Information = "call forwarded on busy", "call forwarded on no reply", "call forwarded unconditional" Or Event information = "progress" and Notification Information = "call is forwarded/deflected"

A.7 Interworking of ISDN Call Deflection (CD) Supplementary Service to SIP Networks

- *Operation without encapsulated ISUP:* In accordance with the procedures described within ATIS-1000642.1995(R2009), recognizing that this standard does not provide a mapping for the ISUP Access Transport Parameter, the aspects of the service supported by this parameter terminate at the IWU. The call shall continue according to the basic call procedures.
- *Operation with encapsulated ISUP:* No additional interworking beyond use of ISUP encapsulation.

A.8 Interworking of ISDN Explicit Call Transfer (ECT) Supplementary Service to SIP Networks

- *Operation without encapsulated ISUP:* ATIS-1000643.1998(R2008) specifies the use of the ISUP Notification Indicator parameter with values that are not mapped to SIP in this standard. Therefore, the service shall be terminated at the IWU and the call shall continue according to the basic call procedures.
- *Operation with encapsulated ISUP:* No additional interworking beyond use of ISUP encapsulation.

A.9 Interworking of ISDN Call Waiting (CW) Supplementary Service to SIP Networks

- *Operation without encapsulated ISUP:* ATIS-1000613.1991(R2012) specifies the use of the ISUP Notification Indicator parameter with value 'Call is a waiting call' and this value is not mapped to SIP in this standard. Therefore, the service shall be terminated at the IWU and the call shall continue according to the basic call procedures.
- *Operation with encapsulated ISUP:* No additional interworking beyond use of ISUP encapsulation.

A.10 Interworking of ISDN Call Hold (HOLD) Supplementary Service to SIP Networks

- *Operation without encapsulated ISUP:* Call Hold is defined as an ISUP supplementary service within ATIS-1000616.1992(R2009).

  A call may be placed on hold by the calling user at any time after the call has been answered or additionally as a service provider option:

  1) After alerting has commenced; or
  2) After the calling user has provided all of the information necessary for processing the call.

  A call may be placed on hold by the called user, at any time after the call has been answered and before call clearing has begun.

  For the Call Hold supplementary service, the Call Progress message containing the Notification Indicator parameter is used to send the appropriate notification towards the remote party.

  The following notification descriptions are used:

  - *"remote hold"*
  - *"remote hold released"*

  The Event Indicator is set to "notification for supplementary service".

  The same service is also available within SIP networks and is defined in RFC 3264. If a party in a call wants to put the other party "on hold" (i.e., request that it temporarily stops sending one or more unicast media streams), a party offers the other an updated SDP. The stream to be placed on hold will be marked with the following attribute:

  - "a=sendonly", if the stream was previously a sendrecv media stream
  - "a=inactive", if the stream was previously a recvonly media stream
- If the party wants to retrieve the call, then the stream to be retrieved will be marked as:
  - "a=sendrecv", if the stream was previously a sendrecv media stream, or the attribute may be omitted, since sendrecv is the default
  - "a=recvonly", if the stream was previously an inactive media stream

The mapping between the ISUP and SIP flows is shown in Table A.22.

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

Call state	ISUP message	Mapping	SIP message
Answered	CPG with *"remote hold"*	<==>	INVITE with the attribute line "a=sendonly" or "a=inactive" for the offered media stream (see above).
Answered	CPG with *"remote hold released"*	<==>	INVITE with the attribute line "a=sendrecv", or omitted attribute line, or "a= recvonly" for the offered media stream (see above)
before answer	CPG with *"remote hold"*	<==>	UPDATE with the attribute line "a=sendonly" or "a=inactive" for the offered media stream (see above).
before answer	CPG with *"remote hold released"*	<==>	UPDATE with the attribute line "a=sendrecv", or omitted attribute line, or "a= recvonly" for the offered media stream (see above)
Mapping: <==> : Mapping in both directions, i.e., from ISUP to SIP and vice versa : Mapping from ISUP to SIP only NOTE – For the "before answer" scenarios mapping applies only for the hold requests sent by the calling party to the called party as the called party cannot put the calling party on hold before answer.			

- *Operation with encapsulated ISUP:* Interworking is via the encapsulated CPG message. No additional interworking is required.

The mapping between the ISUP and SIP-I flows is shown in Table A.23.

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

Call state	ISUP message	Mapping	SIP message
Answered	CPG with *"remote hold"* CPG with *"remote hold"* extracted from the body of the SIP message	==> <==	INVITE with the attribute line "a=sendonly" or "a=inactive" for the offered media stream (see above) and encapsulated ISUP CPG message
Answered	CPG with *"remote hold released"* CPG with *"remote hold released"* extracted from the body of the SIP message	==> <==	INVITE with the attribute line "a=sendrecv", or omitted attribute line, or "a= recvonly" for the offered media stream (see above) and encapsulated ISUP CPG message
before answer	CPG with *"remote hold"* CPG with*"remote hold"* extracted from the body of the SIP message	==> <==	UPDATE with the attribute line "a=sendonly" or "a=inactive" for the offered media stream (see above) and encapsulated ISUP CPG message
before answer	CPG with *"remote hold released"* CPG with *"remote hold released"* extracted from the body of the SIP message	==> <==	UPDATE with the attribute line "a=sendrecv", or omitted attribute line, or "a= recvonly" for the offered media stream (see above) and encapsulated ISUP CPG message
Mapping: <== : Mapping from SIP to ISUP ==> : Mapping from ISUP to SIP			

A.11 Interworking of Completion of Calls to Busy Subscriber (CCBS) Supplementary Service to SIP networks

No procedures defined in this document.

A.12 Interworking of Completion of Calls on No Reply (CCNR) Supplementary Service to SIP networks

No procedures defined in this document.

A.13 Interworking of Terminal Portability (TP) Supplementary Service to SIP Networks

No procedures defined in this document.

A.14 Interworking of ISDN Conference Calling (CONF) Supplementary Service to SIP Networks

- *Operation without encapsulated ISUP:* ATIS-1000647.1995(R2010) and ATIS-1000647.a.1998(R2010) specifies the use of the ISUP Notification Indicator parameter with a number of values that are not mapped to SIP in this standard. Therefore, the service shall be terminated at the IWU and the call shall continue according to the basic call procedures.
- *Operation with encapsulated ISUP:* No additional interworking beyond use of ISUP encapsulation.

A.15 Interworking of Three-Party Service (3PTY) Supplementary Service to SIP Networks

No procedures defined in this document.

A.16 Interworking of Non-ISDN Multi-location Business Group (MBG) Supplementary Service to SIP Networks

- *Operation without encapsulated ISUP:* A number of the SS7 parameters carrying MBG-specific information are not mapped to SIP in this standard. According to ATIS-1000611.1991(R2008), if the MBG-specific SS7 information (e.g., the Business Group parameter in the Initial Address Parameter) is discarded during call setup, the MBG service the MBG service shall be terminated and the call shall continue according to the basic call procedures.
- *Operation with encapsulated ISUP:* No additional interworking beyond use of ISUP encapsulation.

A.17 Interworking of ISDN Multi-Level Precedence & Preemption (MLPP) Supplementary Service to SIP Networks

- *Operation without encapsulated ISUP:* The procedures described within ATIS-1000619.1992(R2010) and ATIS-1000619.a.1994(R2012) require SCCP connectivity between originating and terminating ISDN networks. This connectivity could be available as a bypass to the SIP network.
- *Operation with encapsulated ISUP:* No additional interworking beyond use of ISUP encapsulation.

A.18 Interworking of Global Virtual Network Service (GVNS) Supplementary Service to SIP Networks

No procedures defined in this document.

A.19 Interworking of International Telecommunication Charge Card (ITCC) Supplementary Service to SIP Networks

No procedures defined in this document.

A.20 Interworking of Reverse Charging (REV) Supplementary Service to SIP Networks

No procedures defined in this document.

A.21 Interworking of ISDN User-to-User Signalling (UUS) Supplementary Service to SIP Networks

- *Operation without encapsulated ISUP:* In accordance with the procedures described within ATIS-1000621.1992(R2009), the service shall be terminated at the IWU and the call shall continue according to the basic call procedures.
- *Operation with encapsulated ISUP:* All parameters can be taken from the encapsulated ISUP MIME.

The impact with regard to the full functionality of the UUS is for further study.

A.22 Interworking of Emergency Calling Service to SIP Networks

- *Operation without encapsulated ISUP:* Interworking of Emergency Calling Service as specified in ATIS-1000628.2000(R2010) and ATIS-1000628.a.2001(R2010) with SIP networks is for further study.
- *Operation with encapsulated ISUP:* No additional interworking beyond use of ISUP encapsulation.

A.23 Interworking of ISDN Normal Call Transfer Supplementary Service to SIP networks

- *Operation without encapsulated ISUP:* ATIS-1000632.1993(R2009) specifies the use of the ISUP Notification Indicator parameter with values 'Call transferred-alerting' and 'Call transferred-active' and these values are not mapped to SIP in this standard. Therefore, the service shall be terminated at the IWU and the call shall continue according to the basic call procedures.
- *Operation with encapsulated ISUP:* No additional interworking beyond use of ISUP encapsulation.

A.24 Interworking of ISDN Calling Name Identification Restriction Supplementary Service to SIP Networks

- *Operation without encapsulated ISUP:* ATIS-1000639.1995(R2011) and ATIS-1000639.a.2001(R2011) specify the use of the ISUP Generic Name parameter, which is not mapped to SIP in this standard. Therefore, the service shall be terminated at the IWU and the call shall continue according to the basic call procedures.
- *Operation with encapsulated ISUP:* No additional interworking beyond use of ISUP encapsulation and SCCP connectivity between originating and terminating ISDN networks. This connectivity could be available as a bypass to the SIP network.

A.25 Interworking of ISDN Calling Name Identification Presentation Supplementary Service to SIP Networks

- *Operation without encapsulated ISUP:* ATIS-1000641.1995(R2009) and ATIS-1000641.a.2002(R2012) specify the use of the ISUP Generic Name parameter, which is not mapped to SIP in this standard. Therefore, the service shall be terminated at the IWU and the call shall continue according to the basic call procedures.
- *Operation with encapsulated ISUP:* No additional interworking beyond use of ISUP encapsulation and SCCP connectivity between originating and terminating ISDN networks. This connectivity could be available as a bypass to the SIP network.

A.26 Interworking of ISDN Message Waiting Indicator Control & Notification Supplementary Service to SIP Networks

ISDN Message Waiting Indicator Control & Notification does not invoke ISUP/SIP call setup. Therefore, there is no interaction of the service at the IWU. Interworking of the SS7 TCAP that is used by this service requires SCCP connectivity between originating and terminating ISDN networks that is outside the scope of this standard. However, note that this connectivity could be available as a bypass to the SIP network.

A.27 Interworking of ISDN Call Park Supplementary Service to SIP Networks

In accordance with the procedures described within ATIS-1000653.1996(R2010) and ATIS-1000653.a.1998(R2010), there is no interaction of the service at the IWU.

A.28 Interworking of Non-ISDN Assist Supplementary Service to SIP Networks

- *Operation without encapsulated ISUP:* The procedures described within ATIS-1000611.1991(R2008) require SCCP connectivity between originating and terminating ISDN networks. This connectivity could be available as a bypass to the SIP network.
- *Operation with encapsulated ISUP:* No additional interworking is required beyond the requirement for SCCP connectivity between originating and terminating ISDN networks. This connectivity could be available as a bypass to the SIP network.

A.29 Interworking of Non-ISDN Carrier Selection Supplementary Service to SIP Networks

- *Operation without encapsulated ISUP:* Interworking of this service as specified in ATIS-1000611.1991(R2008) with SIP networks is for further study.
- *Operation with encapsulated ISUP:* No additional interworking beyond use of ISUP encapsulation.

A.30 Interworking of Non-ISDN Directory Assistance with Call Completion Supplementary Service to SIP Networks

No procedures defined in this document.

A.31 Interworking of Non-ISDN User Network Interaction Supplementary Service to SIP Networks

Interworking of the non-ISDN user network interaction supplementary service described within ATIS-1000611.1991(R2008) to SIP networks is for further study.

A.32 Interworking of Non-ISDN Voice Message Waiting Indication Control Supplementary Service to SIP Networks

- *Operation without encapsulated ISUP:* The procedures described within ATIS-1000611.1991(R2008) require SCCP connectivity between originating and terminating ISDN networks. This connectivity could be available as a bypass to the SIP network.
- *Operation with encapsulated ISUP:* No additional interworking beyond use of ISUP encapsulation.

A.33 Interworking of Non-ISDN 950+Call Supplementary Service to SIP Networks

- *Operation without encapsulated ISUP:* Interworking of this service as specified in ATIS-1000611.1991(R2008) with SIP networks is for further study.
- *Operation with encapsulated ISUP:* No additional interworking beyond use of ISUP encapsulation.

A.34 Interworking of Interworking with Non-ISDN Private Network Supplementary Service to SIP Networks

- *Operation without encapsulated ISUP:* ATIS-1000611.1991(R2008) specifies the use of the ISUP Special Processing Request parameter and of the Generic Name parameter values that are not mapped to SIP in this standard. Therefore, the service shall be terminated at the IWU and the call shall continue according to the basic call procedures.
- *Operation with encapsulated ISUP:* No additional interworking beyond use of ISUP encapsulation.

Annex B: SIP/SIP-I References

(normative)

Annex B contains references to normative Internet Engineering Task Force (IETF) RFCs and materials originally sourced from the IETF but deemed normative to this Recommendation.

B.1 SIP/SIP-I Signaling References and Profile

B.1.1 References

IETF RFC 2046 (1996), *Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types*.[31]

IETF RFC 2327 (1998), *SDP: Session Description Protocol*.[31]

IETF RFC 2806 (2000), *URLs for Telephone Calls*. [31]

IETF RFC 2976 (2000), *The SIP INFO Method*. [31]

IETF RFC 3204 (2001), *MIME media types for ISUP and QSIG Objects*. [31]

IETF RFC 3261 (2002), *SIP: Session Initiation Protocol*. [31]

IETF RFC 3262 (2002), *Reliability of Provisional Responses in the Session Initiation Protocol (SIP)*. [31]

IETF RFC 3264 (2002), *An Offer/Answer Model with the Session Description Protocol (SDP)*. [31]

IETF RFC 3311 (2002), *The Session Initiation Protocol UPDATE Method*. [31]

IETF RFC 3312 (2002), *Integration of Resource Management and Session Initiation Protocol (SIP)*. [31]

IETF RFC 3325 (2002), *Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks*. [31]

IETF RFC 3326 (2002), *The Reason Header Field for the Session Initiation Protocol (SIP)*. [31]

B.2 SIP/SIP-I Media References

B.2.1 References

IETF RFC 3550 (2003), *RTP: A Transport Protocol for Real-Time Applications*. [31]

IETF RFC 3551 (2003), *RTP Profile for Audio and Video Conferences with Minimal Control*. [31]

IETF RFC 2833 (2000), *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*. [31]

IETF RFC 3267 (2002), *Real-time Transport Protocol RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs*. [31]

IETF RFC 3389 (2002), *RTP Payload for Comfort Noise*. [31]

ITU-T Recommendation T.38 (02/00), *Procedures for real-time Group 3 facsimile communication over IP networks*.[27]

[31] This document is available from the Internet Engineering Task Force (IETF). < http://www.ietf.org >

Annex C: Interworking scenarios between SIP and ISUP

(informative)

C.1 Scope

This annex defines typical interworking scenarios between ISUP and SIP. ISDN Access flows are included for informational purposes only.

C.2 Definitions

The vertical boxes represent two entities: 1. ISUP; and 2. IWU (SIP-ISUP Interworking Unit).

The vertical dashed lines represent the access interface. Each access interface supports a single access type: *ISDN* or *SIP-NNI*.

Solid horizontal arrows represent signaling messages and indicate their direction of propagation – i.e., to or from the interworking unit. The interaction of messages shown along the vertical represent increasing time in the downward direction. All events on the same vertical line are related – e.g., an incoming message causes voice-path connections and triggers an outgoing message. Events on different vertical lines are not related unless connected by dashed lines; a dashed line indicates that an incoming message may trigger an event at a later time.

Wavy horizontal arrows (~~>) represent tones or announcements sent in-band.

Timers are represented as vertical arrows.

For call control, the following symbols are used within the vertical boxes to indicate the relationship between the incoming and outgoing messages and the call control actions taken.

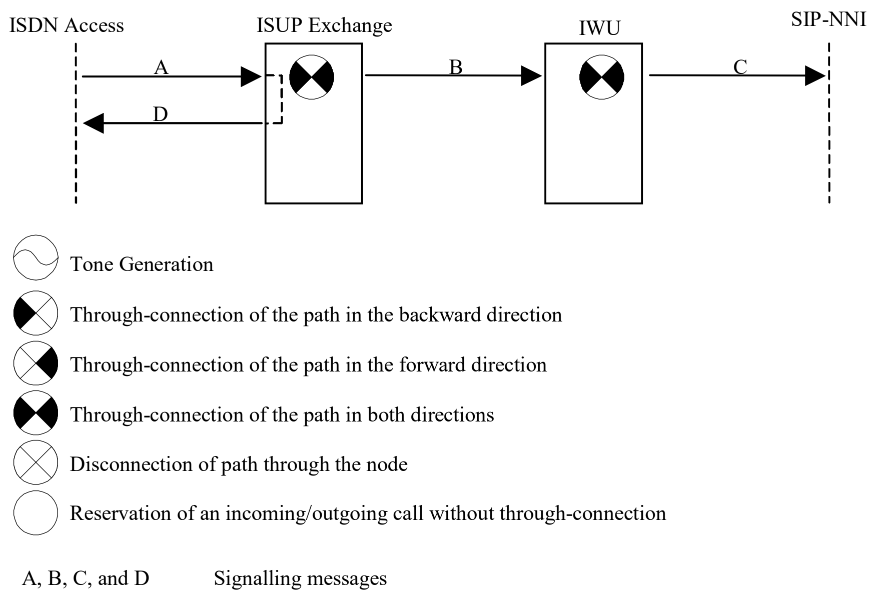


Figure C.1 - Example of a call flow or "arrow" diagram

C.3 Abbreviations

See clause 3 of the main document.

C.4 Methodology

Call flow or “arrow” diagrams are provided to show the temporal relationships between signaling messages during execution of a call control procedure. The general format of an arrow diagram is shown in Figure A.1. The main part of the Recommendation takes precedence over this annex.

C.5 Interworking of SIP Access to ISUP

Clauses C.5.1 and C.5.2 contain information relevant to basic call control. The call flow diagrams are divided into functional subclauses:

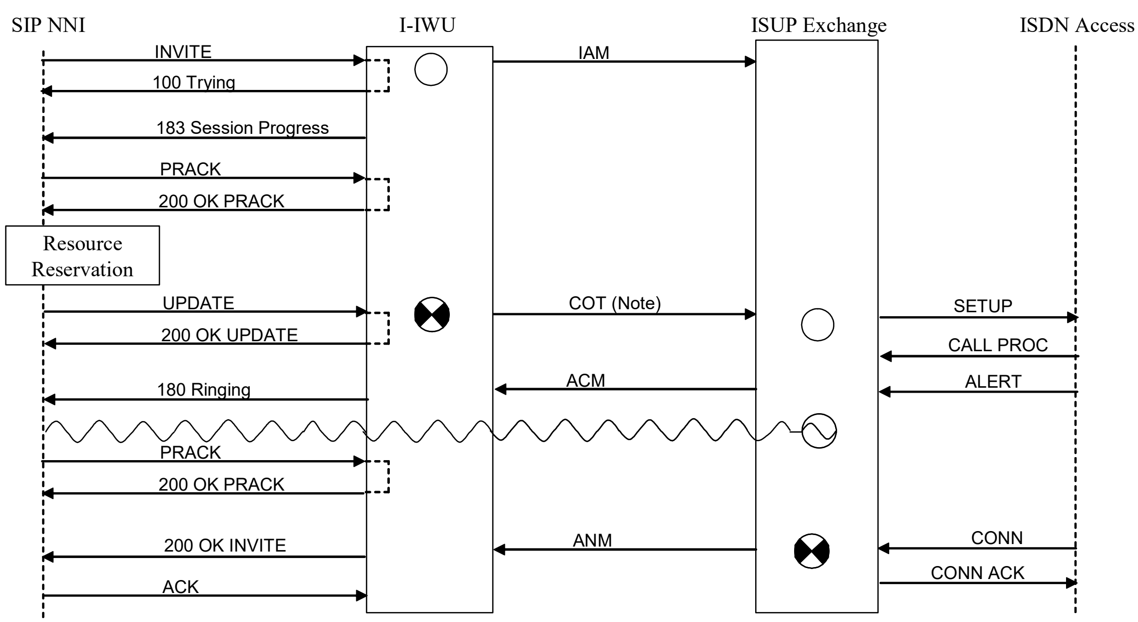
- Successful call set-up procedures;
- Unsuccessful call set-up procedures; and
- Release procedures.

C.5.1 Example Scenarios for Incoming Call Interworking from SIP to ISUP at I-IWU

C.5.1.1 Successful Call Set-Up Procedures/Call Flow Diagrams for Basic Call Control

C.5.1.1.1 SIP Preconditions Used

Figure C.2 shows the sequence of messages for successful call set-up for an outgoing call from SIP to ISUP. In this sequence, the SIP side indicates mandatory local resource reservation (such as sendrecv) in the INVITE. The IAM (with ‘continuity check performed on previous circuit’ or ‘continuity check required on this circuit’ indication) is sent by the IWU once the initial INVITE is received, and a COT message (with ‘continuity check successful’ indication) is sent once the SIP side has reserved resources for the call (confirmed in the UPDATE).

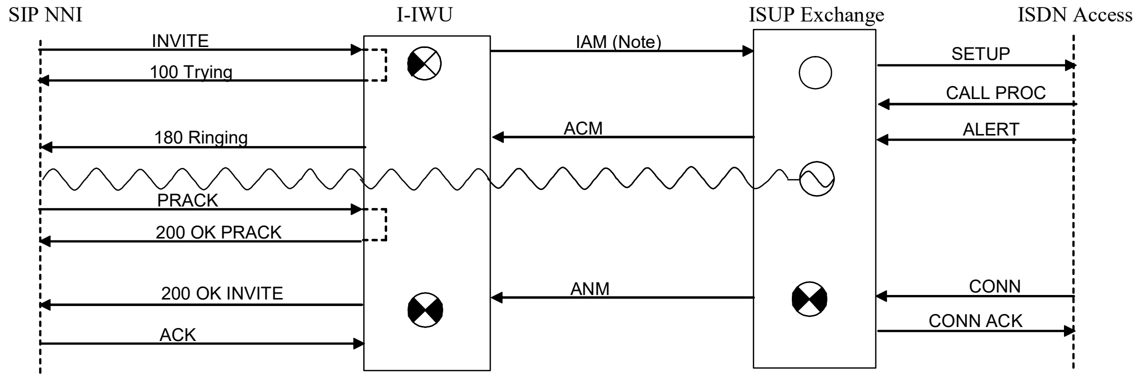


NOTE – The IAM contained the indication”continuity check performed on previous circuit” or”continuity check required on this circuit”.

Figure C.2 - Successful Basic Call Set-Up from SIP to ISUP (SIP Preconditions and Continuity Check Protocol Used)

C.5.1.1.2 SIP Preconditions Not Used

Figure C.3 shows the sequence of messages for successful call set-up for an outgoing call from SIP to ISUP. The IAM (with 'continuity check not required' indication) is sent by the IWU once the initial INVITE is received.

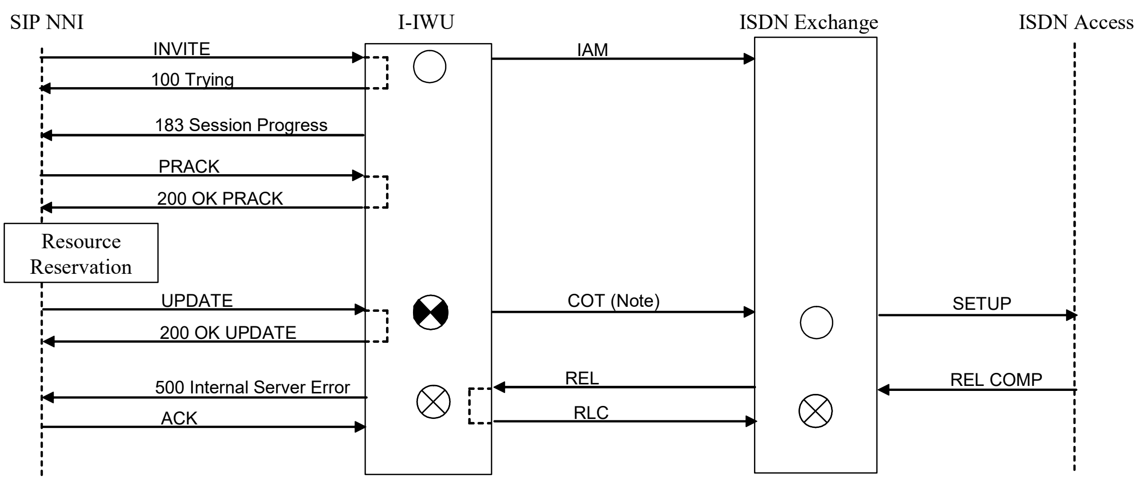


NOTE – The IAM contained the indication *"continuity check not required"*.

Figure C.3 - Successful Basic Call Set-Up from SIP to ISUP (SIP Preconditions and Continuity Check Protocol Not Used)

C.5.1.2 Unsuccessful Call Set-Up Procedures/Call Flow Diagrams for Basic Call Control

Figure C.4 shows the sequence of messages for unsuccessful call set-up for an outgoing call from SIP to ISUP. In this sequence, the IWU sends the 500 Server Internal Error message upon reception of the REL message – with Cause Value No. 34 (resource unavailable) – from the ISUP side of the call.



NOTE – This message is optional, depending on the indication in the IAM

Figure C.4 - Unsuccessful Basic Call Set-Up from SIP to ISUP

C.5.1.3 Normal Call Release Procedure

Figure C.5 shows a normal call release procedure initiated from the SIP side of the call. This call flow assumes that no resource reservation teardown signaling is required on the SIP side.

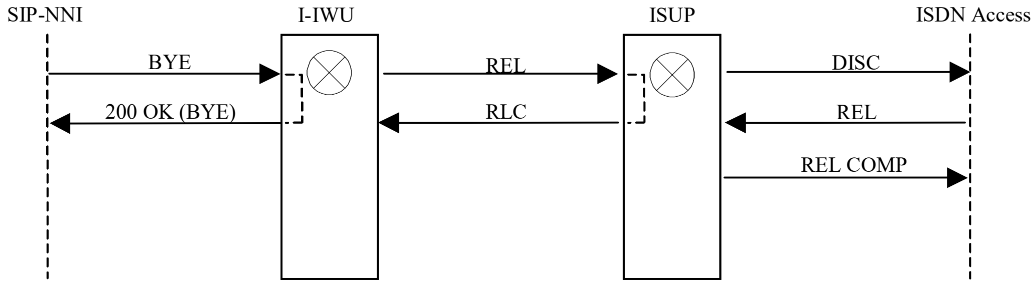


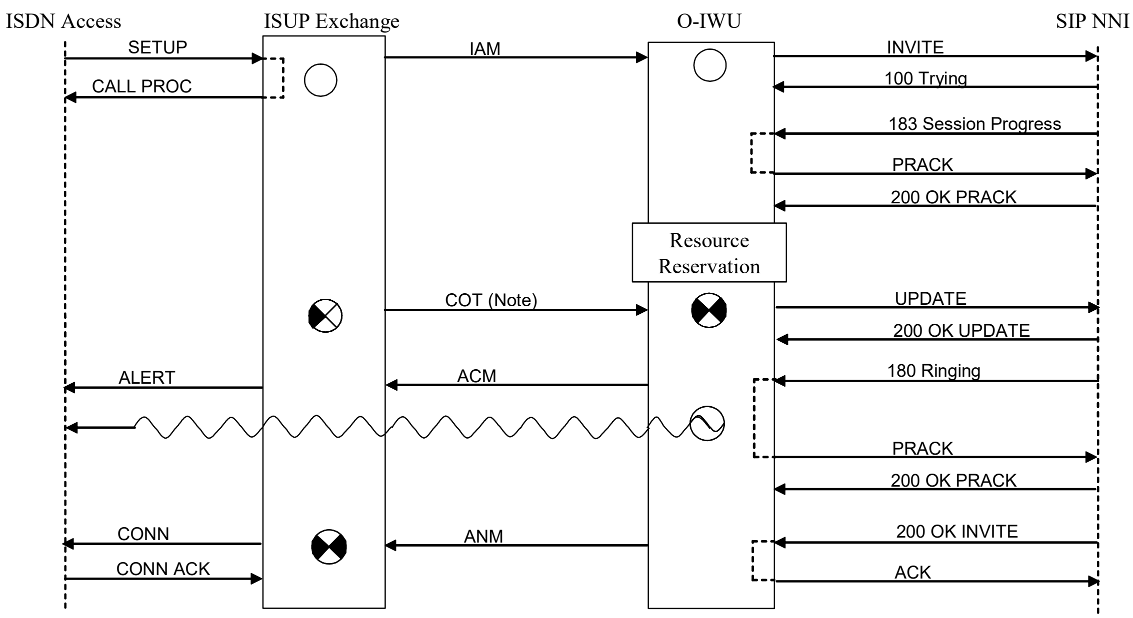
Figure C.5 - Normal Call Release from SIP to ISUP

C.5.2 Example scenarios for Outgoing Call Interworking from ISUP to SIP at O-IWU

C.5.2.1 Successful Call Set-Up Procedures/Call Flow Diagrams for Basic Call Control

C.5.2.1.1 SIP Preconditions Used

Figure C.6 shows a sequence of messages for successful call set-up for an incoming call from ISUP to SIP. In this example, the IWU indicates mandatory local sendrecv preconditions in the INVITE. The IWU then sends the UPDATE message upon reception of a COT message (if the IAM indicated ‘continuity check performed on previous circuit’ or ‘continuity check required on this circuit’) and completion of any local resource reservation. The UPDATE message will confirm that the local preconditions have been met.

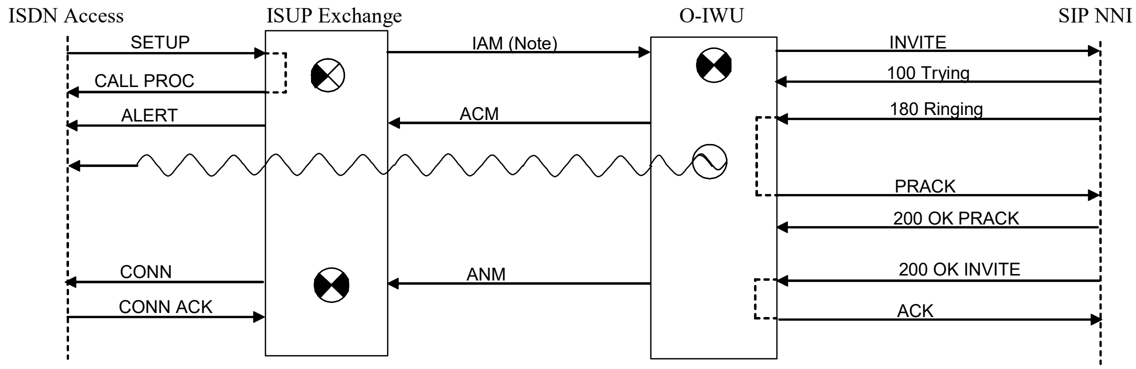


NOTE – This message is optional, depending on the indication in the IAM.

Figure C.6 - Successful Basic Call Set-Up from ISUP to SIP (SIP Preconditions and Continuity Check Protocol Used)

C.5.2.1.2 SIP Preconditions Not Used

Figure C.7 shows a sequence of messages for successful call set-up for an incoming call from ISUP to SIP. In this example, the IWU sends the INVITE message upon reception of an IAM (since the IAM indicated ‘*continuity check not required*’).



NOTE – The IAM contained the indication *"continuity check not required"*.

Figure C.7 - Successful Basic Call Set-Up from ISUP to SIP (SIP Preconditions and Continuity Check Protocol Not Used)

C.2.2.2 Unsuccessful Call Set-Up Procedures/Call Flow Diagrams for Basic Call Control

Figure C.8 shows a sequence of messages for unsuccessful call set-up for an incoming call from ISUP to SIP. In this example, the IWU sends the REL message upon reception of the 484 Address Incomplete message from the SIP side of the call.

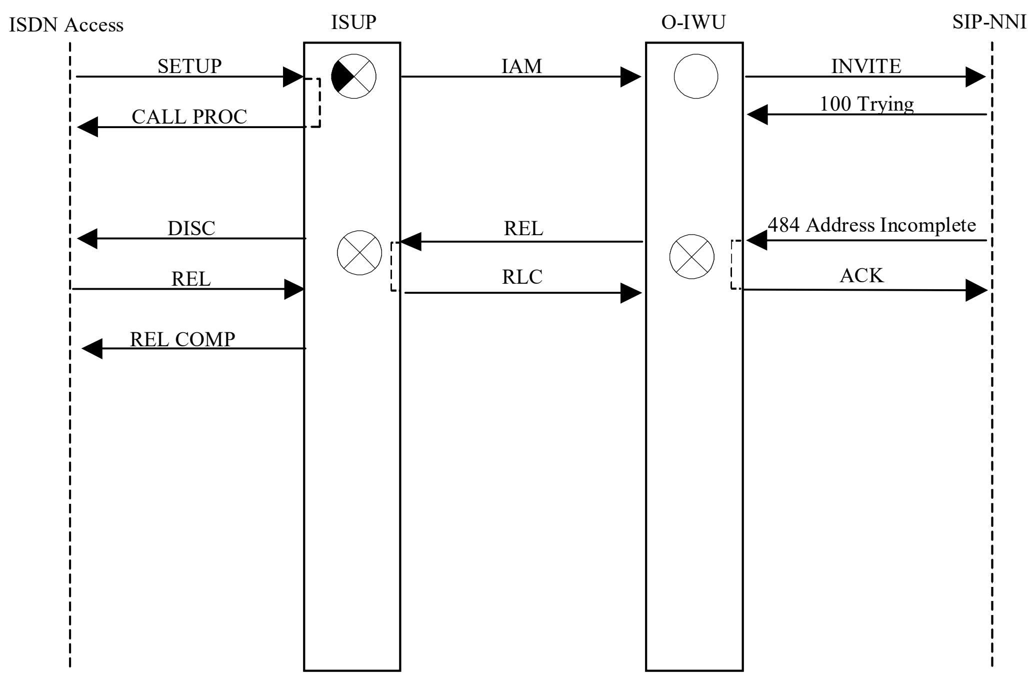


Figure C.8 - Unsuccessful Basic Call Set-Up from ISUP to SIP

C.5.2.3 Normal Call Release Procedure

Figure C.9 shows a normal call release procedure initiated from the ISUP side of the call. This call flow assumes that no resource reservation teardown signaling is required on the SIP side of the call.

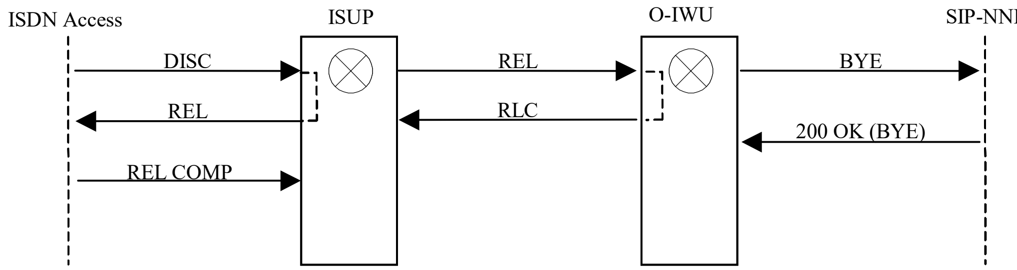


Figure C.9 - Normal Call Release from ISUP to SIP

Annex D: Interworking scenarios between SIP-I and ISUP

(informative)

D.1 Scope

This annex defines typical interworking scenarios between ISUP and SIP-I. ISDN Access flows are included for informational purposes only.

D.2 Definitions

The vertical boxes represent two entities: 1. ISUP; and 2. IWU (SIP-ISUP Interworking Unit).

The vertical dashed lines represent the access interface. Each access interface supports a single access type: *ISDN* or *SIP-NNI*.

Solid horizontal arrows represent signaling messages and indicate their direction of propagation -- i.e., to or from the interworking function. The interaction of messages shown along the vertical represents increasing time in the downward direction. All events on the same vertical line are related -- e.g., an incoming message causes voice-path connections and triggers an outgoing message. Events on different vertical lines are not related unless connected by dashed lines; a dashed line indicates that an incoming message may trigger an event at a later time.

Wavy horizontal arrows (~~>) represent tones or announcements sent in-band.

Timers are represented as vertical arrows.

For call control, the following symbols are used within the vertical boxes to indicate the relationship between the incoming and outgoing messages and the call control action taken.

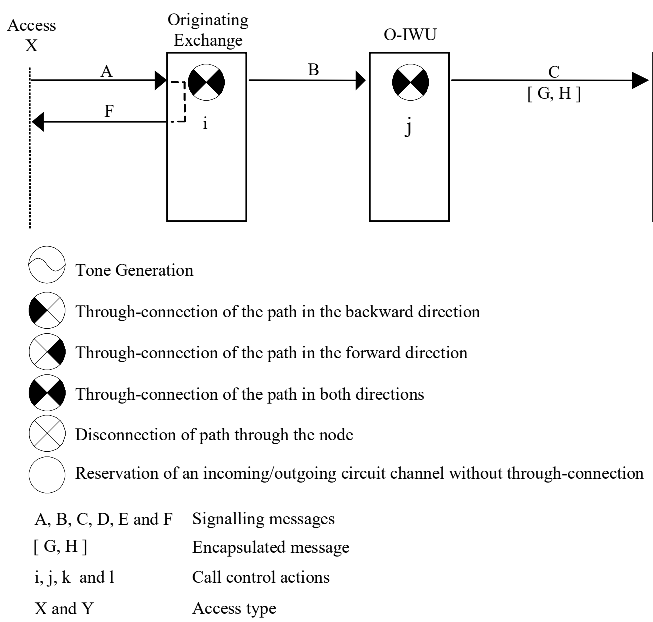


Figure D.1 - Example of a call flow or "arrow" diagram

D.3 Interworking of ISUP with SIP-I

Clauses D.3.1 to D.3.4 contain information relevant to basic call control. The call flow diagrams are divided into functional subclauses:

- Successful call set-up procedures;
- Unsuccessful call set-up procedures;
- Release procedures; and
- Suspend/resume procedures.

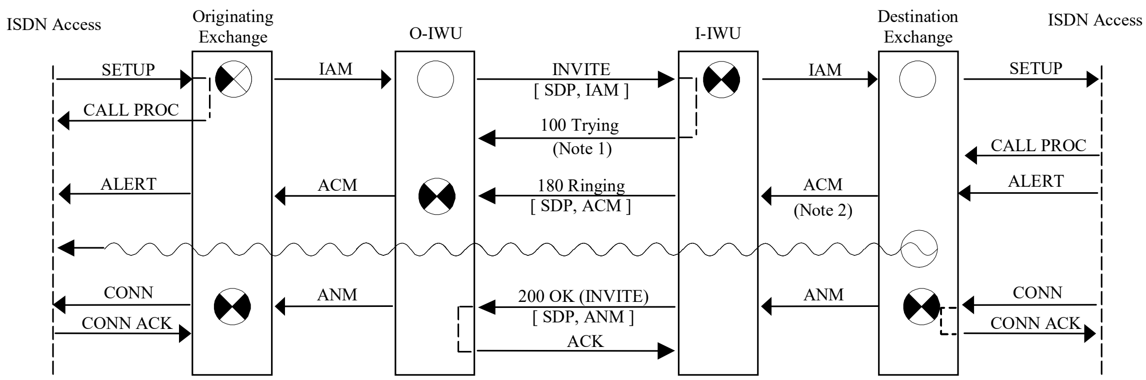
D.3.1 Successful Call Set-up Procedures/Call Flow Diagrams for Basic Call Control

D.3.1.1 En bloc, subscriber free indication

See ATIS-1000113.2005, chapter 4, clause 2.1, and RFC 3261, clause 13.

NOTE – Termed Late ACM.

Figure D.2 shows the sequence of messages for successful call set-up for an incoming ISUP call over SIP-I. The O-IWU performs the through-connection of the bearer path in both directions after the receipt of SDP answer in the 180 response.



NOTE 1 – The generation of the 100 (Trying) response is necessary if the I-IWU knows that it will not generate a provisional or final response within 200 ms.

NOTE 2 – The ACM is generated with following parameters: called party satus = subscriber free; ISDN Access Indicator = ISDN access

Figure D.2 - En bloc, subscriber free indication

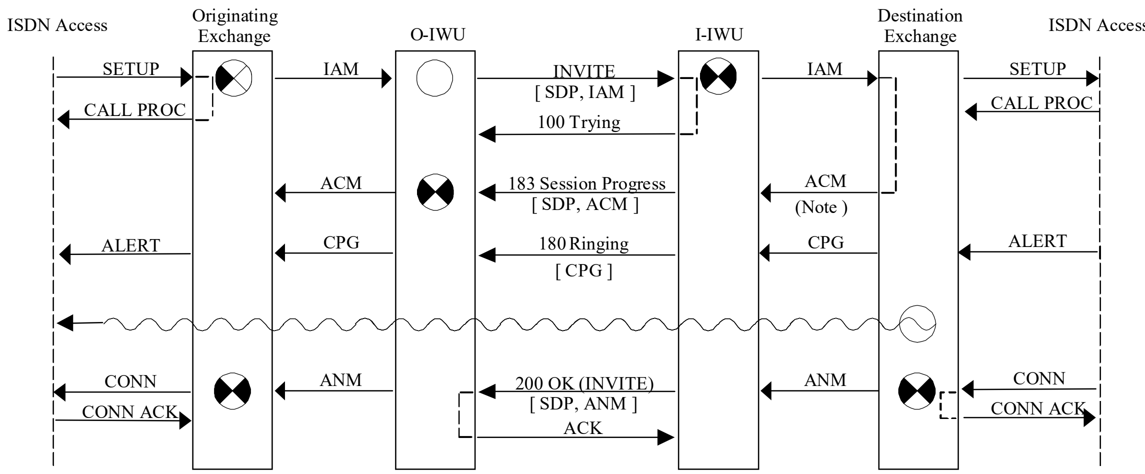
For detailed messages and parameter mapping, refer to:

- *IAM message:* clauses 6.1.3 and 7.1
- *ACM message:* clauses 6.3 and 7.2
- *ANM message:* clauses 6.5 and 7.6

D.3.1.2 En bloc, early ACM

See ATIS-1000113.2005, chapter 4, clause 2.1, and RFC 3261, clause 13.

Figure D.3 shows the sequence of messages for successful call set-up for an incoming ISUP call over SIP-I. At the I-ISN the ACM message is mapped and encapsulated to 183 Session Progress provisional response preserving the ISUP signaling transparency. The O-IWU performs the through-connection of the bearer path in both directions after the receipt of SDP answer in 183 Session Progress response.



NOTE – The method of ACM generating independent of access is termed *Early* ACM. The ACM is independently generated at the destination exchange with the following parameters: called party satus = no indication; ISDN Access Indicator = ISDN access.

Figure D.3 - En bloc, early ACM encapsulation

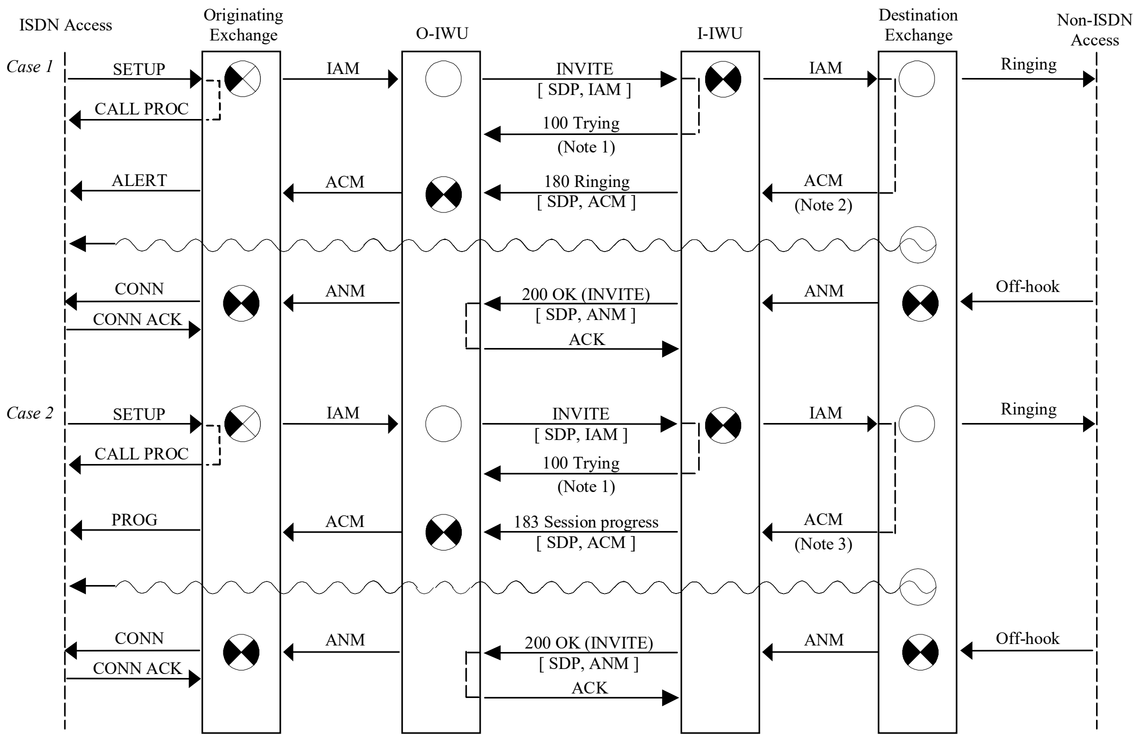
For detailed messages and parameter mapping, refer to:

- *IAM message:* clauses 6.1.3 and 7.1
- *ACM message:* clauses 6.3 and 7.2
- *CPG message:* clauses 6.4 and 7.2
- *ANM message:* clauses 6.5 and 7.6

D.3.1.3 En bloc, early media scenarios

See ATIS-1000113.2005, chapter 4, clause 2.1, and RFC 3261, clause 13.

Figure D.4 shows the sequence of messages for a call from an ISDN access to a non-ISDN access.



NOTE 1 – The generation of the 100 (Trying) response is necessary if the I-IWU knows that it will not generate a provisional or final response within 200 ms.

NOTE 2 – The ACM is independently generated at the destination exchange with the following parameters: called party satus = subscriber free; ISDN Access Indicator = non-ISDN access

NOTE 3 – The ACM is independently generated at the destination exchange with the following parameters: called party satus = no indication; ISDN Access Indicator = non-ISDN access. In order to support user-generated in-band information (e.g from PBX. See 2.1.4.1b/Q.764), the destination exchange may include in the ACM optional backward call indicators = in-band information available and through-connect in the backward direction.

Figure D.4 - Early media call-flows

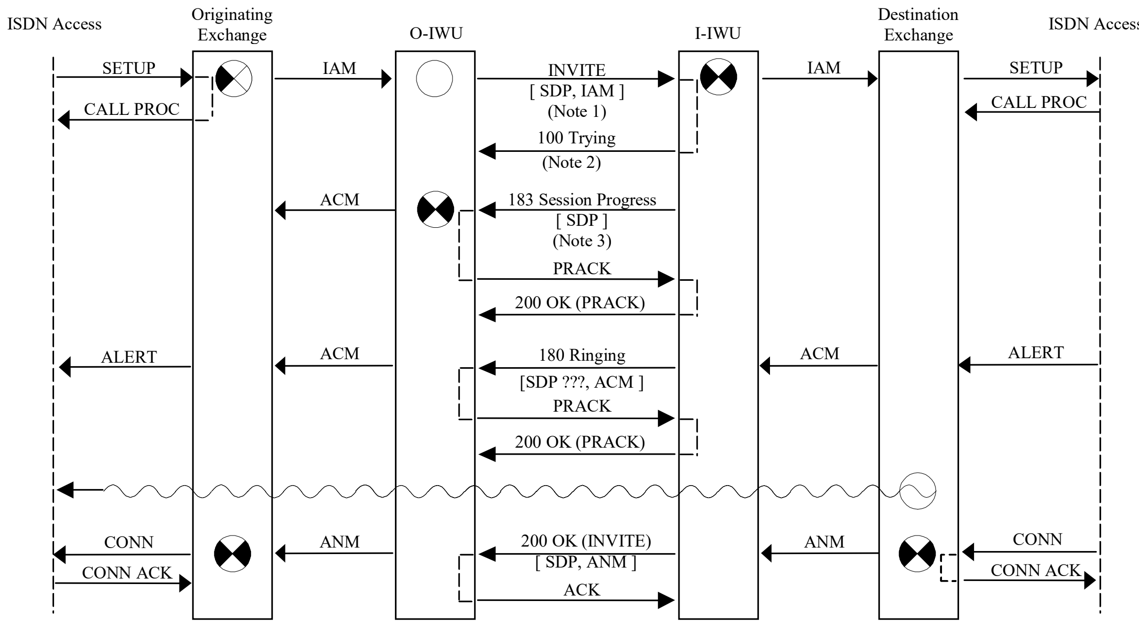
For detailed messages and parameter mapping, refer to:

- *IAM message:* clauses 6.1.3 and 7.1
- *ACM message:* clauses 6.3 and 7.2
- *CPG message:* clauses 6.4 and 7.2
- *ANM message:* clauses 6.5 and 7.6

D.3.1.4 En Bloc, Early Session Description Negotiation

See ATIS-1000113.2005, chapter 4, clause 2.1, and RFC 3261, clause 13.

Figure D.5 shows the sequence of messages for successful call set-up for an incoming ISUP call over SIP-I. The O-IWU indicates the support of reliable provisional responses in INVITE request. The I-IWU uses 183 provisional response to facilitate earlier session description negotiation. After the receipt of SDP answer in 183 response the O-IWU performs the through-connection of the bearer path in both directions.



NOTE 1 – INVITE contains the Supported header field with the option tag 100rel

NOTE 2 – The generation of the 100 (Trying) response is necessary if the I-IWU knows that it will not generate a provisional or final response within 200 ms.

NOTE 3 – I-IWU sends 183 provisional response to facilitate session description negotiation before the setup of SIP dialog

Figure D.5 - En bloc, early session description negotiation

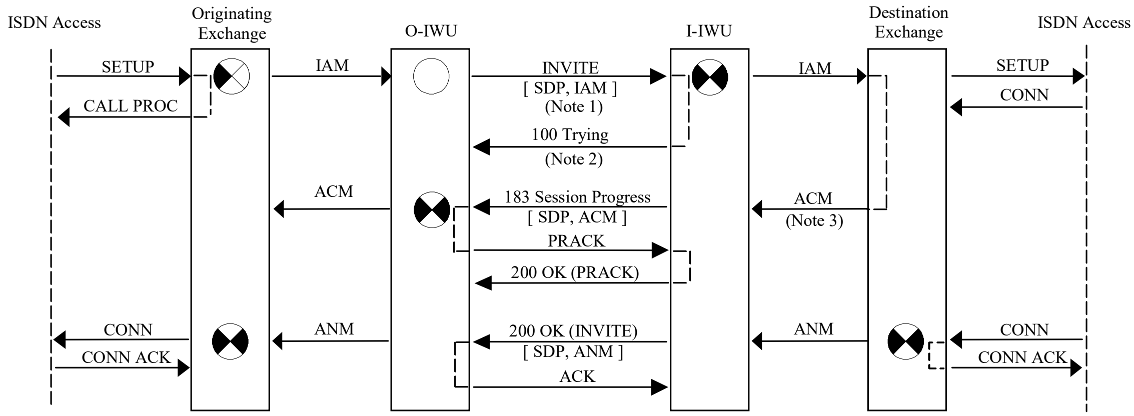
For detailed messages and parameter mapping, refer to:

- *IAM message:* clauses 6.1.3 and 7.1
- *ACM message:* clauses 6.3 and 7.2
- *ANM message:* clauses 6.5 and 7.6

D.3.1.5 En bloc, reliable provisional responses

See ATIS-1000113.2005, chapter 4, clause 2.1, and RFC3262, clause 4.

Figure D.6 shows the sequence of messages for successful call set-up for an incoming ISUP call over SIP-I. The O-IWU indicates the required support of reliable provisional responses by adding option tag 100rel to the Required header field of the INVITE request. At the I-ISN the ACM message is mapped and encapsulated to 183 response preserving the ISUP signaling transparency. The O-IWU confirms the receipt of provisional response with the PRACK request. The 200 OK INVITE contains no SDP, since the offer-answer exchange is completed during the preceding steps. This is only possible where the provisional responses are transmitted reliably.



NOTE 1 – INVITE contains the Required header field with the option tag 100rel

NOTE 2 – The generation of the 100 (Trying) response is necessary if the I-IWU knows that it will not generate a provisional or final response within 200 ms.

NOTE 3 – The ACM is generated with following parameters: called party satus = no indication; ISDN Access Indicator =ISDN access.

Figure D.6 - En bloc, use of reliable provisional responses

For detailed messages and parameter mapping, refer to:

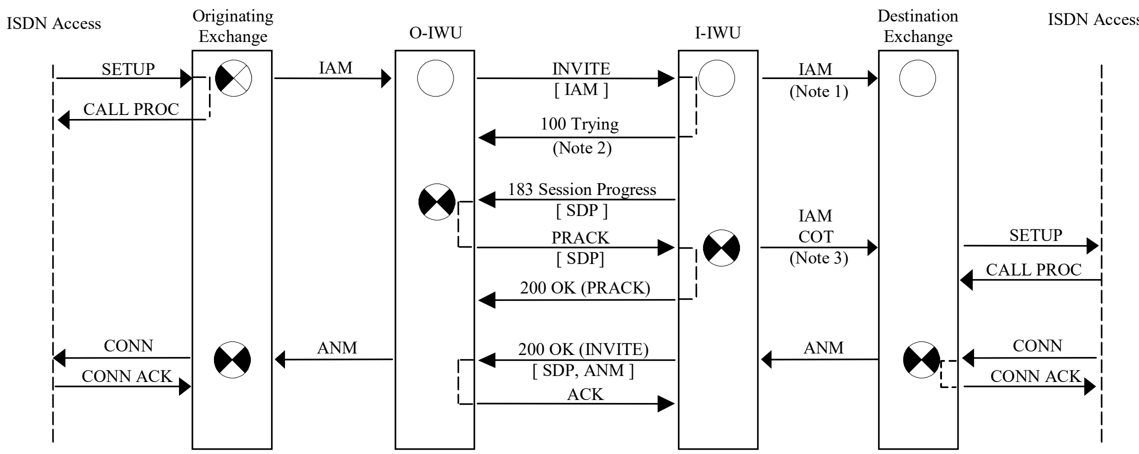
- *IAM message:* clauses 6.1.3 and 7.1
- *ACM message:* clauses 6.3 and 7.2
- *ANM message:* clauses 6.5 and 7.6

D.3.1.6 En Bloc, Backward SDP Offer

See ATIS-1000113.2005, chapter 4, clause 2.1, and RFC 3261, clause 13.

Figure D.7 shows the sequence of messages for successful call set-up for an incoming ISUP call over SIP-I. Depending on configuration, the O-IWU can omit the SDP in the initial INVITE thus asking I-IWU to provide the SDP offer. The indication of reliable provisional responses support is included. The I-IWU, if configured, can transfer SDP offer via 183 response. The O-IWU responds with SDP answer and performs the through-connection of the bearer bath in both directions after the receipt of SDP answer in 183 response.

Depending on configuration, I-IWU can directly send IAM with 'COT on Previous Circuit' indication and continue the call setup by sending COT after receipt of SDP answer. As an alternative it can delay the sending of IAM until the receipt of SDP answer. See 6.1.1(1). In any scenario, the I-IWU through-connects the bearer path on the receipt of SDP answer.



NOTE 1 – In case of direct IAM sending it will contain 'COT on previos Circuit' indication

NOTE 2 – The generation of the 100 (Trying) response is necessary if the I-IWU knows that it will not generate a provisional or final response within 200 ms

NOTE 3 – Sending of IAM or COT message on the receipt of SDP answer depends on the I-IWU configuration. See subsection - xyz

Figure D.7 - En bloc, backward session description initiation

For detailed messages and parameter mapping, refer to:

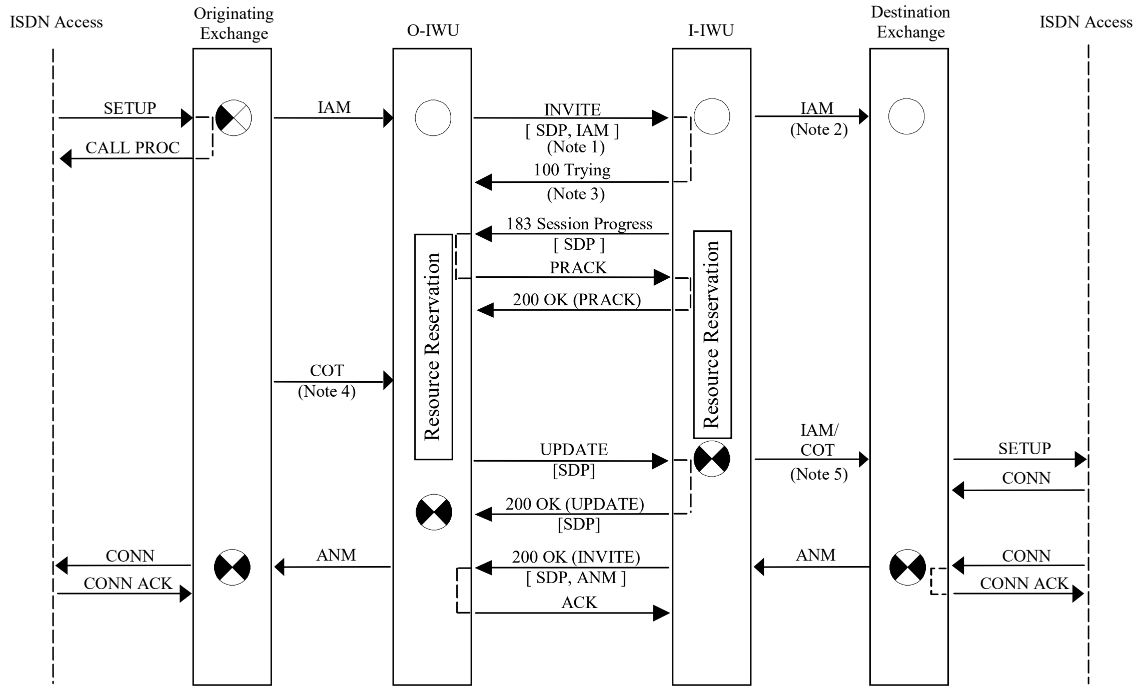
- *IAM message:* clauses 6.1.3 and 7.1
- *ANM message:* clauses 6.5 and 7.6

D.3.1.7 En Bloc, End-to-End Resource Reservation

See 2.1/ATIS-1000113.2005, chapter 4, clause 2.1 and 13.1/RFC3312.

Figure D.8 shows the sequence of messages for successful call set-up for an incoming ISUP call over SIP-I. The O-IWU indicates mandatory end-to-end sendrecv quality of service preconditions in the SDP of initial INVITE and also the required use of reliable provisional responses. The I-ISN requests confirmation from O-IWU of network resource reservation in the SDP of 183 response and begins with own network resource reservation. After successful network resource reservation and reception of a COT message (if the IAM from originating exchange indicated 'COT on Previous Circuit') the O-IWU indicated that in the SDP of UPDATE request. Having already reserved network resources I-IWU confirms the achieved end-to-end sendrecv precondition in the SDP of 200 OK (UPDATE).

Depending on configuration, I-IWU can directly send IAM with 'COT on Previous Circuit' indication and continue the call setup by sending COT after meeting the preconditions. As an alternative it can delay the sending of IAM until the meeting of preconditions. See 6.1.2(2).



NOTE 1 – INVITE contains the mandatory end-to-end sendrecv preconditions in SDP and Required header field with the option tag 100rel

NOTE 2 – In case of direct IAM sending it will contain 'COT on previos Circuit' indication

NOTE 3 – The generation of the 100 (Trying) response is necessary if the I-IWU knows that it will not generate a provisional or final response within 200 ms.

NOTE 4 – This message is optional, depending on the indication in the IAM

NOTE 5 – Sending of IAM or COT message after meeting the preconditions depends on the I-IWU configuration. See subsection - xyz

Figure D.8 - En bloc, end-to-end preconditions for resource reservation

For detailed messages and parameter mapping, refer to:

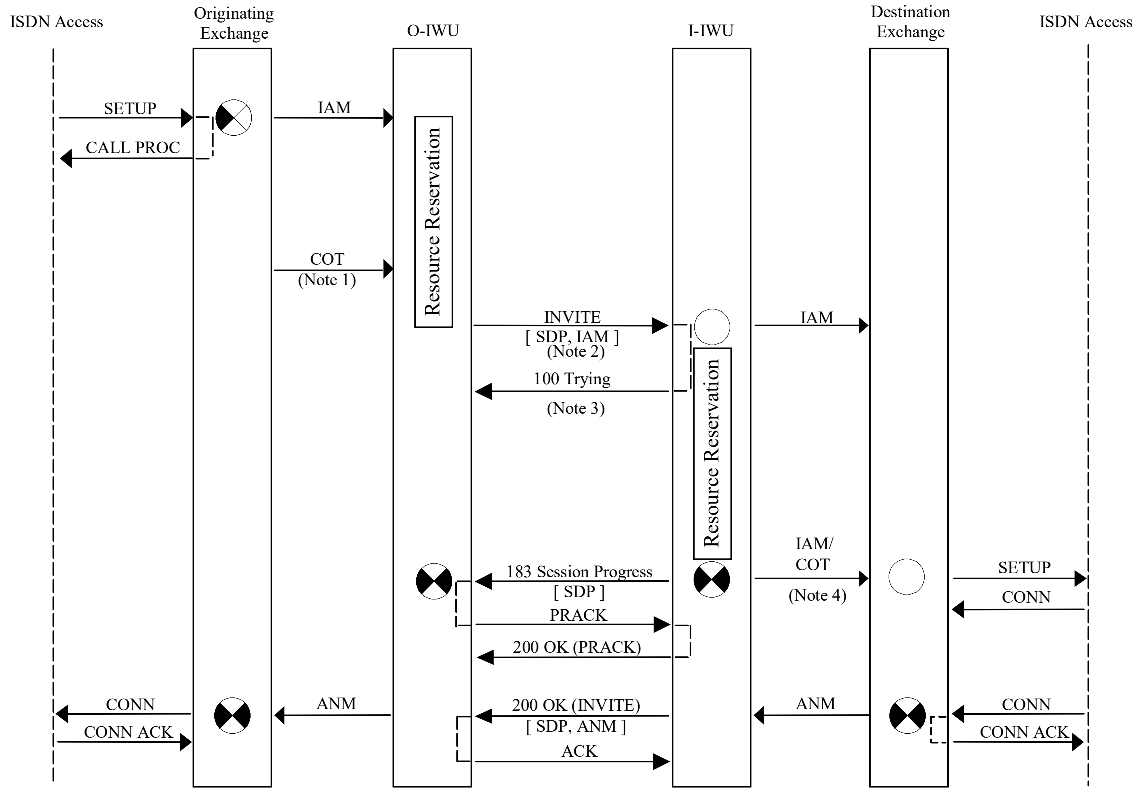
- *IAM message:* clauses 6.1.3 and 7.1
- *COT message:* clauses 6.2 and 7.1
- *ANM message:* clauses 6.5 and 7.6

D.3.1.8 En Bloc, Segmented Resource Reservation

See ATIS-1000113.2005, chapter 4, clause 2.1, and RFC3312, clause 13.2.

Figure D.9 shows the sequence of messages for successful call set-up for an incoming ISUP call over SIP-I. On the receipt of IAM, the O-IWU reserves resources in its local network branch. On successful reservation and reception of a COT message (if the IAM from originating exchange indicated 'COT on Previous Circuit'), O-IWU includes the request for remote local resource reservation in the SDP of initial INVITE and also the required use of reliable provisional responses. After local network resource reservation, the I-ISN notifies the O-IWU with SDP in 183 response that all preconditions are met.

Depending on configuration, I-IWU can directly send IAM with 'COT on Previous Circuit' indication and continue the call setup by sending COT after meeting the preconditions. As an alternative, it can delay the sending of IAM until the meeting of preconditions. See 6.1.2(2).



NOTE 1 – This message is optional, depending on the indication in the IAM

NOTE 2 – INVITE contains the mandatory segmented sendrecv preconditions in SDP and Required header field with the option tag 100rel

NOTE 3 – The generation of the 100 (Trying) response is necessary if the I-IWU knows that it will not generate a provisional or final response within 200 ms.

NOTE 4 – Sending of IAM or COT message after meeting the precondition depends on the I-IWU configuration. See subsection - xyz

Figure D.9 - En bloc, segmented preconditions for resource reservation

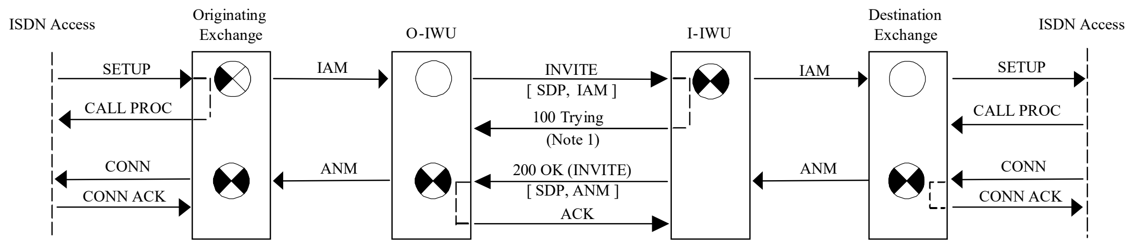
For detailed messages and parameter mapping, refer to:

- *IAM message:* clauses 6.1.3 and 7.1
- *COT message:* clauses 6.2 and 7.1
- *ANM message:* clauses 6.5 and 7.6

D.3.1.9 En Bloc, Automatic Call Answering

See ATIS-1000113.2005, chapter 4, clause 2.1, and RFC 3261, clause 13.

Figure D.10 shows the sequence of messages for successful call set-up for an incoming ISUP call over SIP-I. The I-ISN sends the 200 OK response on the receipt of CONNECT message containing the address complete and the connect indication. Both IWUs perform the through-connection of the bearer path in both directions on the receipt of connect indication.



NOTE 1 – The generation of the 100 (Trying) response is necessary if the I-IWU knows that it will not generate a provisional or final response within 200 ms.

Figure D.10 - En bloc, automatic answering terminal

For detailed messages and parameter mapping, refer to:

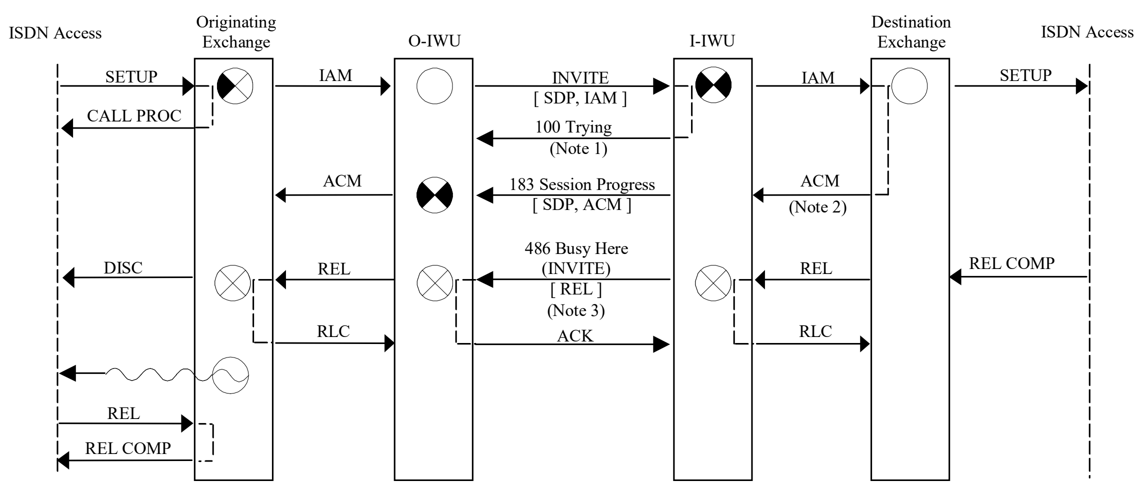
- *IAM message:* clauses 6.1.3 and 7.1
- *ANM message:* clauses 6.5 and 7.6

D.3.2 Unsuccessful Call Setup Procedures/Call Flow Diagrams for Basic Call Control

D.3.2.1 Backward Release During Call Setup

See ATIS-1000113.2005, chapter 4, clause 2.1, and RFC 3261, clause 15.

Figure D.11 shows the unsuccessful call set-up procedure where tones or announcements are generated in the originating exchange. The REL message is mapped and encapsulated into the appropriate SIP unsuccessful response status code depending on the cause value.



NOTE 1 – The generation of the 100 (Trying) response is necessary if the I-IWU knows that it will not generate a provisional or final response within 200 ms.

NOTE 2 – If *Early* ACM is used, the ACM is independently generated with the following parameters: called party satus = no indication; ISDN Access Indicator = ISDN access

NOTE 3 – See the table/section xyz for the mapping of ISUP release causes to the SIP error responses

Figure D.11 -Backward release during call setup

For detailed messages and parameter mapping, refer to:

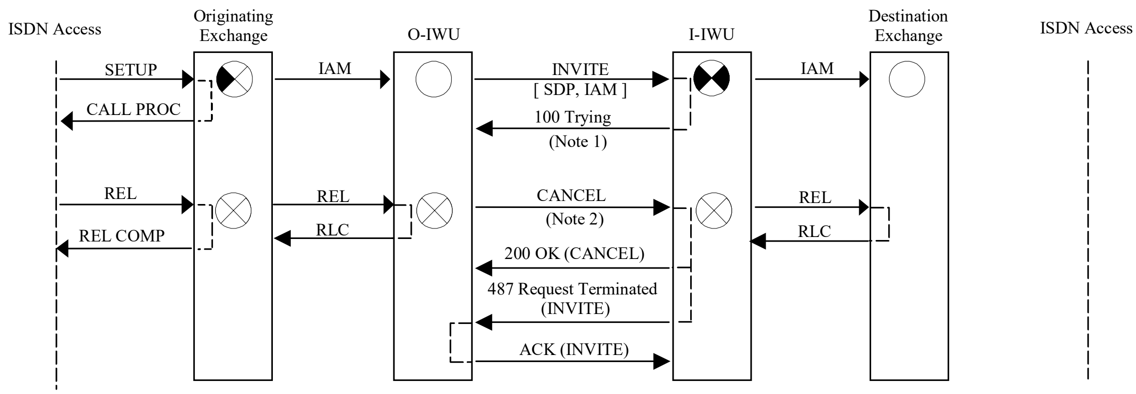
- *IAM message:* clauses 6.1.3 and 7.1
- *ACM message:* clauses 6.3 and 7.2

- *REL message:* clauses 6.13.2 and 7.8.3

8.1.1.1 D.3.2.2 Forward Release During Call Setup, No Early Dialog

See ATIS-1000113.2005, chapter 4, clause 2.1, and RFC 3261, clause 15.

Figure D.12 shows a forward release occurring during call setup, before a dialog has been established.



NOTE 1 – The generation of the 100 (Trying) response is necessary if the I-IWU knows that it will not generate a provisional or final response within 200 ms.

NOTE 2 – REL message can not be encapsulated in CANCEL message because it is a hop by hop request.

Figure D.12 - Forward release during call setup, no early dialog is established

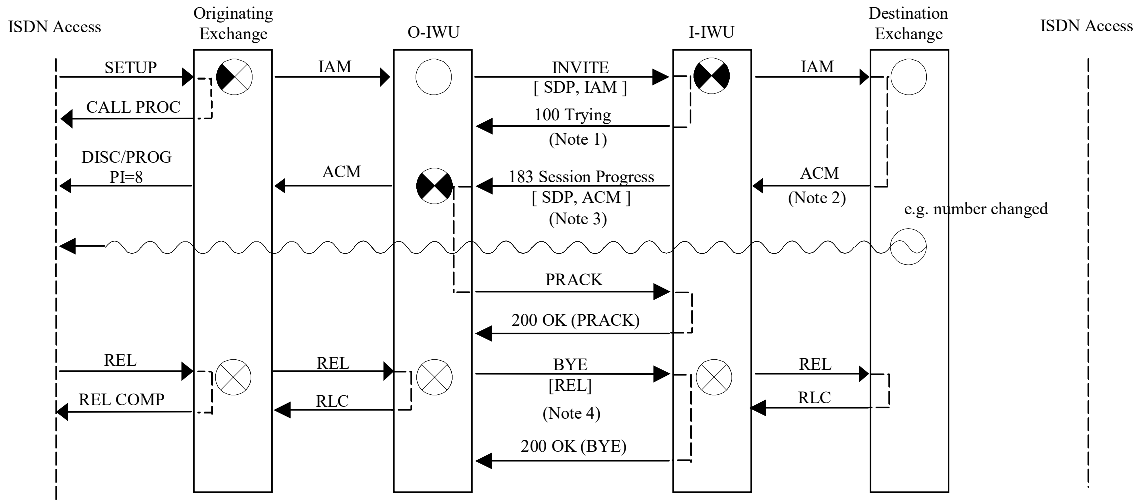
For detailed messages and parameter mapping, refer to:

- *IAM message:* clauses 6.1.3 and 7.1
- *REL message:* clauses 6.13.2 and 7.8.2

D.3.2.3 Forward Release During Call Setup, Early Dialog is Established

See ATIS-1000113.2005, chapter 4, clause 2.1, and RFC 3261, clause 15.

Figure D.13 shows an unsuccessful call set-up where certain tones and announcements are generated in the destination exchange during call establishment. O-IWU indicates the required support of reliable provisional responses by adding option tag 100rel to the Required header field of the INVITE request. The REL message is mapped and encapsulated in the BYE request as early dialog is already established through the reception of To-tag in 183 response.



NOTE 1 – The generation of the 100 (Trying) response is necessary if the I-IWU knows that it will not generate a provisional or final response within 200 ms.

NOTE 2 – The ACM is not mapped from a message from the destination user. It is independently generated at the destination exchange.

NOTE3 – 183 response contains the To-tag for the creation of early dialog.

NOTE4 – O-IWU releases the call with BYE request as early dialog is already established.

Figure D.13 - Forward release during call setup, early dialog is already established

For detailed messages and parameter mapping, refer to:

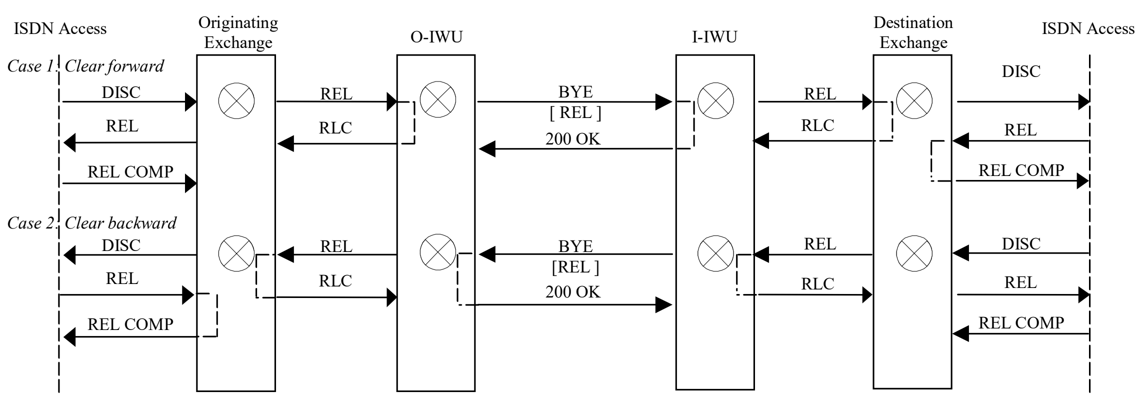
- *IAM message:* clauses 6.1.3 and 7.1
- *ACM message:* clauses 6.3 and 7.2
- *REL message:* clauses 6.13.2 and 7.8.2

D.3.3 Release Procedures/Call Flow Diagrams for Basic Call Control

D.3.3.1 Normal Call Release Procedure without Tone Provision

See ATIS-1000113.2005, chapter 4, clause 2.1, and RFC 3261, clause 15.

Figure D.14 shows the normal call release interworking procedures without tone provision. A REL message is mapped and encapsulated into BYE request to preserve the ISUP signaling transparency.



NOTE 1 – This procedure is applicable in those cases where in band tone/announcements are not provided, e.g. 64 kbit/s unrestricted bearer service.

Figure D.14 - Normal call release procedure without tone provision (Note 1)

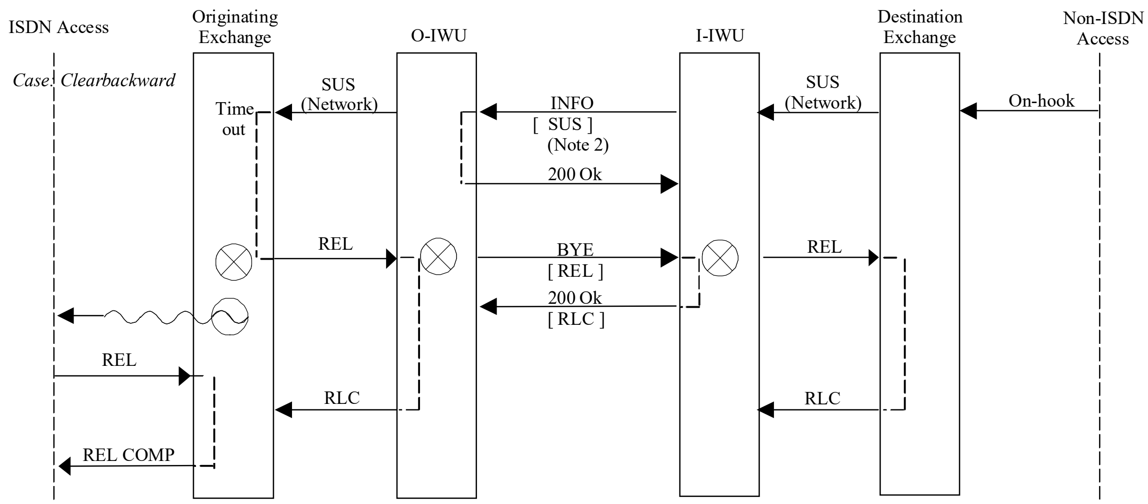
For detailed messages and parameter mapping, refer to:

- *REL message:* clauses 6.13.1, 7.8.2, and 7.8.3

D.3.3.2 Normal Release with SUS Message Encapsulation

See ATIS-1000113.2005, chapter 4, clause 2.1, and RFC 3261, clause 15.

Figure D.15 shows the normal call release procedure being initiated from the terminating non-ISDN access by means of a clear-back signal. At the destination exchange, the clear-back signal is mapped into a SUS with suspend/resume indicator (network initiated). At the I-IWU, the SUS message is mapped and encapsulated into INFO request. The originating ISDN exchange starts a timer. Upon expiry of the timer, if the originating exchange has not received a RES message, the originating exchange initiates clearing by a REL to the preceding exchange.



NOTE 1 – This procedure is applicable in those cases where in band tone/announcements are not provided, e.g. 64 kbit/s unrestricted bearer service.

NOTE 2 – The transparent transmission of SUS (network initiated)message is possible only in the case of ISUP encapsulation.

Figure D.15 - Normal release with SUS message encapsulation

For detailed messages and parameter mapping, refer to:

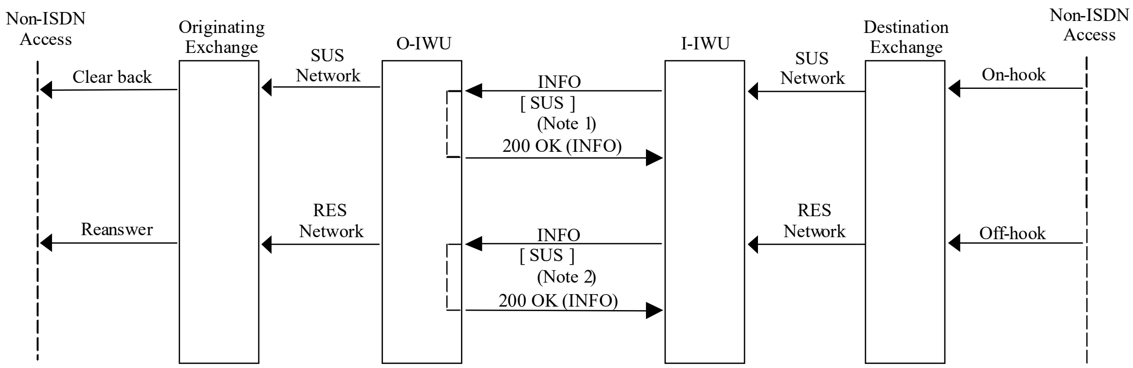
- *SUS message:* clause 6.11
- *REL message:* clauses 6.13.1 and 7.8.2

D.3.4 Suspend/Resume Procedures/Call Flow Diagrams for Basic Call Control

D.3.4.1 Suspend/Resume Non-ISDN Access to Non-ISDN Access

See ATIS-1000113.2005, chapter 4, clause 2.1, and RFC 3261, clause 14.

Figure D.16 illustrates suspend and resume procedures for non-ISDN access – non-ISDN access interworking over SIP-I. At the I-IWU, the SUS message is mapped and encapsulated into INFO request. At the O-IWU, the RES message is also mapped and encapsulated into INVITE request.



NOTE 1– Supervision Control in controlling exchange
NOTE 2– The transparent transmission of SUS/RES (network initiated) messages is possible only in case of ISUP encapsulation.

Figure D.16 - Suspend/resume non-ISDN access to non-ISDN access

For detailed messages and parameter mapping, refer to:

- *SUS message:* clause 6.11
- *RES message:* clause 6.12

Neither message requires interworking beyond de-encapsulation at the O-IWU.

Annex E: Formal Syntax of the P-Charge-Info Header

(normative)

The formal syntax of the P-Charge-Info header shown here is taken from IETF draft-york-dispatch-p-charge-info-01, section 7.

The syntax of the P-Charge-Info header is described as follows:

P-Charge-Info = "P-Charge-Info" HCOLON (name-addr / addr-spec); name-addr and addr-spec are specified in RFC 3261

charge-param = npi-param / noa-param / generic-param

npi-param = ";npi" EQUAL npi-value; generic-param is specified in RFC 3261

npi-value = gen-value

noa-param = ";noa" EQUAL noa-value

noa-value = gen-value

The SIP URI contained in the name-addr/addr-spec is the billing indicator that is passed between the parties.

charge-param is used as a userinfo parameter in P-Charge-Info.

The two optional parameters for PSTN interoperability are mentioned in the previous section and are:

- npi = "Numbering Plan Indicator"
- noa = "Nature of Address"

Values for the "npi-value" are listed in ATIS-1000113, chapter 3, section 3.6.

Values for the "noa-value" are listed in ATIS-1000113, chapter 3, section 3.10.