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SERIES Q: SWITCHING AND SIGNALLING, AND ASSOCIATED MEASUREMENTS AND TESTS

Specifications of signalling related to Bearer Independent Call Control (BICC)

Interworking between session initiation protocol (SIP) and bearer independent call control protocol or ISDN user part

Recommendation ITU-T Q.1912.5

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Recommendation ITU-T Q.1912.5

Interworking between session initiation protocol (SIP) and bearer independent call control protocol or ISDN user part

Summary

Recommendation ITU-T Q.1912.5 defines the signalling interworking between the bearer independent call control (BICC) or ISDN user part (ISUP) protocols and SIP in order to support services that can be commonly supported by BICC or ISUP and SIP-based network domains. It includes assured early dialogue, end-to-end support for ISDN user equipment, support for number portability and support for the interworking of supplementary services.

History

Edition	Recommendation	Approval	Study Group	Unique ID*
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Interworking between session initiation protocol (SIP) and bearer independent call control protocol or ISDN user part

1 Scope

This Recommendation defines the signalling interworking between the bearer independent call control (BICC) or ISDN user part (ISUP) protocols and session initiation protocol (SIP) with its associated session description protocol (SDP) at an interworking unit (IWU). The ISUP is defined in accordance with [ITU-T Q.761] to [ITU-T Q.764] and BICC is defined in accordance with [ITU-T Q.1902.1] to [ITU-T Q.1902.4]. The SIP and SDP are defined by the Internet Engineering Task Force (IETF). The capabilities of SIP and SDP that are needed to interwork with BICC or ISUP are defined in Annex C.

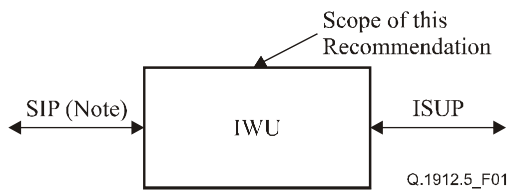
An IWU may stand alone or may be combined with an ISUP exchange or BICC interface serving node (ISN). It is assumed in this Recommendation that the initial service requests are forwarded and/or delivered via a trusted adjacent SIP node (ASN) within a SIP network domain. The ASN is viewed as a trusted network entity rather than an untrusted user entity, and thus the interface between the IWU and the ASN is a network-to-network interface (NNI). Where Profile C (SIP-I) is used, it is assumed that the remote SIP user agent is able to process ISUP. Support for SIP interworking at a user network interface (UNI) is out of the scope of this Recommendation. Interworking with forking in the SIP network is not specified in this Recommendation and is for further study.

The services that can be supported through the use of signalling interworking are limited to the services that are supported by BICC or ISUP and SIP-based network domains. Services that are common in SIP and BICC or ISUP network domains will interwork by using the function of an interworking unit (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 Recommendation is shown in Figures 1 and 2, respectively.

Figure 1 shows the scope of interworking between SIP and ISUP.

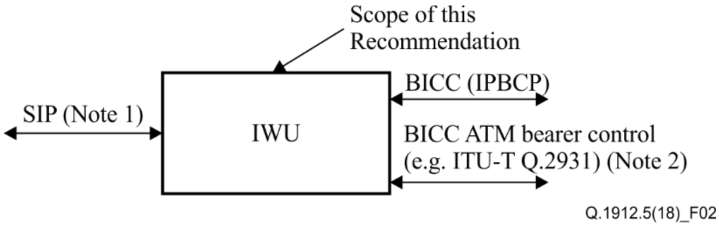
Items relating to security when interworking between two signalling systems in this Recommendation are for further study.



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

Figure 1 – Scope of interworking between SIP and ISUP

Figure 2 shows the scope of interworking between SIP and BICC.



NOTE 1 – The content consists of the SIP headers and message body.
NOTE 2 – Interworking with ATM bearer control in not specified in this Recommendation.

Figure 2 – Scope of interworking between SIP and BICC

[b-ITU-T Q.Sup.45] specifies the set of common capabilities supported by the interworking between SIP and BICC/ISUP for three different profiles (A, B, and C) in the form of tables. Tables 1 and 2 of [b-ITU-T Q.Sup.45] specify the interworking capabilities for Profile A, tables 3 and 4 specify the interworking capabilities for Profile B, and tables 5 and 6 specify the interworking capabilities for Profile C (SIP-I), respectively. The details on the capabilities supported by the different profiles, and all profiles in common, are shown in clause C.1.1.2.

Administrations may require operators to take into account national requirements in implementing this Recommendation, and in particular, in determining the local trust policy for the IWU.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

All IETF Standards Track RFC directly referenced by this Recommendation are listed in clause C.1.

[ITU-T E.164] Recommendation ITU-T E.164 (2010), *The international public telecommunication numbering plan.*

[ITU-T H.248.1] Recommendation ITU-T H.248.1 (2013), *Gateway control protocol: Version 3*.

[ITU-T Q.731.7] Recommendation ITU-T Q.731.7 (1997), *Stage 3 description for number identification supplementary services using Signalling System No. 7: Malicious call identification (MCID).*

[ITU-T Q.732.2-5] Recommendation ITU-T Q.732.2 (1999), *Stage 3 description for call offering supplementary services using Signalling System No. 7: Call diversion services: Call forwarding busy, call forwarding no reply, call forwarding unconditional, call deflection.*

[ITU-T Q.732.7] Recommendation ITU-T Q.732.7 (1996), *Stage 3 description for call offering supplementary services using Signalling System No. 7: Explicit call transfer*.

[ITU-T Q.733.1] Recommendation ITU-T Q.733.1 (1992), *Stage 3 description for call completion supplementary services using Signalling System No. 7: Call waiting (CW).*

[ITU-T Q.733.2] Recommendation ITU-T Q.733.2 (1993), *Stage 3 description for call completion supplementary services using Signalling System No. 7: Call hold (HOLD).*

[ITU-T Q.733.3] Recommendation ITU-T Q.733.3 (1997), *Stage 3 description for call completion supplementary services using Signalling System No. 7: Completion of calls to busy subscriber (CCBS)*.

[ITU-T Q.733.4] Recommendation ITU-T Q.733.4 (1993), *Stage 3 description for call completion supplementary services using Signalling System No. 7: Terminal portability (TP)*.

[ITU-T Q.733.5] Recommendation ITU-T Q.733.5 (1999), *Stage 3 description for call completion supplementary services using Signalling System No. 7: Completion of calls on no reply*.

[ITU-T Q.734.1] Recommendation ITU-T Q.734.1 (1993), *Stage 3 description for multiparty supplementary services using Signalling System No. 7: Conference calling*.

[ITU-T Q.734.2] Recommendation ITU-T Q.734.2 (1996), *Stage 3 description for multiparty supplementary services using Signalling System No. 7: Three-party service*.

[ITU-T Q.735.1] Recommendation ITU-T Q.735.1 (1993), *Stage 3 description for community of interest supplementary services using Signalling System No. 7: Closed user group (CUG)*.

[ITU-T Q.735.3] Recommendation ITU-T Q.735.3 (1993), *Stage 3 description for community of interest supplementary services using Signalling System No. 7: Multi-level precedence and preemption*.

[ITU-T Q.735.6] Recommendation ITU-T Q.735.6 (1996), *Stage 3 description for community of interest supplementary services using Signalling System No. 7: Global virtual network service (GVNS)*.

[ITU-T Q.736.1] Recommendation ITU-T Q.736.1 (1995), *Stage 3 description for charging supplementary services using Signalling System No. 7: International Telecommunication Charge Card (ITCC)*.

[ITU-T Q.736.3] Recommendation ITU-T Q.736.3 (1995), *Stage 3 description for charging supplementary services using Signalling System No. 7: Reverse charging (REV)*.

[ITU-T Q.737.1] Recommendation ITU-T Q.737.1 (1997), *Stage 3 description for additional information transfer supplementary services using Signalling System No. 7: User-to-user signalling (UUS)*.

[ITU-T Q.761] Recommendations ITU-T Q.761 to Q.764 (1999), *Specifications of Signalling System No. 7 ISDN User Part functional description.*

[ITU-T.Q.763] Recommendations ITU-T Q.761 to Q.764 (1999), *Specifications of Signalling System No. 7 ISDN User Part general formats and codes*.

[ITU-T Q.764] Recommendations ITU-T Q.761 to Q.764 (1999), *Specifications of Signalling System No. 7 ISDN User Part signalling procedures*.

[ITU-T Q.769.1] Recommendation ITU-T Q.769.1 (1999), *Signalling system No. 7 – ISDN user part enhancements for the support of number portability*.

[ITU-T Q.850] Recommendation ITU-T Q.850 (1998), *Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN user part*.

[ITU-T Q.931] Recommendation ITU-T Q.931 (1998), *ISDN user-network interface layer 3 specification for basic call control*.

[ITU-T Q.1902.x] Recommendations ITU-T Q.1902.1 to Q.1902.4 (2001), *Specifications of the Bearer Independent Call Control (Capability Set 2)*.

[ITU-T Q.1970] Recommendation ITU-T Q.1970 (2006), *BICC IP bearer control protocol.*

[ITU-T Q.1990] Recommendation ITU-T Q.1990 (2001), *BICC Bearer Control Tunnelling Protocol.*

[ITU-T Q.3403] Recommendation ITU-T Q.3403 (2016), *IP multimedia call control protocol based on the session initiation protocol and the session description protocol - Basic call: Requirements for the user side and the network side*.

[ITU-T Q.3617] Recommendation ITU-T Q.3617 (2015), *Terminating identification presentation and terminating identification restriction using IP multimedia core network subsystem. Protocol specification*.

[ITU-T Q.3620] Recommendation ITU-T Q.3620 (2016), *Communication diversion using IP multimedia core network subsystem – Protocol specification*.

[ITU-T Q.3621] Recommendation ITU-T Q.3621 (2016), *CONF using IP multimedia core network subsystem – Protocol specification*.

[ITU-T Q.3622] Recommendation ITU-T Q.3622 (2016), *Communication waiting using IP multimedia core network subsystem — Protocol specification*.

[ITU-T Q.3624] Recommendation ITU-T Q.3624 (2016), *Malicious communication identification using IP multimedia core network subsystem – Protocol specification*.

[ITU-T Q.3625] Recommendation ITU-T Q.3625 (2016), *Completion of communications to busy subscriber and completion of communications by no reply using IP multimedia core network subsystem – Protocol specification*.

[ITU-T Q.3627] Recommendation ITU-T Q.3627 (2016), *Closed user group using IP multimedia core network subsystem – Protocol specification*.

[ITU-T Q.3629] Recommendation ITU-T Q.3629 v.1 (2016), *Interworking between the IP multimedia core network subsystem and circuit switched networks – Protocol specification*.

[ITU-T T.38] Recommendation ITU-T T.38 (2015), *Procedures for real-time Group 3 facsimile communication over IP networks*.

3 Definitions

For BICC or ISUP specific terminology, reference shall be made to [ITU-T Q.1902.2]. For SIP and SDP specific terminology, reference shall be made to [IETF RFC 3261] and [IETF RFC 2327] respectively. Definitions for additional terminology used in this interworking Recommendation are as follows:

3.1 incoming or outgoing: This term is used in this Recommendation to indicate the direction of a call (not signalling information) with respect to a reference point.

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

3.3 incoming SIP or BICC/ISUP [network]: The network, from which the incoming calls are received, uses the SIP or BICC/ISUP protocol. Without the term "network", it simply refers to the protocol.

3.4 Outgoing Interworking Unit (O-IWU): This physical entity, which can be combined with a BICC ISN or ISUP exchange, terminates incoming calls using BICC or ISUP protocols and originates outgoing calls using the SIP.

3.5 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 [IETF RFC 3261].

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

3.7 SIP precondition: Indicates the support of the SIP "precondition procedure" as defined in [IETF RFC 3312].

3.8 Profile C (SIP-I): This phrase refers to the use of SIP with a message body that encapsulates ISUP information according to the requirements in this Recommendation.

3.9 Type 1 gateway: An interworking unit (IWU) capable of bearer control as well as call control. The IWU interworks between SIP and BICC or ISUP. Bearer control interworking is an internal operation.

NOTE – Because it is internal, bearer control interworking for Type 1 gateways is not specified in this Recommendation.

3.10 Type 2 gateway: An interworking unit capable of call control but not bearer control. The IWU interworks between SIP and BICC. Bearer control interworking is between the external bearer control protocol on the BICC side and SDP within SIP.

NOTE – Bearer control interworking for Type 2 gateways in the particular case of IP Bearer Control (IPBCP) is specified in Annex A.

3.11 Type 3 gateway: An interworking unit capable of bearer control as well as call control. The IWU interworks between SIP-I and BICC or ISUP. Bearer interworking is an internal operation.

NOTE – Because it is internal, bearer control interworking for Type 3 gateways is not specified in this Recommendation.

3.12 Type 4 gateway: An interworking unit capable of call control but not bearer control. The IWU interworks between SIP-I and BICC. Bearer control interworking is between the external bearer control protocol on the BICC side and SDP within SIP.

NOTE – Bearer control interworking for Type 4 gateways in the particular case of IP Bearer Control (IPBCP) is specified in Annex A.

In addition, this Recommendation makes use of the terms header field, message, message body, method, request, provisional and final response, dialogue and User Agent, which are defined in section 6 of [IETF RFC 3261]. It uses the term payload type as defined in [IETF RFC 3550], and static and dynamic payload type as defined in that RFC. Finally, it uses the terms attribute and session as defined in [IETF RFC 2327].

Within this Recommendation the following terminology is used:

– "pass to BICC/ISUP procedures" describes an operation internal to the IWU;
– "send" describes the transmission of a message on the applicable external network interface.

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

General

ABNF Augmented Backus-Naur Form (see [IETF RFC 2234])

AMR Adaptive Multi-rate (codec)

ASN Adjacent SIP Node

ATM Asynchronous Transfer Mode

B2BUA	Back-to-Back User Agent
BICC	Bearer Independent Call Control
BC-IWF	Bearer Control-Interworking Function
BNC	Backbone Network Connection
BNF	Backus-Naur Form
CC	Country Code
CLI	Calling Line Identification
CONN	CONNECT message (see [ITU-T Q.931])
DISC	DISCONNECT message (see [ITU-T Q.931])
FFS	For Further Study.
I-IWU	Incoming (to BICC/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
MGW	Media Gateway
MIME	Multi-purpose Internet Mail Extensions
NDC	National Destination Code
NNI	Network-to-Network Interface
O-IWU	Outgoing (from BICC/ISUP) Interworking Unit
PSTN	Public Switched Telephone Network
PT	Payload Type
RFC	Request For Comments
RTP	Real-time Transport Protocol
SCCP	Signalling Connection Control Part
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SIP-I	SIP with encapsulated ISUP
SN	Subscriber Number
TLS	Transport Layer Security
UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
UNI	User-to-Network Interface
URI	Universal Resource Identifier

BICC/ISUP messages

ACM	Address Complete Message
ANM	Answer Message
APM	Application Transport Mechanism

BAT	Bearer Association Transport
CGB	Circuit Group Blocking
CON	Connect message
COT	Continuity message
CPG	Call progress
GRS	Circuit Group Reset message
IAM	Initial Address Message
REL	Release message
RES	Resume message
RLC	Release Complete
RSC	Reset Circuit message
SGM	Segmentation Message
SAM	Subsequent Address Message
SUS	Suspend message

BICC/ISUP parameters and values

ACgPN	*"additional calling party number"* (value of Number Qualifier indicator within Generic Number)
APP	Application Transport Parameter
APRI	Address Presentation Restricted Indicator
ATP	Access Transport Parameter
BCI	Backward Call Indicator
CgPN	Calling Party Number
CIC	Circuit Identification Code (ISUP)
CIC	Call Instance Code (BICC)
FCI	Forward Call Indicator
HLC	High Layer Compatibility
NOA	Nature of Address indicator
NP	*"network provided"* (Screening Indicator value)
TMR	Transmission Medium Requirement
UPVP	*"user provided, verified and passed"* (Screening Indicator value)
USI	User Service Information

5 Conventions and Methodology

5.1 Conventions for representation of BICC/ISUP PDU

1) The first letter of each major word is capitalized in the names of BICC/ISUP:
   - messages (e.g., Initial Address Message, User-to-user Information);
   - parameters (e.g., Nature of Connection Indicators, Calling Party's Category); and
   - parameter information (e.g., Nature of Address Indicator, Address Signals, Cause Value).

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.

   Example: 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 section 25 of [IETF 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 user agent (UA) and shall support the applicable RFCs as indicated in clause C.1. The ISUP interface shall support the protocol as defined in the ISUP [ITU-T Q.761] to [ITU-T Q.764] (1999). The BICC interface shall support the protocol as defined in the BICC [ITU-T Q.1902.1] to [ITU-T Q.1902.4].

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

– For profiles A and B, the IWU shall act as a Type A exchange for the purposes of ISUP and BICC compatibility procedures.
– For Profile C (SIP-I): for the mapping of BICC/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.

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 either one of the signalling systems: SIP, ISUP or BICC), are outside the scope of this Recommendation, as they cannot be interworked.

Where the IWU is combined with a BICC ISN or an ISUP exchange, it shall provide interworking between the bearer network connections on the SIP and on the ISUP or BICC network domain sides.

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 determining that the 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).

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 determining 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).

5.3.1 Identification of call, dialogue and call control association

The IWU shall establish a one-to-one relationship between a SIP dialogue and a BICC/ISUP call/bearer control instance so that interworking is between signalling information related to the same call. For overlap sending, the same BICC/ISUP call/bearer control instance at the IWU may be associated with a succession of SIP dialogues until address signalling is complete.

5.3.2 General principles specific to Profile C (SIP-I)

In the case of Profile C (SIP-I), the following ISUP timers defined in [ITU-T Q.764] shall not be supported by ISUP procedures on the SIP side of the IWU: T1, T4, T5, T10, T12 to T32, T36 and T37.

Where the SIP dialogue terminates and the ISUP state machine is still running (except as provided by clauses 6.2 and 7.2.1, dealing with overlap sending), an implementation-dependent function will release the call.

The following general principles of ISUP encapsulation apply within this Recommendation.

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 Recommendation. 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 may be required to construct an appropriate ISUP message using the SIP information received. Clauses 6 and 7 provide all the information that the IWU requires to be able to perform this task.

5.3.3 Interworking of ISUP overlap signalling

This Recommendation provides the interworking procedures for the case when overlap signalling is propagated into the SIP network and the case where overlap signalling is converted to *en bloc* signalling at the O-IWU. Additionally, procedures are outlined (in clause 6) to address situations where overlap signalling is received on the SIP side of the I-IWU. While this Recommendation covers procedures for propagating overlap signalling across the SIP network, it is recommended that SIP *en* *bloc* signalling is used, i.e., the use of overlap signalling within the SIP network should be avoided. Thus, the preferred scenario is to convert ISUP overlap signalling to SIP *en bloc* signalling at the O-IWU. Nevertheless, the decision regarding how to configure a particular IWU with respect to overlap signalling is a matter of local policy/network configuration.

In the particular case of SIP overlap to ISUP overlap signalling interworking at the I-IWU, the SIP network must deliver all INVITEs with the same Call-ID and From tag which have enough addressing information to reach an I-IWU to the same I-IWU.

Detailed overlap procedures are provided within the appropriate sections in clauses 6 and 7 of this Recommendation.

NOTE 1 – When an O-IWU knows that a SIP network will be used as a transit network between two PSTN end points, it may find it appropriate to propagate overlap signalling through the SIP network, so that ISUP overlap signalling appears in the destination ISUP network.

NOTE 2 – It is expected that INVITEs will be delivered in order to the I-IWU. The I-IWU does not buffer and reorders INVITEs that it receives as part of an overlapped call; instead, by analysing the Request-URI, it determines if the INVITE received is the most recent INVITE based upon the number of digits present compared with the number of digits that have already been received at the I-IWU. Procedures within clause 6 outline how the I-IWU processes any INVITEs that are received out of sequence.

5.4 ISUP encapsulation – detailed procedures

This clause is relevant to Profile C (SIP-I) only. It builds on the general principles of ISUP encapsulation outlined in clause 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 clause 5.4.3) in a relevant SIP message (see clause 5.4.1.3). Setting of header fields relating to the handling of the ISUP body is specified in clause 5.4.1.2.

Similarly, an I-IWU receiving backwards ISUP information which is not excluded from encapsulation (see clause 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 clause 5.4.1.3). Setting of header fields relating to the handling of the ISUP body is specified in clause 5.4.1.2.

5.4.1.2 Header fields for ISUP MIME bodies

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

Content-Type: application/ISUP; version = itu-t92+;

NOTE – itu-t92+ means ISUP '92 plus every later ISUP version. However, no action is taken by the IWU 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 which SIP message to use to encapsulate the ISUP message

For basic call set-up, 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 Recommendation 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 from the O-IWU.

For other messages see clause 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 BICC/ISUP network. This processing is specified in clauses 5.4.2.1.1 to 5.4.2.1.3.

5.4.2.1.1 Alignment of SIP headers and ISUP body contents

On receipt of a SIP message containing encapsulated ISUP, the IWU shall use the procedures outlined in this Recommendation 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). The alignment rules regarding which header overrides which BICC/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 profiles A and/or B as described. For profile C (SIP-I) the ISUP field shall be derived from the encapsulated ISUP MIME body and local policy.

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

5.4.2.1.2 Setting of ISUP parameters by IWU

After following the procedures in clause 5.4.2.1.1, 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.

5.4.2.1.3 Passing resulting ISUP message to BICC/ISUP procedures and sending of message

After following the procedures in clause 5.4.2.1.2, the IWU shall pass the ISUP information to the relevant BICC/ISUP procedures. The message (if any) which results from the application of the relevant BICC/ISUP procedures is the message which is sent on the BICC/ISUP interface.

5.4.3 Exclusions and special considerations

The ISUP messages listed in Table 5-1 are either not encapsulated within any SIP message, or receive a special treatment with regard to ISUP encapsulation. The clause number shown in the reference column for each message contains the procedures applicable to that message. This table applies not only to messages received on the BICC/ISUP side and interworked but also to messages generated internally.

NOTE – Table 5-1 shows only those messages within [ITU-T Q.763] which are not marked "national use". Messages marked "national use" (in [ITU-T Q.763]) are outside the scope of this Recommendation.

Table 5-1 – ISUP messages for special consideration

ISUP message	Reference clause
Reset Circuit	5.4.3.1 (Note 1)
Circuit Group Blocking	5.4.3.1
Circuit Group Blocking Acknowledgement	5.4.3.1
Group Reset	5.4.3.1

Table 5-1 – ISUP messages for special consideration

ISUP message	Reference clause
Circuit Group Reset Acknowledgement	5.4.3.1
Confusion	5.4.3.1 or 5.4.3.2 (Note 2)
Facility reject	5.4.3.1 or 5.4.3.2 (Note 2)
User to User information	5.4.3.2
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
Facility Accepted	5.4.3.2
Facility Request	5.4.3.2
User part test	5.4.3.1
User part available	5.4.3.1
Facility	5.4.3.2
Network Resource management	5.4.3.2
Identification Request	5.4.3.2
Identification response	5.4.3.2
Segmentation	5.4.3.3
Loop prevention	5.4.3.2
Application Transport	5.4.3.2
Pre-Release information	5.4.3.2
Release Complete	5.4.3.4
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). NOTE 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 the 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 dialogue 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 (BICC/ISUP side interface) will reassemble 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 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 BICC/ISUP procedures.

5.5 sip: and sips: URIs

Wherever this Recommendation makes reference to a sip: URI as defined in [IETF 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 BICC/ISUP at I-IWU

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

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

The I-IWU receives forward and backward signalling information from the incoming SIP and outgoing BICC/ISUP sides, respectively. After receiving this signalling information and performing appropriate call/service processing, the I-IWU may signal forward to subsequent BICC/ISUP nodes or backward to preceding SIP entities. This clause specifies the signalling interworking requirements for basic call at the I-IWU. This clause is split into subclauses based upon the messages sent or received on the outgoing BICC/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 that:

a) the I-IWU supports originating basic calls only; and

b) calls originated from the SIP network domain do not require equivalent PSTN/ISDN service interworking.

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

In the case of Type 2 or Type 4 gateways, as defined in [b-ITU-T Q.Sup.45], the I-IWU shall (in addition to the procedures outlined within this clause) follow the BICC-specific procedures outlined in clause A.2.

The I-IWU shall include a To tag in the first backward non-100 provisional response, in order to establish an early dialogue as described in section 12 of [IETF RFC 3261].

For Profile C (SIP-I) operation, ISUP message segmentation must be handled as described in clause 5.4.3.3.

The receipt of a P-Early-Media header set to 'supported' indicates the capability of the remote network entity to authorize early media.

6.1 Sending of Initial Address Message (IAM)

If an INVITE is received which has enough digits to route to the BICC/ISUP network and which cannot be associated with an existing call, the IAM resulting from the "receipt of INVITE" interworking procedures (see clauses 6.1.1 and 6.1.2) or (in the case of Profile C operation) the de-encapsulated IAM (as updated by the SIP-ISUP interworking procedures within clause 6.1.3 and associated subclauses) shall be passed to BICC/ISUP procedures. For the overlap operation only, if an INVITE is received with the same Call-Id and From tag values as the previous INVITE for which a call is currently active, the procedures of clause 6.2 apply.

NOTE – If an INVITE is received which does not have enough digits to route to the BICC/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 first INVITE for which an IAM is sent. The procedures for the sending of the IAM then depend on whether the INVITE received from the SIP network contains an SDP Offer. See clauses 6.1.1 and 6.1.2.

The IAM parameters are coded according to clause 6.1.3.

6.1.1 INVITE received without an SDP offer

Upon receipt of the first INVITE with sufficient digits for an IAM to be sent, 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 a media description, the content of which is determined using local policy within a 183 Session Progress message, subject to the following rules if the I-IWU operates as an international incoming gateway and if ITU-T G.711 encoding is used:

   i) If the call is to be routed to an A-law PSTN network, then it shall send an SDP offer with A-law (PCMA), but not µ-law (PCMU) included in the media description.

   ii) If the call is to be routed to a µ-law PSTN network, then it shall send an SDP offer with both A-law (PCMA) and µ-law (PCMU) included in the media description and µ-law (PCMU) shall take precedence over A-law (PCMA).

   These procedures reflect the requirement that transcoding between A-law and µ-law must occur in µ-law networks only.

   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 the procedure described in item 2 of clause 6.1.2.

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 or continuation from clause 6.1.1 1)

If the I-IWU operates as an international incoming gateway, and if ITU-T G.711 encoding is used, then the following procedures apply. These procedures reflect the requirement that transcoding between A-law and µ-law must occur in µ-law networks only.

i) If the call is to be routed to an A-law PSTN network then it shall delete µ-law (PCMU), if present, from the media description that it will send back in the SDP answer.

ii) If the call is to be routed to a µ-law PSTN network, and if both A-law (PCMA) and µ-law (PCMU) were present in the offer, then the I-IWU shall delete A-law (PCMA) from the media description that it will send back in the SDP answer.

The processing continues as follows:

1) If SIP preconditions are not in use, the I-IWU shall immediately send out the IAM.

2) If SIP preconditions are in use, then:

   a) If outgoing BICC/ISUP signalling on the subsequent network supports the use of the continuity check procedure, the IAM shall be sent out immediately on the BICC/ISUP side with the following coding of the Nature of Connection Indicators parameter:

      i) If the subsequent network is a BICC network: The Continuity indicator of the Nature of Connection Indicators parameter shall be set to "*COT to be expected*".

      ii) If the subsequent network is an ISUP network: The Continuity check indicator in the Nature of Connection Indicators parameter is 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.

      iii) The 183 Session Progress indicates the support of preconditions shall not include the P-Early-Media header field.

   b) If outgoing BICC/ISUP signalling on a subsequent network does not support the use of the continuity check procedure, the sending of the IAM shall be deferred until all preconditions have been met.

In all cases, clause 6.1.3 gives specific details related to the population of specific parameters of the IAM. Table 6-1 gives a summary of parameters within the IAM that are interworked from the INVITE along with a reference to the subclauses of clause 6.1.3 in which the specific interworking is described.

The I-IWU shall reject an INVITE request for a session only containing unsupported media types by sending a status code 488 "Not Acceptable Here". If several media streams are contained in a single INVITE request, and if the I-IWU does not support multimedia interworking, then the I-IWU shall select one of the supported media streams, reserve the codec(s) for that media stream, and reject the other media streams and unselected codecs in the SDP answer, as detailed in [IETF RFC 3264]. If supported audio media stream(s) and supported non-audio media stream(s) are contained in a single INVITE request, an audio stream should be selected.

The I-IWU shall include a To tag in the first backward non-100 provisional response, in order to establish an early dialogue as described in [IETF RFC 3261].

If an IWU discovers an emergency call it shall, depending on national requirements, map that to the appropriate indication in ISUP.

6.1.3 IAM parameters

Table 6-1 indicates the IAM parameters that interwork from SIP.

Table 6-1 – Interworked contents of the Initial Address Message

Parameter	Clause
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
Transmission Medium Requirement	6.1.3.5
Transmission Medium Requirement Prime	6.1.3.5.2
Calling Party Number	6.1.3.6.1
Generic Number	6.1.3.6.2
User Service Information	6.1.3.7
User Service Information Prime	6.1.3.7
Application Transport: BAT (BICC only)	6.1.3.8
Hop Counter	6.1.3.9

6.1.3.1 Called Party Number (mandatory)

It is required that the Request-URI contain a sip: URI with the user = phone parameter, where the userinfo part of the URI is an E.164 number encoded as specified by the telephone-subscriber rule of [IETF RFC 2806]. 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. Table 6-2 summarizes this mapping.

Table 6-2 – Coding of the Called Party Number

INVITE→	IAM→
Request-URI	Called Party Number
E.164 address (format +CC NDC SN) (e.g., as User info portion of a SIP URI with user=phone, or as tel URI)	**Address Signal**: Analyse the information contained in received E.164 address. If CC is the country code of the network in which the next hop terminates, then remove "+CC" and use the remaining digits to fill the Address signals. If CC is not the country code of the network in which the next hop terminates, then remove "+" and use the remaining digits to fill the Address signals. (Note 2)
	Odd/even Indicator: set as required
	Nature of Address Indicator: Analyse the information contained in received E.164 address. If CC is the country code of the network in which the next hop terminates, then set Nature of Address Indicator to "National (significant) number. If CC is not the country code of the network in which the next hop terminates, then set Nature of Address Indicator to "International number'. (Note 1)

Table 6-2 – Coding of the Called Party Number

INVITE→	IAM→
	Internal Network Number Indicator: 1 routing to internal network number not allowed
	Numbering Plan Indicator: 001 ISDN (Telephony) numbering plan ([ITU-T. E.164])
NOTE 1 – The usage of "nature of address indicator" value "unknown" is allowed but the mapping is not specified in this Recommendation. NOTE 2 – If PSTN XML and ISUP Sending Terminated (ST) signal are supported as a network option, then the PSTN XML sendingCompleteIndication, if present, is mapped to the sending terminated digit (hexadecimal digit F) in the address signals field of the Called Party Number parameter.	

6.1.3.2 Calling Party's Category (mandatory)

For profiles A and B, the following codes should be set by the I-IWU as default in the Calling Party's Category parameter.

Bits/Codes	Meaning
0000 1010	"*Ordinary calling subscriber*"

Table 6-3 shows the mapping of a "cpc" URI parameter received within tel URI or the userinfo part of SIP URI with user="phone" in a P-Asserted-Identity header in the initial INVITE request to the Calling Party's Category parameter in the ISUP IAM. When the "cpc" URI parameter value "operator" is received the I-IWU shall use an Accept-Language header field to determine the value of the Calling Party's Category parameter.

Table 6-3 – Mapping of the CPC parameter to the ISUP Calling Party's Category parameter

SIP Parameters		ISUP Parameters
"cpc" URI parameter in P-Asserted-Identity (Note 2)	**Accept-Language**	**Calling Party's Category**
operator	fr	operator, language French
operator	en	operator, language English
operator	de	operator, language German
operator	ru	operator, language Russian
operator	es	operator, language Spanish
ordinary		ordinary calling subscriber
test		test call
payphone		payphone
unknown		calling party's category unknown at this time (national use)
mobile-hplmn		mobile terminal located in the home PLMN
mobile-vplmn		mobile terminal located in a visited PLMN

Table 6-3 – Mapping of the CPC parameter to the ISUP Calling Party's Category parameter

SIP Parameters	ISUP Parameters
NOTE 1 – This is a national/regional specific value. Interworking shall only occur when interconnecting with indicated national network. NOTE 2 – In case the "cpc" URI parameter is absent or contains values that are not mapped as per this table then the ISUP shall contain the default CPC value "ordinary calling subscriber".	

In case the Accept-Language header field is not received or contains values that are not in this table then based on operator policy the Calling Party's Category parameter shall contain the CPC value "operator, language X" (where X is one of the following languages: French, English, German, Russian or Spanish) or national/regional specific value.

For Profile C (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 Indicators parameter which are set by the I-IWU are as follows:

Bits	Indicators in Nature of Connection Indicators parameter
AB	Satellite Indicator
DC	Continuity Check Indicator (ISUP)/ Continuity Indicator (BICC)
E	Outgoing Echo Control Device

Other fields in the Nature of Connection Indicators should follow the current BICC/ISUP Recommendation.

For profiles A and B: The codes in Table 6-4 should be set by the I-IWU as default in the Nature of Connection Indicators parameter fields:

Table 6-4 – Default Nature of Connection Indicator values

Bits	Codes	Meaning	Conditions
AB	01	"*No satellite circuit in the connection*"	
DC (Note)	00	*"Continuity check not required (ISUP)/no COT to be expected (BICC)"*	Without pending precondition request (all profiles).
	10	*"Continuity check performed on a previous circuit (ISUP)/COT to be expected (BICC)"*	With pending precondition request (all profiles).
	1	*"Outgoing echo control device included"*	For speech calls, e.g., TMR is "3.1KHz audio".
	0	*"Outgoing echo control device not included*"	For known data calls, e.g., TMR "64 kBit/s unrestricted" **or** HLC "Facsimile Group 2/3".
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 clause 6.3.			

For Profile C (SIP-I), with the exception of Continuity Indicator (BICC)/Continuity Check Indicator (ISUP) which receives special treatment in clauses 6.1.1 and 6.1.2, 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 parameter which are set by the I-IWU, are as follows:

Bits	Indicators in FCI parameter
CB	End-to-end Method Indicator
D	Interworking Indicator
E	End-to-end Information Indicator (national use)
F	ISUP/BICC Indicator
HG	ISUP/BICC Preference Indicator
I	ISDN Access Indicator

Other fields in the FCI parameter should follow the current BICC/ISUP Recommendation.

For profiles A and B, the indicator values in Table 6-5 should be set by the I-IWU as default in the FCI parameter:

Table 6-5 – Default values for Forward Call Indicators

Bits	Codes	Meaning
CB	00	*"No end-to-end method available (only link-by-link method available)"*
D	1	*"Interworking encountered".*
	0	*"No interworking encountered"* NOTE 1
E	0	*"No end-to-end information available"*
F	0	*"ISDN user part/BICC not used all the way".*
	1	*"ISDN user part/BICC used all the way"* NOTE 1
HG	01	*"ISDN user part/BICC not required all the way"*
	00	*"ISDN user part/BICC preferred all the way"* NOTE 2
	10	*"ISDN user part/BICC required all the way"* NOTE 2
I	0	*"Originating access non-ISDN"*
	1	*"Originating access ISDN" NOTE 1*
NOTE 1 – As a network operator option if the TMR = 64 kBit/s unrestricted is used NOTE 2 – Depending on operator policy		

For profiles A and B, the appropriate values of the FCI parameter are determined based on analysis of various parameters (from signalling, internal states or configuration) at the I-IWU.

For Profile B and if the PSTN XML is supported as a network option (see Annex H), the Forward Call Indicators derived as shown in Table 6-6 shall take precedence.

Table 6-6 – Mapping of PSTN XML elements to Forward Call Indicators parameter

INVITE →	IAM→
PSTN XML	**Forward Call Indicators parameter**
PSTN XML with Progress Indicator with Progress Description value 6 **(Meaning: originating access ISDN)**	bit D Interworking Indicator 0 "no interworking encountered (No. 7 signalling all the way)" bit F ISDN User Part Indicator 1 "ISDN User Part used all the way" bit I ISDN Access Indicator 1 "originating access ISDN"

Progress Indicator with Progress Description value "6" shall not be included in an ATP within the IAM.

For Profile C (SIP-I), the Forward Call Indicators parameter shall be generated by the I-IWU using the Forward Call Indicators parameter present within the received encapsulated ISUP message.

6.1.3.5 Transmission Medium Requirement (mandatory), Transmission Medium Requirement Prime (optional), User Service Information (optional), User Service Information Prime (optional)

For Profile A, the TMR parameter is set to 3.1 kHz audio, the USI parameter is not sent and transcoding is applied when required. The remainder of this clause applies to profiles B and C.

For profiles A and B

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 TMR, USI and HLC shall be derived from the SDP as described in clause 6.1.3.5.1. Otherwise, they shall be set in accordance with local policy.

If ITU-T G.711 is used, the I-IWU is an international gateway, and the incoming call is treated as an ISDN originated call, then the User Information Layer 1 Protocol Indicator of the USI parameter shall be set in accordance with the encoding law of the subsequent BICC/ISUP network.

If the I-IWU transcodes, it shall select the TMR parameter to "3.1 kHz audio".

If the I-IWU supports the PSTN XML body as a network option, and if a PSTN XML body is received in the INVITE request and the I-IWU selects media encoded in any of the formats in Table 6-11 (ITU-T G.711, CLEARMODE or t38) among the offered media, the I-IWU shall derive these parameters from the XML body instead, as detailed in Table 6-11.

The I-IWU should only apply the mapping in tables 6-7 and 6-9 if the TMR and USI values derived from the selected codec according to Table 6-11 are equivalent with the values within the first Bearer Capability element in the PSTN XML; otherwise the I-IWU should apply the mapping according to Table 6-11.

6.1.3.5.1 Transmission Medium Requirement

For *Profile* B and as a network option if the PSTN XML schema is supported, the following interworking applies:

The first PSTN XML BearerCapability element appears within the INVITE Request is mapped into the Transmission medium requirement in the sent IAM as described in Table 6-7.

Table 6-7 – Mapping of PSTN XML element into ISUP Transmission Medium Requirement parameter

INVITE →		IAM →	
PSTN XML	**Value**	**ISUP Parameter**	**Content**
BearerCapability (InformationTransferCapability)	00000	TMR	Speech
	10000		3.1 kHz audio
	01000		64 kbit/s unrestricted

6.1.3.5.2 Transmission Medium Requirement Prime (optional)

For Profile B and as a network option if the PSTN XML schema is supported and the I-IWU supports forwarding fallback signalling, the following interworking applies:

If two PSTN BearerCapability elements appear within the INVITE Request and the following conditions apply:

– one m-line with at least two formats;

– the first stated codec is a CLEARMODE codec;

– the second stated codec is an ITU-T G.711 codec;

– the second PSTN BearerCapability element (InformationTransferCapability) is set to 'Unrestricted digital information with tones/announcements' (10001),

deviant from the mapping described in Table 6-7, the Transmission Medium Requirement parameter in the sent IAM is set to "64 kBit/s preferred". The Transmission Medium Requirement Prime parameter is set as described in Table 6-8.

Table 6-8 – Mapping of PSTN XML element into ISUP Transmission Medium Requirement Prime parameter

INVITE →		IAM →	
PSTN XML	**Value**	**ISUP Parameter**	**Content**
BearerCapability (InformationTransferCapability)	00000	TMR Prime	Speech
	10000		3.1 kHz audio

6.1.3.5.3 User Service Information (optional)

For Profile B and as a network option if the PSTN XML schema is supported, the following interworking applies:

The first PSTN XML BearerCapability element appears within the INVITE Request is mapped into the ISUP User Service Information Parameter in the sent IAM as described in Table 6-9.

Table 6-9 – Mapping of PSTN XML element into ISUP User Service Information parameters

INVITE →		IAM →	
PSTN XML	**Value**	**ISUP Parameter**	**Content**
BearerCapability (InformationTransferCapability)	00000	User Service Information	Speech
	10000		3.1 kHz audio
	10001		Unrestricted digital information with tones/announcements

6.1.3.5.4 User Service Information Prime (optional)

For Profile B, and as a network option if the PSTN XML schema is supported and the I-IWU supports forwarding fallback signalling, the following interworking applies:

If two PSTN BearerCapability elements appear within the INVITE Request and the following conditions apply:

– one m-line with at least two formats;

– the first stated codec is a CLEARMODE codec;

– the second stated codec is an ITU-T G.711 codec;

– the second PSTN BearerCapability element (InformationTransferCapability) is set to 'Unrestricted digital information with tones/announcements' (10001).

The User Service Information Prime parameter in the sent IAM is set as described in Table 6-10.

Table 6-10 – Mapping of PSTN XML element into ISUP User Service Information Prime parameters

INVITE →		IAM →	
PSTN XML	**Value**	**ISUP Parameter**	**Content**
BearerCapability (InformationTransferCapability)	10001	User Service Information Prime	Unrestricted digital information with tones/announcements

6.1.3.5.5 No support of TMR "64 kBit/s preferred"

For profiles A and B, if TMR "64 kBit/s preferred" is not supported at the succeeding ISUP trunk or the I-IWU has the knowledge that the succeeding network is unable to perform the fallback:

– apply the procedures as described within clauses 6.1.3.5.1 and 6.1.3.5.3, using the first Bearer Capability element in the PSTN XML and the second codec in the m-line;

– discard the second Bearer Capability element in the PSTN XML;

– select the second format in the m-line within the SDP answer, and

– configure the MGW.

For Profile C (SIP-I)

The TMR, USI and HLC shall be taken from the encapsulated ISUP.

If the USI parameter is present in the encapsulated ISUP, ITU-T G.711 is used, and the I-IWU is an international gateway, then the User Information Layer 1 Protocol Indicator of the USI parameter shall be set in accordance with the encoding law of the subsequent BICC/ISUP network.

6.1.3.5.6 Transcoding not available at the I-IWU (profiles A and B only)

NOTE – If the outgoing signalling is BICC, the SDP will also interwork with other BICC parameters (APP with BAT) relating to the bearer control signalling information of the selected outgoing bearer. This additional interworking specification is addressed in Annex A.

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 parameters, TMR, USI and HLC.

The first subfield (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 kbit/s or "b=" line is absent, then TMR 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 the USI parameter should be set to "*G.711 μ–law*" or "*G.711 A law*".

Table 6-11 provides the default mapping relations based on the above procedure.

Table 6-11 – Coding of TMR/USI/HLC from SDP: SIP to BICC/ISUP

m= line			**b= line**	**a= line**	**TMR parameter**	**USI parameter (Note 1)**		**HLC parameter**
<media>	<transport>	<fmt-list>	<modifier>: <bandwidth-value> NOTE – <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.	a = rtpmap: <payload type> <encoding name>/ <clock rate> [/<encoding parameters>]	TMR codes	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	*"3.1 kHz audio"*	*"3.1 kHz audio"*	*"G.711 μ-law"*	(Note 3)
audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap: <payload type> PCMU/8000	*"3.1 kHz audio"*	*"3.1 kHz audio"*	*"G.711 μ-law"*	(Note 3)
audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A	*"3.1 kHz audio"*	*"3.1 kHz audio"*	*"G.711 A-law"*	(Note 3)
audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap: <payload type> PCMA/8000	*"3.1 kHz audio"*	*"3.1 kHz audio"*	*"G.711 A-law"*	(Note 3)
audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s RTP/UDP/IP overhead	rtpmap: <payload type> CLEARMODE/8000	*"64 kBit/s preferred" (Note 4)*	*"Unrestricted digital inf. w/tones/ann"*		
audio	RTP/AVP	Dynamic PT	AS:64 kbit/s	rtpmap: <payload type> CLEARMODE/8000 (Note 2)	*"64 kbit/s unrestricted"*	*"Unrestricted digital information"*		
image	udptl	t38	N/A or up to 64 kbit/s	Based on [ITU-T T.38]	*"3.1 kHz audio"*	*"3.1 kHz audio"*		*"Facsimile Group 2/3"*
image	tcptl	t38	N/A or up to 64 kbit/s	Based on [ITU-T T.38]	*"3.1 kHz audio"*	*"3.1 kHz audio"*		*"Facsimile Group 2/3"*

NOTE 1 – In this table, the codec G.711 is used only as an example. Other codecs are possible.

NOTE 2 – CLEARMODE is specified in [IETF RFC 4040].

NOTE 3 – HLC is normally absent in this case. It is possible for HLC to be present with the value "Telephony", although clause 6.3.1 of [b-ITU-T Q.939] indicates that this would normally be accompanied by a value of "Speech" for the Information Transfer Capability element.

NOTE 4 – The TMR value "64 k/bits preferred" should only be used if the Clearmode codec appears together with speech codecs in the same m-line and two PSTN XML BearerCapability elements appear in the initial INVITE request as described in clause 6.1.3.5.

6.1.3.6 BICC/ISUP Calling Line Identification (CLI) parameters

Table 6-12 summarizes the cases for mapping from the SIP INVITE header fields to the BICC/ISUP CLI parameters. Table 6-13 provides details when the Calling Party Number parameter is given a network provided value. Table 6-14 provides details for Calling Party Number parameter mapping in other cases. Finally, Table 6-15 provides details for mapping to the Generic Number parameter when this is possible.

For Profile C (SIP-I)

If the address within the Calling Party Number or Generic Number after application of the mapping in this clause and processing by BICC/ISUP procedures is the same as the respective value contained in the encapsulated ISUP, no additional interworking is needed for that parameter beyond the use of ISUP encapsulation. The contrary case is treated in the same way as for profiles A and B.

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

Table 6-12 – Mapping of SIP From/P-Asserted-Identity/Privacy header fields to BICC/ISUP CLI parameters

Has a SIP P-Asserted-Identity containing a URI (Note 2) with an identity in the format "+" CC + NDC + SN been received?					
	Has a SIP From (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**	**Generic Number ("*Additional calling party number*") Address Signals**	**Generic Number parameter APRI**
No	No	Network option to either include a network provided E.164 number (see Table 6-13) or omit the Address Signals. (Note 4)	If a Privacy header field was received, set APRI as indicated in Table 6-14, otherwise, network option to set APRI to "*presentation restricted*" or "*presentation allowed*" (Note 4)	Parameter not included	Not applicable
No	Yes	Network option to either include a network provided E.164 number (See Table 6-13) or omit the Address Signals. (Note 4)	If a Privacy header field was received, set APRI as indicated in Table 6-14, otherwise, network option to set APRI to either "*presentation restricted*" or "*presentation allowed*" (Note 4)	Network option to either omit the parameter (if CgPN has been omitted) or derive from the SIP From (see Table 6-15) (Note 1)	See Table 6-15
Yes	No	Derive from SIP P-Asserted-Identity (See Table 6-14)	APRI = "*presentation restricted*" or "*presentation allowed*" depending	Not included	Not applicable

Table 6-12 – Mapping of SIP From/P-Asserted-Identity/Privacy header fields to BICC/ISUP CLI parameters

			on SIP Privacy header. (See Table 6-14)		
Yes	Yes	Derived from SIP P-Asserted-Identity (See Table 6-14)	APRI = "*presentation restricted*" or "*presentation allowed*" depending on SIP Privacy. (See Table 6-14)	Network Option to either omit the parameter or derive from the SIP From (Note 1) (See Table 6-15)	APRI = "*presentation restricted*" or "*presentation allowed*" depending on SIP Privacy. (See Table 6-15)

NOTE 1 – This mapping effectively gives the equivalent of Special Arrangement to all SIP UAC with access to the I-IWU.

NOTE 2 – 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.

NOTE 3 – The SIP From header field may contain an "Anonymous URI". An "Anonymous URI" includes information that does not point to the calling party. [IETF RFC 3261] recommends that the display-name component contain "*Anonymous*". [IETF RFC 3323] recommends that the Anonymous URI itself have the value "*anonymous@anonymous.invalid*".

NOTE 4 – A national option exists to set the APRI to "*Address not available"*.

6.1.3.6.1 Calling Party Number

Table 6-13 – Setting of the network-provided BICC/ISUP Calling Party Number parameter with a CLI (network option)

BICC/ISUP CgPN parameter field	Value
Screening Indicator	*"network provided"*
Number Incomplete Indicator	*"complete"*
Numbering Plan Indicator	*"ISDN/Telephony (E.164)"*
Address Presentation Restricted Indicator	*"Presentation allowed/restricted*" (see Table 6-12)
Nature of Address Indicator	If next BICC/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 6-14 – Mapping of P-Asserted-Identity and Privacy header fields to the BICC/ISUP Calling Party Number parameter

Source SIP header field and component	Source component value	Calling Party Number parameter field	Derived value of parameter field
–	–	Number Incomplete Indicator	"*complete*"

Table 6-14 – Mapping of P-Asserted-Identity and Privacy header fields to the BICC/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 BICC/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.

NOTE 2 – It is possible to receive two priv-values, one of which is "*none*", the other "*id*". In this case, APRI shall be set to "*presentation restricted*".

6.1.3.6.2 Generic Number

Table 6-15 – Mapping of SIP From header field to BICC/ISUP Generic Number (*"additional calling party number"*) parameter (network option)

Source SIP header field and component	Source component value	Generic Number parameter field	Derived value of parameter field
–	–	Number Qualifier Indicator	"*additional calling party number*"
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 BICC/ISUP node is located in the same country, then set to *"national (significant) number"* else set to "*international number*"
–	–	Number Incomplete Indicator	"*complete*"
–	–	Numbering Plan Indicator	"*ISDN (Telephony) numbering plan (ITU-T E.164)*"
–	–	Address Presentation Restricted Indicator (APRI)	Use same setting as for calling party number.
–	–	Screening Indicator	"*user provided, not verified*"
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 number*" then set to CC + NDC + SN

6.1.3.7 User Service Information (Optional)

See clause 6.1.3.5.

6.1.3.8 Application Transport: BAT (BICC only)

See Annex A.

6.1.3.9 Hop Counter (Optional)

For Profile C (SIP-I), the I-IWU acting as an independent exchange shall perform the normal BICC/ISUP Hop Counter procedure using the Hop Counter taken from the encapsulated IAM if the Hop Counter parameter is available. The procedure applicable to profiles A and B shall also be used for Profile C, if no Hop Counter parameter is received in the encapsulated IAM and the succeeding network supports the Hop Counter procedure.

For profiles A and B, 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 6-16, where the factor is constructed according to the following principles:

a) 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.

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

Table 6-16 – 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.

6.1.3.10 Higher Layer Compatibility information element and Low Layer Compatibility information element within Access Transport parameter (optional), User Teleservice Information parameter

For profiles A and B and if the I-IWU supports the PSTN XML body as a network option and an INVITE containing a PSTN XML body is received, the PSTN XML elements indicated in Table 6-17 shall be mapped:

Table 6-17 – Mapping of PSTN XML elements into ISUP Parameters

INVITE →		IAM →	
PSTN XML	**Value**	**ISUP Parameter**	**Content**
HighLayerCompatibility		Access Transport parameter	High layer compatibility (Note 1)
LowLayerCompatibility			Low layer compatibility
ProgressIndicator			Progress indicator
HighLayerCompatibility		User Tele Service (Note 2)	High layer compatibility
NOTE 1 – If two High Layer Compatibility information elements are received, they shall be transferred in the same order as received in the PSTN XML body in the INVITE message. NOTE 2 – The support of the ISUP User Tele Service parameter is a network option.			

A ProgressIndicator with Progress Description value 6 shall not be included into the ISUP ATP, and is mapped instead to the Forward Call Indicators parameter according to Table 6-6.

6.1.3.11 Progress Indicator

For profiles A and B: If the I-IWU supports the PSTN XML body as a network option and an INVITE containing a PSTN XML body is received, an available "ProgressIndicator" element in the PSTN XML body shall be mapped into a Progress Indicator in the Access Transport parameter of the sent IAM as shown in Table 6-18.

Table 6-18 – Contents of the Access Transport parameter

INVITE →	IAM→
PSTN XML	**Access Transport parameter**
ProgressIndicator	Progress Indicator (Note)
NOTE – A ProgressIndicator with Progress Description value 6 shall not be included into the ISUP ATP, and is mapped instead to Forward call indicators parameter according to Table 6-6.	

6.1.3.12 Location Number

Location Number is defined in clause 3.30 of [ITU-T Q.763].

For profiles A and B: If the received INVITE message contains a P-Access-Network-Info header and the P-Access-Network-Info header contains an access-type with the value "GSTN" the IWU shall:

1) If an access-info field with the value "gstn-location" is present and the value "operator-specific-GI" is not present, include an ISUP Location Number parameter in the outgoing IAM set as shown in Table 6-19.

As a network option:

1) If an access-info field with the value "operator-specific-GI" is present and the value "gstn-location" is not present, include an ISUP Location Number parameter in the outgoing IAM set as shown in Table 6-21.

2) If both values are present, include an ISUP Location Number parameter in the outgoing IAM set as shown in Table 6-19 or Table 6-21, based on local policy.

For profiles A and B: As a network option, if the P-Access-Network-Info header contains an access-type different from "GSTN" and an access-info parameter with the value "operator-specific-GI", the I-IWU shall include an ISUP Location Number parameter in the outgoing IAM as shown in Table 6-21.

Table 6-19 – Contents of the location number parameter

INVITE →	IAM→
	Location Number parameter
P-Access-Network-Info with access-type="GSTN" and gstn-location parameter	Binary value derived from the gstn-location parameter of the P-Access-Network-Info
NOTE – The binary value shall be obtained by removing the quotes and converting each pair of consecutive character strings into an octet that has the equivalent hexadecimal representation.	

Table 6-20 – Mapping of P-Access-Network-Info to Location Number

INVITE→	IAM→
P-Access-Network-Info	**Location Number**
access type="GSTN" gstn-location=value	**Address Signal:** Copied from octet 3 to n of the binary representation of the gstn-location field

Table 6-20 – Mapping of P-Access-Network-Info to Location Number

INVITE→	IAM→
	Odd/even Indicator: Copied from bit 8 octet 1 of the binary representation of the gstn-location field
	Nature of Address Indicator: Copied from bit 7 to 1 of octet 1 of the binary representation of the gstn-location field
	Internal Network Number Indicator: Copied from bit 8 of octet 2 of the binary representation of the gstn-location field
	Numbering Plan Indicator: Copied from bit 7 to 5 of octet 2 of the binary representation of the gstn-location field
	Address Presentation Restricted Indicator: If the SIP Privacy header field = header, APRI is set to "01 (presentation restricted)" otherwise APRI is copied from bit 4 and 3 of octet 2 of the binary representation of the gstn-location field
	Screening Indicator: If the np parameter is present in the P-Access-Network-Info header field, set to "11 (network provided)", otherwise set from bit 2 and 1 of octet 2 of the binary representation of the gstn-location field

Table 6-21 – Mapping of P-Access-Network-Info operator-specific-GI to Location Number

INVITE →	IAM→
	Location Number parameter
P-Access-Network-Info with access-info="operator-specific-GI" and operator-specific-GI parameter	
P-Access-Network-Info	**Location Number**
access-info="operator-specific-GI" operator-specific-GI=value	**Address Signal:** Copy the digits set by 2 digits per byte. In case of an odd number of address signals, the filler code 0000 is inserted after the last address signal.
	Odd/even Indicator: Set as required
	Nature of Address Indicator: Set from an operator-configured value
	Internal Network Number Indicator: 1 routing to internal network number not allowed
	Numbering Plan Indicator: Set from an operator-configured value

Table 6-21 – Mapping of P-Access-Network-Info operator-specific-GI to Location Number

INVITE →	IAM→
	Address Presentation Restricted Indicator: If the SIP Privacy header field = header, APRI is set to "01 (presentation restricted)"
	Screening Indicator: If the np parameter is present in the P-Access-Network-Info header field, set to "11 (network provided)"
NOTE – This mapping is only possible if the operator-specific-GI field of the P-Access-Network-Info header filed contains a sequence of digits.	

6.1.3.13 UID Capability Indicators (National option)

For profiles A and B: UID Capability Indicators parameter is defined in [ITU-T Q.763].

When a both-way media path is available, then it is possible to support user interactive dialogues prior to answer through the use of the P-Early-Media header field authorizing both-way media.

If the received initial INVITE request contains a P-Early-Media header field with the "supported" value and either the SDP in the INVITE request allows a send/receive media path or the INVITE request does not contain an SDP body then the UID Capability Indicators parameter in the outgoing IAM message may be set as shown in Table 6-22.

Table 6-22 – Contents of the UID Capability Indicators parameter

INVITE →	IAM→	
P-Early-Media with "supported" and either SDP with "sendrecv" (explicit or implied) or no SDP present	**UID Capability Indicators parameter**	
	Through-connection Indicator	1 "through-connection modification possible"
	T9 Timer Indicator	0 "no indication"

6.1.3.14 Coding of the IAM when Number Portability is supported

For profiles A and B: This clause describes optional coding procedures when Number Portability is supported.

6.1.3.14.1 Coding of the IAM when a Number Portability Routing Number is available

For profiles A and B: [ITU-T Q.769.1] describes three possible addressing methods for signalling of the Called Party E.164 address and Number Portability Routing Number ([ITU-T Q.769.1] uses the terms directory number and network routing number respectively). The choice of these methods is based on network operator and national requirements.

The following clauses describe how the IAM is populated, based on these methods, when a Number Portability Routing Number is available in the Request URI in the form of a Tel URI "rn=" parameter.

When the optional Number Portability Routing Number is available and supported, these procedures take precedence over procedures for coding of the Called Party Number described in clause 6.1.3.1.

If the Number Portability Database Dip Indicator is present within the Request-URI the procedures described in clause 6.1.3.14.5 apply. When a Number Portability Routing Number is not available, the Called Party Number parameter is populated as described in clause 6.1.3.1.

6.1.3.14.2 Separate Directory Number Addressing Method

Table 6-23 – Coding of the Called Party Number and Called Directory Number with Number Portability Separate Directory Number Addressing Method

INVITE→	**IAM→**	
Request-URI	**Called Party Number**	**Called Directory Number**
Called Party E.164 address (format +CC NDC SN) (e.g., as User info in SIP URI with user=phone, or as tel URL) plus Number Portability Routing Number (format +CC NDC SN) (e.g., as Tel URI rn= parameter) plus Number Portability Database Dip Indicator as defined in [IETF RFC 4694] (e.g., as Tel URI npdi parameter)	**Address Signal:** Analyse the information contained in received Number Portability Routing Number. If the Number Portability Routing number contains an E.164 address, then remove "+CC" and use the remaining digits to fill the Address signal. Otherwise, use the digits of the Number Portability Routing number to fill the Address signal. (Note)	**Address Signal:** Remove "+CC" from the Called Party E.164 address and use the remaining digits to fill the Address signals.
	Odd/even Indicator: set as required	**Odd/even Indicator**: set as required
	Nature of Address Indicator: Set Nature of Address Indicator to "Network routing number in national (significant) number format". "National (significant) number" and "Network routing number in network specific number format" may alternately be chosen as described in [ITU-T Q.769.1]	**Nature of Address Indicator**: Set Nature of Address Indicator to "National (significant) number".
	Internal Network Number Indicator: 1 routing to internal network number not allowed	**Internal Network Number Indicator**: 1 routing to internal network number not allowed
	Numbering Plan Indicator: 001 ISDN (Telephony) numbering plan [ITU-T E.164]	**Numbering Plan Indicator**: 001 ISDN (Telephony) numbering plan [ITU-T E.164]
NOTE – If PSTN XML and ISUP Sending Terminated (ST) signal are supported as a network option, then the PSTN XML sendingCompleteIndication, if present, is mapped to the sending terminated digit (hexadecimal digit F) in the address signals field of the Called Party Number parameter.		

6.1.3.14.3 Concatenated Addressing Method

Table 6-24 – Coding of the Called Party Number with Number Portability Concatenated Addressing Method

INVITE→	IAM→
Request-URI	**Called Party Number**
Called Party E.164 address (format +CC NDC SN) (e.g., as User info in SIP URI with user=phone, or as tel URL) plus Number Portability Routing Number (format +CC NDC SN) (e.g., as Tel URI rn= parameter) plus Number Portability Database Dip Indicator as defined in [IETF RFC 4694] (e.g., as Tel URI npdi parameter)	**Address Signal:** Analyse the information contained in received Number Portability Routing Number. If the Number Portability Routing number contains an E.164 address, then remove "+CC" and use the remaining digits to fill the Address signal. Otherwise, use the digits of the Number Portability Routing number to fill the Address signal. Remove the "+CC" from the Called Party E.164 address and append the remaining digits to the Address signal. (Note)
	Odd/even Indicator: set as required
	Nature of Address Indicator: set Nature of Address Indicator to "Network routing number concatenated with called directory number" or "National (significant) number" as described in [ITU-T Q.769.1]
	Internal Network Number Indicator: 1 routing to internal network number not allowed
	Numbering Plan Indicator: 001 ISDN (Telephony) numbering plan [ITU-T E.164]
NOTE – If PSTN XML and ISUP Sending Terminated (ST) signal are supported as a network option, then the PSTN XML sendingCompleteIndication, if present, is mapped to the sending terminated digit (hexadecimal digit F) in the address signals field of the Called Party Number parameter.	

6.1.3.14.4 Separate Network Routing Number Addressing Method

Table 6-25 – Coding of the Network Routing Number and Called Party Number with Number Portability Separate Network Routing Number Addressing Method

INVITE→	IAM→	
Request-URI	**Network Routing Number**	**Called Party Number**
Called Party E.164 address (format +CC NDC SN) (e.g., as User info in SIP URI with user=phone, or as tel URL) plus	**Address Signal**: Analyse the information contained in received Number Portability Routing Number. If the Number Portability Routing number contains an E.164 address, then remove "+CC" and use the remaining digits to fill the Address signal. Otherwise, use the digits of the Number Portability Routing number to fill the Address signal.	**Address Signal**: Remove "+CC" from the Called Party E.164 address and use the remaining digits to fill the Address signals. (Note)
	Odd/even Indicator: set as required	**Odd/even Indicator**: set as required

Table 6-25 – Coding of the Network Routing Number and Called Party Number with Number Portability Separate Network Routing Number Addressing Method

INVITE→	IAM→	
Request-URI	**Network Routing Number**	**Called Party Number**
Number Portability Routing Number (format +CC NDC SN) (e.g., as Tel URI rn= parameter) plus Number Portability Database Dip Indicator as defined in [IETF RFC 4694] (e.g., as Tel URI npdi parameter)	**Nature of Address Indicator**: Set Nature of Address Indicator to "Network routing number in national (significant) number format". "Network routing number in network specific number format" may alternately be chosen as described in [ITU-T Q.769.1]	**Nature of Address Indicator**: Set Nature of Address Indicator to "National (significant) number".
		Internal Network Number Indicator: 1 routing to internal network number not allowed
	Numbering Plan Indicator: 001 ISDN (Telephony) numbering plan [ITU-T E.164]	**Numbering Plan Indicator**: 001 ISDN (Telephony) numbering plan [ITU-T E.164]
NOTE – If PSTN XML and ISUP Sending Terminated (ST) signal are supported as a network option, then the PSTN XML sendingCompleteIndication, if present, is mapped to the sending terminated digit (hexadecimal digit F) in the address signals field of the Called Party Number parameter.		

6.1.3.14.5 Number Portability Forward Information

For profiles A and B: Network operator or national policy may allow the forward transfer of Number Portability status information, as described in [ITU-T Q.769.1]. In this case, the following coding applies.

Table 6-26 – Coding of the Number Portability Forward Information

INVITE→	IAM→
Request-URI	**Number Portability Forward Information**
Called Party E.164 address (format +CC NDC SN) (e.g., as User info in SIP URI with user=phone, or as tel URL) plus Number Portability Routing Number (format +CC NDC SN) (e.g., as Tel URI rn= parameter) plus Number Portability Database Dip Indicator as defined in [IETF RFC 4694] (e.g., as Tel URI npdi parameter)	If the Number Portability Database Dip Indicator is present, and there is no Number Portability Routing Number, set to "number portability query done for called number, non-ported called subscriber". If the Number Portability Database Dip Indicator is present, and a Number Portability Routing Number is present, set to "number portability query done for called number, ported called subscriber". If there is no Number Portability Database Dip Indicator, set to "number portability query not done for called number"

6.1.3.16 Coding of the IAM for Carrier Routing

For profiles A and B: This clause describes optional coding procedures for carrier-based routing. The interworking of the Carrier Identification Code (CIC) parameter is defined.

6.1.3.16.1 Coding of the IAM when a Carrier Identification Code (CIC) is present

For profiles A and B: The procedures followed in clause 6.1.3.1 apply with the following addition.

Based on network configuration, if the tel-URI parameter in a tel-URI or the userinfo part of a SIP URI with user=phone in the Request-URI of an initial INVITE request, contains a "cic=" parameter, as defined in [IETF RFC 4694], the I-IWU may extract the carrier identification code from the "cic=" field for routing the call. If the outgoing IAM message contains the Transit Network Selection (TNS) parameter, as defined in [ITU-T Q.763], based on network configuration the TNS may be populated using the carrier identification code from the "cic=" field. The format of the "cic" parameter (e.g., global-cic and local-cic) shall be compliant to [IETF RFC 4694].

6.2 Receipt of subsequent INVITE

This clause applies when the overlap operation is supported across the I-IWU. Other configurations are handled by the SIP or BICC/ISUP state machines operating separately.

If the I-IWU receives an INVITE with the same Call-ID and From tag as a previous INVITE which was associated with a BICC/ISUP call/bearer control instance currently existing on the BICC/ISUP side, then:

a) If the number of digits in the Request-URI is greater than the number of digits already accumulated for the call, the I-IWU shall generate a SAM and pass it to outgoing BICC/ISUP procedures. The SAM shall contain in its Subsequent Number parameter only the additional digits received in this Request-URI compared with the digits already accumulated for the call. For Profile C (SIP-I), any encapsulated IAM is ignored during this process and is not used. Any earlier INVITE is replied to with a 484 Address Incomplete response if this has not already been done.

b) If the number of digits in the Request-URI is equal to or less than the number of digits already accumulated for the call, then the I-IWU shall immediately send a 484 Address Incomplete response for this INVITE. In this case, no SAM is sent to BICC/ISUP procedures.

6.2.1 Independence of session negotiation and receipt of address information

As a general principle, the overlap procedures allow for session negotiation (and, in particular, the negotiation and confirmation of preconditions) to continue independently of the receipt of address information. On sending of a 484 Address Incomplete message for an INVITE transaction, the I-IWU considers any offer-answer exchange initiated by the INVITE to be terminated. The new INVITE initiates a new offer-answer exchange. However, if resources have already been reserved and they can be reused within the new offer-answer exchange, the precondition signalling shall reflect the current status of the affected preconditions.

6.3 Sending of COT

When the I-IWU determines that all the preconditions on the incoming SIP side have been met and any continuity procedures on the outgoing BICC/ISUP side have been successfully completed, the I-IWU shall send the COT message coded as follows:

1) If the subsequent network is a BICC network, the Continuity Indicator in the COT message shall be set to "*Continuity*".

2) If the subsequent network is an ISUP network, the Continuity Indicator in the COT message shall be set to "*Continuity check successful*".

6.4 Receipt of Connect message (CON)

Table 6-27 indicates the mapping of the Connect message.

Table 6-27 – Message sent to SIP upon receipt of CON

← Message sent to SIP	← Message Received from BICC/ISUP
200 OK INVITE	CON

When Profile C (SIP-I) is applicable, the Connect message is encapsulated in a 200 OK INVITE final response.

6.4.1 Mapping of PSTN XML elements

For profiles A and B and if the I-IWU supports the PSTN XML body as a network option, the I-IWU shall map the Access Transport parameter received in the CON into PSTN XML elements as shown in Table 6-28 and include this PSTN XML body in the 200 OK (INVITE).

On receipt of a CON message containing the ATP including the Bearer Capability set to "unrestricted digital information with tones/announcement" without the TMU parameter, the 200 OK message shall contain the PSTN XML Bearer Capability "unrestricted digital information with tones/announcement".

For profiles A and B: If the I-IWU supports the PSTN XML body as a network option, the I-IWU shall map an available BCI element in the CON into a ProgressIndicator in the PSTN XML body as shown in Table 6-29; the I-IWU shall include both a ProgressIndicator mapped from a possibly received Progress indicator element in the Access Transport parameter and ProgressIndicators derived according to Table 6-29 in the PSTN XML.

NOTE 1 – The order of ProgressIndicators within the same PSTN XML body is irrelevant.

Table 6-28 – Mapping of ISUP ATP parameter into PSTN XML elements

←200 OK	← CON	
PSTN XML	**ISUP Parameter**	**Content**
ProgressIndicator	Access Transport parameter	Progress indicator
HighLayerCompatibility (Note 2)		High layer compatibility
LowLayerCompatibility (Note 2)		Low layer compatibility
BearerCapability (Note 1, Note 2)		Bearer Capability
BearerCapability (Note 1, Note 2)	Transmission Medium Used parameter (Note 1)	
NOTE 1 – See clause 6.5.4 Transmission Medium Used parameter (TMU). NOTE 2 – The I-IWU shall only provide this IE if it interworks media encoded in any of the formats in Table 6-11 (ITU-T G.711, Clearmode or t38) without transcoding. If both TMU and a BC in the ATP have been received, the BC in the ATP shall be mapped.		

Table 6-29 – Mapping of ISUP BCI and optional BCI parameters into PSTN XML ProgressIndicator

← 200 OK	← CON
PSTN XML body with ProgressIndicator with "Progress Description" value No (Value of PI) (NOTE)	**Content** **Backward Call Indicators parameter** **Optional Backward Call Indicators parameter**
No. 1 ("Call is not end-to-end ISDN: further call progress information may be available in-band")	Backward Call Indicators parameter ISDN User Part Indicator 0 "ISDN User Part not used all the way"
No. 2 ("Destination address is non-ISDN")	Backward Call Indicators parameter ISDN User Part Indicator 1 "ISDN User Part used all the way" ISDN Access Indicator 0 "Terminating access non-ISDN"
No. 7 ("Terminating access ISDN")	Backward Call Indicators parameter ISDN User Part Indicator 1 "ISDN User Part used all the way" ISDN access Indicator 1 "Terminating access ISDN"
No. 8 ("In-band information or an appropriate pattern is now available")	Optional Backward Call Indicators parameter In-band Information Indicator 1 "in-band information or an appropriate pattern is now available"
NOTE – The ProgressIndicator "Coding Standard" parameter shall be set to "00 (ITU-T standardized coding)". The default value for the ProgressIndicator "Location" parameter is "0011 (Transit Network)".	

For profiles A and B and if the I-IWU supports the PSTN XML body and receives a Transmission Medium Used (TMU) parameter,

NOTE 2 – The I-IWU will only receive a TMU parameter if it has applied the fallback related procedures in clause 6.1.3.5, including both a USI and TMR Prime parameter in the IAM, and fallback to the bearer capability identified in USI and TMR Prime occurred at the terminating side.

then the I-IWU shall:

– if a BC is not available in the ATP in the CON, map the TMU value (Speech or 3.1 kHz audio) to the PSTN XML BearerCapability element;

– if a BC is available in the ATP in the CON, include the received BC in the PSTN XML BearerCapability element;

– configure the MGW to use the second format in the m-line in the SDP that has been received in the INVITE as codec at the IMS termination; and

– send SDP selecting the second format in the m-line of the SDP that has been received in the INVITE as soon as allowed according to SIP rules.

For Profile B and if the I-IWU supports the PSTN XML body, if it has applied the fallback related procedures in clause 6.1.3.5, including both a TMR and TMR Prime in the IAM, and did not receive TMU in the CON, or any previous ISUP message,

NOTE 3 – Fallback to the bearer capability identified in TMR did not occur at the terminating side.

then the I-IWU shall:

– configure the MGW to use the first format in the m-line in the SDP offer that has been received in the INVITE as codec at the network termination; and

– send SDP selecting the first format in the m-line in the SDP offer at the first possibility according to SIP rules.

6.5 Receipt of ACM

Table 6-30 provides a summary of how the Address Complete Message (ACM) 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 Backward Call Indicators parameter of the ACM.

1) If the BCI (Called Party's Status Indicator) is set to "*subscriber free*" then:

   – In the case of Profile A or Profile B, the 180 Ringing SIP response is sent from the I-IWU; the handling of P-Early-Media header and PSTN XML is described in clause 6.5.1.

   – In the case of Profile C (SIP-I), a 180 Ringing SIP response is sent from the I-IWU. The ACM is encapsulated within this response.

   The P-Early-Media header is supported: If the Transmission Medium Requirement parameter in the sent IAM was set to "speech", "3.1 kHz audio" or "64 kBit/s preferred" and if a P-Early-Media header was included in the received INVITE, the 180 Ringing shall contain a P-Early-Media header field authorizing early media, except when

   – the I-IWU has already sent a reliable provisional response (see [IETF RFC 3262]) including a P-Early-Media header, as defined in [IETF RFC 5009], and

   – the most recently sent P-Early-Media header field authorized early media.

   NOTE – If the I-IWU signals the P-Early-Media header field authorizing early media, then the IMS can expect tones or announcements to the calling party to flow from the CS network via an MGW controlled by the I-IWU. In particular, once the I-IWU sends the 180 Ringing response, ringback is expected in media from the CS network.

   The interworking of the PSTN XML element as a network option is described in clause 6.5.1, Tables 6-31 and 6-32.

   As a network option: if the P-Early-Media header was not present in the received INVITE request, a P-Early-Media header the I-IWU may include a P-Early-Media header in the 180 Ringing response authorizing backward early media.

   As a network option: the I-IWU may insert a Call-Info header field or an Alert-Info header field as described in [IETF RFC 3261] in the 180 Ringing response. The URL of these two headers provides early media instead of the ringback tone from the PSTN.

2) BCI (Called Party's Status Indicator) = "*no indication*" or any value other than "*subscriber-free*": If this parameter is not set to "*subscriber-free*" then one of the options below applies:

   – the ACM is not interworked;

     NOTE 1 – A backward path is available as soon as the IAM is sent and an appropriate SDP is received from the calling end.

   – in the case of profiles A and B if the P-Early-Media header is supported: If the Transmission Medium Requirement parameter in the sent IAM was set to "speech", "3.1 kHz audio" or "64 kBit/s preferred" and if a P-Early-Media header was included in the received INVITE and a P-Early-Media header was not sent in a previous provisional response, a 183 Session Progress is sent and the P-Early-Media header is present authorizing early media. The interworking of the PSTN XML element as a network option is described in clause 6.5.1, and Tables 6-31 and 6-33;

– in the case of Profile C (SIP-I), a 183 Session Progress response is sent from the I-IWU. (see Table 6-30). The ACM is encapsulated within this response.

NOTE 2 – The ACM with Cause parameter is not interworked (except for encapsulation in Profile C (SIP-I) operation). Protection against indefinite prolongation of the call is provided by T9 and other timers.

Table 6-30 – Message sent to SIP upon receipt of ACM

← Message sent to SIP	← ACM
	Backward Call Indicators parameter Called Party's Status Iindicator
183 Session Progress in case of Profile C For Profile B see clause 6.5.1	00 *"No indication"*
180 Ringing	01 *"Subscriber free"*

6.5.1 Interworking of P-Early-Media header and PSTN XML element.

For profiles A and B: As a network option if the I-IWU supports the PSTN XML schema the I-IWU shall map the Access ISUP Transport parameter received in the ACM into PSTN XML elements as described in Table 6-31 into the 183 Session Progress or 180 Ringing.

Table 6-31 – Interworking of ACM Access Transport parameter into PSTN XML element in the 18x

<table>
<tr><th>← 18x</th><th colspan="2">← ACM</th></tr>
<tr><th>PSTN XML</th><th>ISUP parameter</th><th>Content</th></tr>
<tr><td>ProgressIndicator</td><td rowspan="4">Access Transport parameter</td><td>Progress indicator</td></tr>
<tr><td>HighLayerCompatibility (Note 2)</td><td>High layer compatibility</td></tr>
<tr><td>LowLayerCompatibility (Note 2)</td><td>Low layer compatibility</td></tr>
<tr><td>BearerCapability (Note 1, Note 2)</td><td>Bearer Capability</td></tr>
<tr><td>BearerCapability (Note 1, Note 2)</td><td>Transmission Medium Used parameter (Note 1)</td><td></td></tr>
<tr><td colspan="3">NOTE 1 – See clause 6.5.4 Transmission Medium Used parameter (TMU).
NOTE 2 – The I-IWU shall only provide this IE if it interworks media encoded in any of the formats in Table 6-11 (ITU-T G.711, Clearmode or t38) without transcoding. If both TMU and a BC in the ATP have been received, the BC in the ATP shall be mapped.</td></tr>
</table>

6.5.1.1 Mapping of ACM into 180 Ringing

The ACM with Backward Call Indicator Called Party's Status Indicator set to "subscriber free" received a 180 Ringing is sent.

For profiles A and B: As a network option if the I-IWU supports the PSTN XML schema and the P-Early-Media header and a P-Early-Media header was received in the initial INVITE request: The I-IWU shall include the P-Early-Media header authorizing early media if a P-Early-Media header was not sent in a previous reliable provisional response and shall map the Backward Call Indicator or the optional Backward Call Indicator in the received ACM (if present) into a 180 Ringing PSTN XML ProgressIndicator element or an additional PSTN XML ProgressIndicator element if an ATP Progress Indicator IE was received as described in Table 6-32.

Table 6-32 – Interworking of ACM parameter into 180 Ringing

← 180 Ringing	← ACM
PSTN XML element "ProgressIndicator" "ProgressDescription" value	**Backward Call Indicator** **Optional Backward Call Indicator**
Value 8 ("In-band information or an appropriate pattern is now available")	Optional Backward Call Indicators parameter In-band Information Indicator "in-band information or an appropriate pattern is now available"
Value 1 ("Call is not end-to-end ISDN: further call progress information may be available in-band")	Backward Call Indicators parameter ISDN User Part Indicator 0 "ISDN User Part not used all the way"
Value 2 ("Destination address is non-ISDN")	Backward Call Indicators parameter ISDN User Part Indicator 1 "ISDN User Part used all the way" ISDN Access Indicator 0 "Terminating access non-ISDN"
Value 7 ("Terminating access ISDN")	Backward Call Indicators parameter Interworking Indicator 0 no interworking encountered (Signalling System No. 7 all the way) ISDN User Part Indicator 1 "ISDN User Part used all the way" ISDN Access Indicator 1 "Terminating access ISDN"
NOTE – The ProgressIndicator "Coding Standard" parameter shall be set to "00 (ITU-T standardized coding)". The default value for the ProgressIndicator "Location" parameter is "0011 (Transit Network)".	

NOTE – The order of ProgressIndicators elements within the same PSTN XML element is irrelevant.

6.5.1.2 Mapping of ACM into 183 Session Progress

The ACM with Backward Call Indicator Called Party's Status Indicator set to "no indication" received a 183 Session Progress is sent under the conditions described in this clause.

For profiles A and B: As a network option if the I-IWU supports the PSTN XML schema and the P-Early-Media header and a P-Early-Media header was received in the initial INVITE request: The I-IWU shall include the P-Early-Media header authorizing early if a P-Early-Media header was not sent in a previous provisional response and shall map the Backward Call Indicator or the optional Backward Call Indicator or UID Action Indicators parameter in the received ACM (if present) into a 183 Session Progress PSTN XML ProgressIndicator element or an additional PSTN XML ProgressIndicator element if an ATP Progress Indicator IE was received as described in Table 6-32.

Based on local configuration, the I-IWU may also send a 183 Session Progress response with a P-Early-Media header field authorizing early media if it receives an ACM with other parameters than those described in Table 6-33.

Table 6-33 – Interworking of ACM parameter into 183 Session Progress

← 183 Session Progress	← ACM
PSTN XML element "ProgressIndicator" "ProgressDescription" value	**Backward Call Indicator** **Optional Backward Call Indicator** **UID Action Indicators parameter**
Value 8 ("In-band information or an appropriate pattern is now available") P-Early-Media header field authorizing backward early media, if not already sent (NOTE 2)	Optional Backward Call Indicators parameter In-band Information Indicator "in-band information or an appropriate pattern is now available"
Value 1 ("Call is not end-to-end ISDN: further call progress information may be available in-band")	Backward Call Indicators parameter ISDN User Part Indicator 0 "ISDN User Part not used all the way"
Value 2 ("Destination address is non-ISDN")	Backward Call Indicators parameter ISDN User Part Indicator 1 "ISDN User Part used all the way" ISDN Access Indicator 0 "Terminating access non-ISDN"
Value 7 ("Terminating access ISDN")	Backward Call Indicators parameter Interworking Indicator 0 no interworking encountered (Signalling System No. 7 all the way) ISDN User Part Indicator 1 "ISDN User Part used all the way" ISDN Access Indicator 1 "Terminating access ISDN"
P-Early-Media header field authorizing backward early media, if not already sent (NOTE 2)	UID Action Indicators parameter Through-connection instruction indicator 1 "through-connect in both directions"
NOTE 1 – The ProgressIndicator "Coding Standard" parameter shall be set to "00 (ITU-T standardized coding)". The default value for the ProgressIndicator "Location" parameter is "0011 (Transit Network)". NOTE 2 – Setting of the P-Early-Media header field based on the UID Action Indicators parameter value of "through-connect in both directions" takes precedence over a setting based on the Optional Backward Call Indicators parameter value of "In-band info or an appropriate pattern is now available".	

NOTE – As a network option the I-IWU can also map ACM into 183 in other cases than those described in Table 6-33.

6.5.3 Fallback occurs in the I-IWU

For profiles A and B: As a network option if the I-IWU supports the PSTN XML schema: If the I-IWU receives two PSTN XML Bearer Capability elements within the initial INVITE Request, however, the I-IWU has the knowledge that the succeeding ISUP node does not have the capability to support TMR "64 kBit/s preferred", the I-IWU shall send a 18x provisional response and the following PSTN XML elements are included:

– a BearerCapability element copied from the first received PSTN XML BearerCapability in the initial INVITE request;

– if two HighLayerCompatibility elements were present in the PSTN XML body in the received INVITE, then one HighLayerCompatibility element shall be copied from the first HighLayerCompatibility element;

– a ProgressIndicator element with "Progress Description" value 5 ("Interworking has occurred and has resulted in a telecommunication service change"),"Coding Standard" value "00 (ITU-T standardized coding)", and default value "0011 (Transit Network)" for the "Location" parameter.

Select the second stated codec in the SDP offer in the initial INVITE request and instruct the Media Gateway to use the selected codec.

6.5.4 Fallback in the succeeding network

For profiles A and B: As a network option if the I-IWU supports the PSTN XML schema: If the I-IWU receives two PSTN XML BearerCapability elements within the initial INVITE Request and the procedure in clause 6.1.3.5 is applied, on receipt of an ACM containing a Transmission Medium Used parameter and optional an Access Transport Parameter, the I-IWU shall send a 180 Ringing or 183 Session Progress with the following PSTN XML elements present:

– If no Bearer Capability IE is present in the ACM ATP, copy the first received PSTN XML BearerCapability element, the InformationTransferCabability equal to the TMU value from the initial INVITE request and include it in the sent 18x response.

– If a Bearer Capability IE is present in the ACM ATP then map this parameter into the sent 18x response.

Select the second stated codec in the SDP offer of the initial INVITE request and:

– Send this codec as the first stated codec in the 'm' line of the SDP answer in the 18x response.

– Instruct the Media Gateway to use the selected codec.

Table 6-34 – Sending of Fallback indication

← 180 Ringing or 183 Session Progress	←ACM
PSTN XML BearerCapability = "Speech"	TMU "Speech" ATP No BC
PSTN XML BearerCapability = "3.1 kHz audio"	TMU "3.1 kHz audio" ATP No BC
PSTN XML BearerCapability received in the ATP ("speech" or "3.1 kHz audio")	TMU "Speech or 3.1 kHz audio" ATP BC ("speech" or "3.1 kHz audio")

6.5.5 Sending of 181 Call is being forwarded

For profiles A and B: As a network option if the I-IWU supports the P-Early-Media header and a P-Early was received in the initial INVITE request when an ACM is received and a Redirection number, a Call Diversion Information parameter and the Notification subscription options are not set to 'presentation not allowed', and an optional Backward Call Indicators parameter, indicate that in-band information is available, a 181 Call is being forwarded is sent with a present P-Earl-Media header. The mapping of the Call diversion parameters are described in clause B.6.

6.6 Receipt of CPG

For profiles A and B as a network option if the I-IWU supports the P-Early-Media header and a P-Early-Media header was received in the initial INVITE request and the PSTN XML schema the interworking is described in clause 6.6.1 otherwise the CPG is not interworked.

For Profile C (SIP-I), 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 6-35. This response shall encapsulate the CPG message.

Table 6-35 – 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 Profile C (SIP-I) otherwise not interworked. For Profile B the interworking is described in sub clause 6.6.1	000 0010 (*"progress"*) or 000 0011 (*"in-band information or an appropriate pattern is now available"*)

6.6.1 Interworking of CPG

For profiles A and B: As a network option if the I-IWU supports the P-Early-Media header and a P-Early was received in the initial INVITE request: If the Transmission Medium Requirement parameter in the sent IAM was set to "speech", "3.1 kHz audio" or "64 kbit/s preferred" and a CPG message is received the Event Indicator is set to "ALERTING" a 180 Ringing is sent and the P-Early-Media header shall be included authorizing early media if a P-Early-Media header was not sent in a previous reliable provisional response. The most recently sent P-Early-Media header field authorized backward early media.

As a network option: if the P-Early-Media header was not present in the received INVITE request, the I-IWU may include a P-Early-Media header in the 180 Ringing response authorizing backward early media.

As a network option: the I-IWU may insert a Call-Info header field or an Alert-Info header field as described in [IETF RFC 3261] in the 180 Ringing response. The URL of these two headers provides early media instead of the ringback tone from the PSTN.

If an Access Transport parameter is present in the received CPG, the mapping is described in Table 6-36.

Table 6-36 – Interworking of CPG Access Transport parameter into PSTN XML element in the 18x

<table>
<tr><th>← 18x</th><th colspan="2">← CPG</th></tr>
<tr><th>PSTN XML</th><th>ISUP parameter</th><th>Content</th></tr>
<tr><td>ProgressIndicator</td><td rowspan="4">Access Transport parameter</td><td>Progress indicator</td></tr>
<tr><td>HighLayerCompatibility (Note 2)</td><td>High layer compatibility</td></tr>
<tr><td>LowLayerCompatibility (Note 2)</td><td>Low layer compatibility</td></tr>
<tr><td>BearerCapability (Note 1, Note 2)</td><td>Bearer Capability</td></tr>
<tr><td>BearerCapability (Note 1, Note 2)</td><td>Transmission Medium Used parameter (Note 1)</td><td></td></tr>
<tr><td colspan="3">NOTE 1 – See clause 6.5.4 Transmission Medium Used parameter (TMU).</td></tr>
</table>

Table 6-36 – Interworking of CPG Access Transport parameter into PSTN XML element in the 18x

← 18x	← CPG	
PSTN XML	**ISUP parameter**	**Content**
NOTE 2 – The I-IWU shall only provide this IE if it interworks media encoded in any of the formats in Table 6-11 (ITU-T G.711, Clearmode or t38) without transcoding. If both TMU and a BC in the ATP have been received, the BC in the ATP shall be mapped.		

6.6.1.1 Mapping of CPG into 180 Ringing

CPG with Event information set to "ALERTING" received a 180 Ringing is sent.

For profiles A and B: As a network option if the I-IWU supports the PSTN XML schema and the P-Early-Media header and a P-Early was received in the initial INVITE request: The I-IWU shall include the P-Early-Media header authorizing early media if a P-Early-Media header was not sent in a previous reliable provisional response and shall map the Backward Call Indicator or the optional Backward Call Indicator in the received CPG (if present) into a 180 Ringing PSTN XML ProgressIndicator element or an additional PSTN XML ProgressIndicator element if an ATP Progress Indicator IE was received as described in Table 6-37.

Table 6-37 – Interworking of CPG parameter into 180 Ringing

← 180 Ringing	← CPG
PSTN XML element "ProgressIndicator" "ProgressDescription" value	**Backward Call Indicator** **Optional Backward Call Indicator**
Value 8 ("In-band information or an appropriate pattern is now available")	Optional Backward Call Indicators parameter In-band Information Indicator "in-band information or an appropriate pattern is now available"
Value 1 ("Call is not end-to-end ISDN: further call progress information may be available in-band")	Backward Call Indicators parameter ISDN User Part Indicator 0 "ISDN User Part not used all the way"
Value 2 ("Destination address is non-ISDN")	Backward Call Indicators parameter ISDN User Part Indicator 1 "ISDN User Part used all the way" ISDN Access Indicator 0 "Terminating access non-ISDN"
Value 7 ("Terminating access ISDN")	Backward Call Indicators parameter Interworking Indicator 0 no interworking encountered (Signalling System No. 7 all the way) ISDN User Part Indicator 1 "ISDN User Part used all the way" ISDN Access Indicator 1 "Terminating access ISDN"
NOTE – The ProgressIndicator "Coding Standard" parameter shall be set to "00 (ITU-T standardized coding)". The default value for the ProgressIndicator "Location" parameter is "0011 (Transit Network)".	

NOTE – The order of ProgressIndicators elements within the same PSTN XML element is irrelevant.

6.6.1.2 Mapping of CPG into 183 Session Progress

For profiles A and B: The CPG with Event Indicator set to "in-band information or an appropriate pattern is now available" is received. As a network option if the I-IWU supports the P-Early-Media header and a P-Early was received in the initial INVITE request: A 183 Session Progress is sent and the P-Early-Media header authorizing early media shall be included.

For profiles A and B: The CPG with Event Indicator set to "PROGRESS" received a 183 Session Progress is sent under the conditions described in this clause. As a network option if the I-IWU supports the PSTN XML schema and the P-Early-Media header and a P-Early was received in the initial INVITE request: The I-IWU shall include the P-Early-Media header authorizing early media if a P-Early-Media header was not sent in a previous provisional response and map the Event Indicator or Backward Call Indicator or the optional Backward Call Indicator or UID Action Indicators parameter in the received CPG (if present) into a 183 Session Progress PSTN XML ProgressIndicator element or an additional PSTN XML ProgressIndicator element if an ATP Progress Indicator IE was received, as described in Table 6-36.

For profiles A and B: Based on local configuration, the I-IWU may also send a 183 Session Progress response with a P-Early-Media header field authorizing early media if it receives a CPG with other parameters than those described in Table 6-38.

Table 6-38 – Interworking of CPG parameter into 183 Session Progress

← 183 Session Progress	← CPG
PSTN XML element "ProgressIndicator" "ProgressDescription" value	**Event Indicator** **Backward Call Indicator** **Optional Backward Call Indicator** **UID Action Indicators parameter**
Value 8 ("In-band information or an appropriate pattern is now available") (Note 3) P-Early-Media header field authorizing backward early media, if not already sent (Note 4)	**Event Indicator** 000 0010 (progress) **Optional Backward Call Indicators parameter** In-band Information Indicator 1 "In-band info or an appropriate pattern is now available
Value 1 ("Call is not end-to-end ISDN: further call progress information may be available in-band") (Note 3)	**Event Indicator** 000 0010 (progress) **Backward Call Indicators parameter** ISDN User Part Indicator 0 "ISDN User Part not used all the way"
Value 2 ("Destination address is non-ISDN") (Note 3)	**Backward Call Indicators parameter** ISDN User Part Indicator 1 "ISDN User Part used all the way" ISDN Access Indicator 0 "Terminating access non-ISDN"

Table 6-38 – Interworking of CPG parameter into 183 Session Progress

← 183 Session Progress	← CPG
Value 7 ("Terminating access ISDN") (Note 3)	**Backward Call Indicators parameter** Interworking Indicator 0 no interworking encountered (Signalling System No. 7 all the way) ISDN User Part Indicator 1 "ISDN User Part used all the way" ISDN Access Indicator 1 "Terminating access ISDN"
P-Early-Media header field authorizing backward early media, if not already sent (Note 4)	**UID Action Indicators parameter** Through-connection instruction indicator 1 "through-connect in both directions"
NOTE 1 – The mapping of the contents in the CPG message is only relevant if the information received in the message is different to earlier received information, e.g., in the ACM message or a CPG message received prior to this message. NOTE 2 – 183 Session Progress message including a P-Early-Media header authorizing early media may only be sent for a speech call. NOTE 3 – The ProgressIndicator "Coding Standard" parameter shall be set to "00 (ITU-T standardized coding)" The default value for the ProgressIndicator "Location" parameter is "0011 (Transit Network)". NOTE 4 – Setting of the P-Early-Media header field based on the UID Action Indicators parameter value of "through-connect in both directions" takes precedence over a setting based on the Optional Backward Call Indicators parameter value of "In-band info or an appropriate pattern is now available".	

6.6.2 Fallback in the succeeding network

For profiles A and B: As a network option if the I-IWU supports the PSTN XML schema: If the I-IWU receives two PSTN XML BearerCapability elements within the initial INVITE Request and the procedure in clause 6.1.3.5 is applied, on receipt of a CPG containing a Transmission Medium Used parameter and optional an Access Transport parameter, the I-IWU shall send a 180 Ringing or 183 Session Progress with the following PSTN XML elements present:

– If no Bearer Capability IE is present in the CPG ATP, copy the first received PSTN XML BearerCapability element, the InformationTransferCabability equal to the TMU value from the initial INVITE request and include it in the sent 18x response.

– If a Bearer Capability IE is present in the CPG ATP then map this parameter into the sent 18x response.

Select the second stated codec in the SDP offer of the initial INVITE request and:

– Send this codec as the first stated codec in the 'm' line of the SDP answer in the 18x response.

– Instruct the Media Gateway to use the selected codec.

Table 6-39 – Sending of Fallback indication

← 180 Ringing or 183 Session Progress	←CPG
PSTN XML BearerCapability = "Speech"	TMU "Speech" ATP No BC
PSTN XML BearerCapability = "3.1 kHz audio"	TMU "3.1 kHz audio" ATP No BC
PSTN XML BearerCapability received in the ATP ("speech" or "3.1 kHz audio")	TMU "Speech or 3.1 kHz audio" ATP BC ("speech" or "3.1 kHz audio")

6.6.3 Sending of 181 Call is being forwarded

For profiles A and B: As a network option if the I-IWU supports the P-Early-Media header and a P-Early was received in the initial INVITE request when an CPG is received and a Redirection number, a Call Diversion Information parameter and the Notification subscription options is not set to 'presentation not allowed', and an optional Backward Call Indicators parameter, indicate that in-band information is available, a 181 Call is being forwarded is sent with a present P-Earl-Media header. The mapping of the Call diversion parameters are described in clause B.6.

6.7 Receipt of Answer Message (ANM)

The mapping of ANM is shown in Table 6-40. On receipt of BICC/ISUP ANM, the I-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 TMR/USI used on the BICC/ISUP side.

Table 6-40 – Receipt of ANM at the I-IWU

← Message sent to SIP	← Message received from BICC/ISUP
200 OK INVITE	ANM

When Profile C is applicable, the Answer message is encapsulated in a 200 OK INVITE final response.

For profiles A and B: As a network option if the I-IWU supports the PSTN XML schema and the P-Early-Media header and a P-Early-Media header was received in the initial INVITE request: The I-IWU shall map the Access Transport parameter received in the ANM into the PSTN XML element in the sent 200 OK INVITE.

Table 6-41 – Interworking of ANM Access Transport parameter into PSTN XML element in the 200 OK

← 200	← ANM	
PSTN XML	**ISUP parameter**	**Content**
ProgressIndicator	Access Transport parameter	Progress indicator
HighLayerCompatibility (Note 2)		High layer compatibility
LowLayerCompatibility (Note 2)		Low layer compatibility
BearerCapability (Note 1, Note 2)		Bearer Capability
BearerCapability (Note 1, Note 2)	Transmission Medium Used parameter (Note 1)	
NOTE 1 – See clause 6.7.2 Transmission Medium Used parameter (TMU). NOTE 2 – The I-IWU shall only provide this IE if it interworks media encoded in any of the formats in Table 6-11 (ITU-T G.711, Clearmode or t38) without transcoding. If both TMU and a BC in the ATP have been received, the BC in the ATP shall be mapped.		

For profiles A and B: On receipt of an ANM and the Access Transport parameter contains a Bearer Capability Information element set to 'unrestricted digital information with tones/announcement' and no Transmission Medium Used parameter is present, the PSTN XML BearerCapability element is set to '**10001**' (unrestricted digital information with tones/announcement).

6.7.1 Mapping of ATP parameter into PSTN XML elements in the 200 OK

For profiles A and B: As a network option if the I-IWU supports the PSTN XML schema: The I-IWU maps the Backward Call Indicator or the optional Backward Call Indicator in the received ANM (if present) into a 200 OK INVITE response PSTN XML ProgressIndicator element or an additional PSTN XML ProgressIndicator element if an ATP Progress Indicator IE was received as described in Table 6-42.

Table 6-42 – Interworking of ANM parameter into 200 OK INVITE

← 200 OK	← ANM
PSTN XML element "ProgressIndicator" "ProgressDescription" value	**Backward Call Indicator** **Optional Backward Call Indicator**
Value 8 ("In-band information or an appropriate pattern is now available")	Optional Backward Call Indicators parameter In-band Information Indicator "in-band information or an appropriate pattern is now available"
Value 1 ("Call is not end-to-end ISDN: further call progress information may be available in-band")	Backward Call Indicators parameter ISDN User Part Indicator 0 "ISDN User Part not used all the way"
Value 2 ("Destination address is non-ISDN")	Backward Call Indicators parameter ISDN User Part Indicator 1 "ISDN User Part used all the way" ISDN Access Indicator 0 "Terminating access non-ISDN"

Table 6-42 – Interworking of ANM parameter into 200 OK INVITE

<table>
<tr><th>← 200 OK</th><th>← ANM</th></tr>
<tr><td>Value 7 ("Terminating access ISDN")</td><td>Backward Call Indicators parameter
Interworking Indicator
0 no interworking encountered (Signalling System No. 7 all the way)
ISDN User Part Indicator
1 "ISDN User Part used all the way"
ISDN Access Indicator
1 "Terminating access ISDN"</td></tr>
<tr><td colspan="2">NOTE – The ProgressIndicator "Coding Standard" parameter shall be set to "00 (ITU-T standardized coding)". The default value for the ProgressIndicator "Location" parameter is "0011 (Transit Network)".</td></tr>
</table>

NOTE – The order of ProgressIndicators elements within the same PSTN XML element is irrelevant.

6.7.2 Fallback in the succeeding network

For profiles A and B: As a network option if the I-IWU supports the PSTN XML schema: If the I-IWU receives two PSTN XML BearerCapability elements within the initial INVITE Request and the procedure in clause 6.1.3.5 is applied, on receipt of a ANM containing a Transmission Medium Used parameter and optional an Access Transport Parameter, the I-IWU shall send a 200 OK INVITE response with the following PSTN XML elements present:

– If no Bearer Capability IE is present in the ANM ATP, copy the first received PSTN XML BearerCapability element, the InformationTransferCabability equal to the TMU value from the initial INVITE request and include it in the sent 200 OK response.

– If a Bearer Capability IE is present in the ANM ATP then map this parameter into the sent 200 OK response.

Select the second stated codec in the SDP offer of the initial INVITE request and:

– Send this codec as the first stated codec in the 'm' line of the SDP answer in the 200 OK response.

– Instruct the Media Gateway to use the selected codec.

Table 6-43 – Sending of Fallback indication

← 200 OK	←ANM
PSTN XML BearerCapability = "Speech"	TMU "Speech" ATP No BC
PSTN XML BearerCapability = "3.1 kHz audio"	TMU "3.1 kHz audio" ATP No BC
PSTN XML BearerCapability received in the ATP ("speech" or "3.1 kHz audio")	TMU "Speech or 3.1 kHz audio" ATP BC ("speech" or "3.1 kHz audio")

For profiles A and B: If the procedure in clause 6.1.3.5 is applied and no TMU parameter in the ANM is received, select the first stated codec in the SDP offer of the received initial INVITE request and

– send this codec as the first stated codec in the 'm' line of the SDP answer in the 200 OK response.

– Instruct the Media Gateway to use the selected codec.

NOTE – Fallback does not occur at the terminating side.

6.8 Through connection of the bearer path

Through connection of the bearer path is applicable to Type 1 or Type 3 gateways only.

6.8.1 Through connection of the bearer path (ISUP)

Through connection at the I-IWU shall apply in both directions.

For the Profile C (SIP-I) case, the I-IWU shall follow the through-connection procedures in [ITU-T Q.764] for the transit exchange.

6.8.2 Through connection of the bearer path (BICC)

The bearer path shall be connected in both directions when both of the following conditions are satisfied:

- The BICC outgoing bearer set-up procedure [ITU-T Q.1902.4] is successfully completed, and;
- the I-IWU determines (using the procedures defined in [IETF RFC 3312]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable).

In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "*notification not required*", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in [IETF RFC 3312]) that sufficient preconditions have been met for the session to proceed.

6.9 Receipt of Suspend message (SUS) network initiated

If the I-IWU is the controlling exchange for the Suspend procedure, the actions taken on the BICC/ISUP side upon receipt of the Suspend message (SUS) are described in clauses 2.4.1c of [ITU-T Q.764] and 10.2.1c of [ITU-T Q.1902.4].

SUS is not interworked in Profile A or B operation. In the Profile C (SIP-I) case, the SUS is encapsulated in the MIME body of an INFO request. This is summarized in Table 6-44.

Table 6-44 – INFO sent to SIP upon receipt of SUS (Profile C only)

← Message sent to SIP	← Message received from BICC/ISUP
INFO	SUS

6.10 Receipt of Resume message (RES) network initiated

If the I-IWU is the controlling exchange for the Resume procedure, the actions taken on the BICC/ISUP side upon receipt of the Resume message (RES) are described in clauses 2.4.2c of [ITU-T Q.764] and 10.2.2c of [ITU-T Q.1902.4].

RES is not interworked in Profile A or B operation. In the Profile C (SIP-I) case, the I-IWU shall encapsulate the RES in an INFO method. This is summarized in Table 6-45.

Table 6-45 – Receipt of Resume message (RES) network initiated (Profile C only)

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

6.11 Release procedures at the I-IWU

6.11.1 Receipt of BYE/CANCEL

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

On receipt of SIP BYE or CANCEL, the I-IWU shall invoke the BICC Release sending procedure [ITU-T Q.1902.4] on the BICC side.

In the case of Profile C (SIP-I), the encapsulated REL received in a BYE message shall be passed to BICC/ISUP procedures without modification. A received CANCEL message shall be treated as described for Profile A or B below.

For Profile A or B

If the Reason header field with ITU-T Q.850 Cause Value is included in the BYE or CANCEL, then the Cause Value shall be mapped to the ISUP Cause Value field in the ISUP REL. The mapping of the Cause Indicators parameter to the Reason header is shown in Table 6-46. Table 6-47 shows the coding of the Cause Value in the REL if it is not available from the Reason header field. In both cases, the Location Field shall be set to "*network beyond interworking point*".

Table 6-46 – Mapping of SIP Reason header fields into Cause Indicators parameter

Component of SIP Reason header field	Component value	BICC/ISUP Parameter field	Value
protocol	"*Q.850*"	Cause Indicators parameter	–
protocol-cause	"*cause = XX*" (Note 1)	Cause Value	"*XX*" (Note)
–	–	Location	*"network beyond interworking point"*
NOTE – "XX" is the Cause Value as defined in [ITU-T Q.850].			

Table 6-47 – 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 call clearing)
CANCEL	Cause Value No. 31 (normal, unspecified)

For profiles A and B: As a network option: If the PSTN XML schema is supported, the PSTN XML elements shall be interworked into the ISUP Access Transport parameter as described in Table 6-48.

Table 6-48 – Mapping of PSTN XML elements into ISUP ATP parameters

BYE or CANCEL →	REL →	
PSTN XML	**ISUP Parameter**	**Content**
HighLayerCompatibility	Access Transport parameter	High layer compatibility
LowLayerCompatibility		Low layer compatibility

6.11.2 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 reselection, an ISUP RLC is returned to the ISUP side.

On receipt of a BICC REL, the I-IWU invokes the BICC Release reception procedures (clause 11.6 of [ITU-T Q.1902.4]), on the BICC side.

The above paragraphs are applicable to Type 1 or 3 gateways only.

A Reason header field containing the received ([ITU-T Q.850]) Cause Value of the REL shall 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 6-49.

Table 6-49 – Mapping of Cause Indicators parameter into SIP Reason header fields

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

On receipt of REL before receiving ANM or CON, the I-IWU shall send the appropriate SIP status code in a final response to the SIP peer. See Table 6-50 for the mapping from BICC/ISUP Cause Value to SIP status code. A BICC/ISUP Cause Value that does not appear in Table 6-50 shall have the same mapping as the appropriate ITU-T Q.850 class defaults.

For Profile C (SIP-I), 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 6-50 for profiles A and B.

Table 6-50 – Receipt of the Release message (REL)

← SIP Message	← REL Cause Indicators parameter
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*")

Table 6-50 – Receipt of the Release message (REL)

← SIP Message	← REL Cause Indicators parameter
500 Server Internal Error	Cause Value No. 4 ("*Send special information tone*")
404 Not Found	Cause Value No. 5 ("*Misdialled trunk prefix*")
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*")
410 Gone	Cause Value No. 23 ("*redirection to new destination*")
433 Anonymity Disallowed	Cause value No 24 (Call rejected due to feature at the destination)
483 Too Many Hops	Cause Value No. 25 ("*Exchange routing error*")
480 Temporarily unavailable	Cause value No 26 (Non-selected user clearing)
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)
486 Busy here if Diagnostics Indicator includes the (CCBS indicator = "*CCBS possible*") else 503 Service Unavailable	Cause Value in the Class 010 (resource unavailable, Cause Value No. 34)
500 Server Internal Error	Cause value No 38 (Network out of order)
503 Service Unavailable	Cause value No 41 (Temporary failure)
503 Service Unavailable	Cause value No 42 (Switching equipment congestion)
500 Server Internal Error	Cause value No 43 (Access information discarded)
503 Service Unavailable	Cause value No 44 (Requested channel not available)
500 Server Internal Error	Cause value No 46 (Precedence call blocked)
503 Service Unavailable	Cause value No 47 (Resource unavailable, unspecified) - (class default)
488 Not acceptable here	Cause Value No. 50 ("*requested facility not subscribed*")
603 Decline	Cause Value No. 55 ("*incoming calls barred within CUG*")
603 Decline	Cause Value No. 57 ("*bearer capability not authorized*")

Table 6-50 – Receipt of the Release message (REL)

← SIP Message	← REL Cause Indicators parameter
503 Service Unavailable	Cause Value No. 58 ("*bearer capability not presently available*")
501 Not Implemented	Cause Value No. 63 ("*service or 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)
403 Forbidden	Cause Value No. 87 ("*user not member of CUG*")
606 Not Acceptable	Cause Value No. 88 ("*incompatible destination*")
403 Forbidden	Cause Value No. 90 ("*Non-existent CUG*")
500 Server Internal Error	Cause Value No. 91 ("*invalid transit network selection*")
513 Message too large	Cause Value No. 95 ("*invalid message, unspecified*") (Class default)
501 Not Implemented	Cause Value No. 97 ("*Message type non-existent or not implemented*")
501 Not Implemented	Cause value No 98 (Message not compatible with call state or 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)

On receipt of REL after receiving ANM or CON, the I-IWU shall send BYE. For Profile C (SIP-I), this BYE message shall encapsulate the received REL message.

For profiles A and B: As a network option: If the I-IWU supports the PSTN XML schema, the I-IWU shall interwork a received ISUP Access Transport parameter in the REL into the corresponding PSTN XML element in an unsuccessful final response or BYE message as described in Table 6-51.

Table 6-51 – Mapping of ISUP ATP parameters into PSTN XML elements

←4xx,5xx,6xx or BYE	←REL	
PSTN XML	**ISUP Parameter**	**Content**
ProgressIndicator	Access Transport parameter	Progress indicator
HighLayerCompatibility		High layer compatibility
LowLayerCompatibility		Low layer compatibility

6.11.3 Autonomous release at I-IWU

Table 6-52 shows the trigger events at the I-IWU and the release initiated by the I-IWU when the call is traversing from SIP to BICC/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 (BICC) side are required.

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

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

For Profile C (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 6-50 for profiles A and B.

Table 6-52 – Autonomous release at I-IWU

← SIP	Trigger event	REL →
		Cause Indicators parameter
484 Address Incomplete	Determination that insufficient digits are received. See Note in clause 6.1. Receipt of subsequent INVITE within overlap procedure, see clause 6.2.	Not applicable.
480 Temporarily Unavailable	Congestion at the IWU.	Not applicable.
BYE	BICC/ISUP procedures result in release after answer.	According to BICC/ISUP procedures.
BYE	SIP procedures result in release after answer.	127 (Interworking unspecified)
500 Server Internal Error	Call release due to the BICC/ISUP compatibility procedure (Note)	According to BICC/ISUP procedures.

Table 6-52 – Autonomous release at I-IWU

← SIP	Trigger event	REL →
		Cause Indicators parameter
484 Address Incomplete	Call release due to expiry of T7 within the BICC/ISUP procedures	According to BICC/ISUP procedures.
480 Temporarily Unavailable	Call release due to expiry of T9 within the BICC/ISUP procedures	According to BICC/ISUP procedures.
480 Temporarily Unavailable	Other BICC/ISUP procedures result in release before answer	According to BICC/ISUP procedures.
NOTE – If the I-IWU receives unrecognized ISUP or BICC signalling information and determines that the call needs to be released based on the coding of the compatibility indicators, then see clauses 2.9.5.2 of [ITU-T Q.764] and 13.4.3 of [ITU-T Q.1902.4].		

6.11.4 Receipt of RSC, GRS or CGB (ISUP)

Table 6-53 shows the message sent by the I-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.

a) The I-IWU sends BYE if it has already received an ACK for the 200 OK INVITE it had sent.

b) If I-IWU has sent 200 OK INVITE but has not yet received an ACK for the 200 OK INVITE, then the I-IWU shall wait until it receives the ACK for the 200 OK INVITE before sending the BYE.

c) In all other cases the I-IWU sends 480 Temporarily Unavailable.

– The affected circuit in case of an RSC is identified by the CIC value.

– The affected circuit(s) in case of CGB are (is) identified by the CIC value and the Range and Status parameter.

– The affected circuit(s) in case of CGB are (is) identified by the CIC value and the Range and Status parameter, the status subfield is absent.

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.

A Reason header shall be included in the BYE message or unsuccessful final response, set to the Cause value of the REL message that is internally generated on the ISUP side.

In the Profile C (SIP-I) case, the SIP BYE or 480 Temporarily Unavailable Error message shall encapsulate the REL generated by ISUP procedures, rather than the RSC, GRS or CGB message which caused it to be generated.

Table 6-53 – Receipt of RSC, GRS or CGB messages (ISUP)

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

6.11.5 Receipt of RSC or GRS (BICC)

Table 6-54 shows the message sent by the I-IWU upon receipt of a BICC RSC message or GRS message, when at least one backward BICC message relating to the call has already been received.

a) The I-IWU sends BYE if it has already received an ACK for the 200 OK INVITE it had sent.

b) If the I-IWU has sent 200 OK but has not yet received an ACK for the 200 OK INVITE, then the I-IWU shall wait until it receives the ACK for the 200 OK INVITE before sending the BYE.

c) In all other cases, the I-IWU sends 480 Temporarily Unavailable.

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

In the Profile C (SIP-I) case, the SIP BYE or 480 Temporarily Unavailable message shall encapsulate the REL generated by BICC procedures, rather than the RSC or GRS message which caused it to be generated.

Table 6-54 – Receipt of RSC or GRS messages (BICC)

← SIP	← Message received from BICC
480 Temporarily Unavailable or BYE	Reset CIC message (RSC)
480 Temporarily Unavailable or BYE	CIC Group Reset message (GRS)

6.11.6 Receipt of REFER request

For profiles A and B: On receipt of a REFER request the I-IWU rejects this request by sending a 403 Forbidden final response.

NOTE – The I-IWU may also decide, for example, to execute the REFER request as specified in [IETF RFC 3515] as an operator option, but such handling is outside of the scope of this Recommendation.

7 Outgoing call interworking from BICC/ISUP to SIP at O-IWU

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

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

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

If the address information received from the preceding BICC/ISUP exchange is not in the form of an ITU-T 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 signalling interworking requirements for a basic call at the O-IWU. It is split into subclauses based upon the messages sent or received on the outgoing (SIP) interface of the O-IWU. Only messages that are generated as a result of interworking to/from the incoming BICC/ISUP side of the O-IWU are considered in this interworking. Messages that are generated as a result of a local protocol state machine are not redescribed in this Recommendation.

In the case of Type 2 or 4 gateways as defined in [b-ITU-T Q.Sup.45], the O-IWU shall (in addition to the procedures outlined within this clause) follow the BICC-specific procedures outlined in clause A.2.

For Profile C (SIP-I) operation, ISUP message segmentation must be handled as described in clause 5.4.3.3.

7.1 Sending of the first INVITE

After performing the normal BICC/ISUP handling for incoming address messages (IAM possibly followed by SAMs) and choosing to route the call to the SIP network domain, the O-IWU determines from configuration whether *en bloc* addressing is to be applied on the SIP side.

1) If *en bloc* addressing is to be used, the O-IWU shall determine the end of address signalling from the earlier of the following criteria a to d and then invoke the appropriate outgoing SIP signalling procedure as described in this clause.

   End of address signalling is determined by the following criteria:

   a) by receipt of an end-of-pulsing (ST) signal; or

   b) by receipt of the maximum number of digits used in the national numbering plan; or

   c) by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party; or

   d) by observing that timer T_{OIW1} has expired.

   If end of address signalling is determined in accordance with criteria a, b and c above, timer T_{OIW2} shall be started on sending of the INVITE. For profiles A and B: Also, if the PSTN XML body is supported as a network option, the O-IWU shall insert the PSTN XML sendingCompleteIndication.

NOTE 1 – *En bloc* is preferred, and is required for Profile A.

2) If overlap addressing is to be used towards the SIP network, then, after the minimum number of digits required for routing the call has been received, the O-IWU shall:

   – start timer T_{OIW2} and invoke the appropriate outgoing SIP signalling procedure as described in this clause; and

   – be prepared to process SAM as described in clause 7.2.1.

   – be prepared to handle incoming SIP 404 or 484 error responses as detailed in clause 7.7.6.1

NOTE – An SIP INVITE request with incomplete address information will be rejected with a SIP 404 or 484 error response.

An O-IWU shall support both the SIP preconditions and 100 rel extensions and indicate the support of the SIP preconditions and 100rel extensions in the INVITE request, unless the note below applies.

NOTE – If the O-IWU is deployed in an IMS network that by local configuration serves no user requiring preconditions, it may send the INVITE request without indicating support of preconditions.

The O-IWU will invoke the outgoing SIP signalling 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/SAM.

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

C) Send INVITE without precondition upon receipt of BICC IAM/SAM.

D) Send INVITE with precondition upon receipt of BICC IAM/SAM.

Details of the procedures are described in this clause. Coding of the INVITE sent by the O-IWU is specified in clauses 7.1.2 to 7.1.6.

For Profile C (SIP-I), the IAM resulting from the application of BICC/ISUP procedures and the procedures of this clause is encapsulated in the outgoing INVITE.

If timer T_{OIW2} expires, an early ACM is sent to the ISUP or BICC network. See clause 7.4.

A) Sending INVITE without precondition for ISUP IAM/SAM

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

INVITE is sent when the ISUP IAM (possibly followed by SAMs) 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/SAM

INVITE with precondition is sent on receipt of ISUP IAM (possibly followed by SAMs). 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 2 – Configured procedures may delay the INVITE until local resources have been reserved on the outgoing bearer path.

The O-IWU should initiate the precondition signalling procedure using the SDP offer in the INVITE. The precondition signalling 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. An SDP offer (e.g., a SIP UPDATE request, as defined in [IETF RFC 3311]) shall be sent for each early SIP dialogue for which the received provisional response indicated support of preconditions confirming that all the required local preconditions have been met.

2) The requested preconditions are met in the SIP network.

NOTE 3 – For Profile A, the signalling of "preconditions being met" always occurs within the SDP offer in the UPDATE message.

Otherwise, the O-IWU shall indicate whether the preconditions are met, depending on the status of the local resource reservation. If the local preconditions are not met the O-IWU should set the media stream to inactive mode (by including an attribute "a=inactive"). If the local configuration indicates that O-IWU is deployed in the IMS network that serves users supporting SIP precondition mechanism, the attribute "a=inactive" may be omitted when the initial SDP offer indicates local preconditions are not met. If the initial SDP offer indicates local preconditions are fulfilled, the O-IWU shall not set the media stream to inactive mode.

CANCEL or BYE (according to the rule in clause 7.7.1) 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 clause 7.7.1) shall be sent on the SIP side if internal resource reservation was unsuccessful. See clause 7.7.3 for further details.

C) INVITE without precondition for BICC IAM/SAM

Incoming BICC procedures apply, with the following clarifications and exceptions as to when the INVITE is to be sent.

The sending of the INVITE is delayed until all the following conditions are satisfied:

1) If the incoming IAM indicated "*COT to be expected*", a Continuity message, with the Continuity Indicators parameter set to "*continuity*" shall be received.
2) One of the following events, which indicate successful completion of bearer set-up, shall be received by the Incoming bearer set-up procedure (clause 7.5 of [ITU-T Q.1902.4]):
    2.1) Bearer Set-up indication for the forward bearer set-up case where the incoming Connect Type is "*notification not required*".
    2.2) APM with Action indicator set to "*Connected*" for the forward bearer set-up cases (with or without bearer control tunnelling) where the incoming Connect Type is "*notification required*", and for the fast set-up (backward) case.
    2.3) Bearer Set-up Connect indication for the backward bearer set-up case.
    2.4) BNC set-up success indication for cases using bearer control tunnelling, except as identified in item 2.2 above.

INVITE shall not be sent if the Continuity message is not received, i.e., the BICC timer T8 expires.

D) INVITE with precondition for BICC IAM/SAM

INVITE with precondition is sent on receipt of BICC IAM (possibly followed by SAMs). Incoming BICC procedures apply, with the following clarifications and exceptions as to when a confirmation of the precondition being met is to be sent.

NOTE 4 – Configured procedures may delay the INVITE until local resources have been reserved on the outgoing bearer path.

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

1) If the incoming IAM indicated "*COT to be expected*", a Continuity message, with the Continuity Indicators parameter set to "*continuity*" shall be received.

2) One of the following events, which indicate successful completion of bearer set-up, shall also be received by the Incoming bearer set-up procedure (clause 7.5 of [ITU-T Q.1902.4]), depending on the procedure being applied:

2.1) Bearer Set-up indication for the forward bearer set-up case where the incoming Connect Type is "*notification not required*".

2.2) APM with Action indicator set to "*Connected*" for the forward bearer set-up cases (with or without bearer control tunnelling) where the incoming Connect Type is "*notification required*", and for the fast set-up (backward) case.

2.3) Bearer Set-up Connect indication for the backward bearer set-up case.

2.4) BNC set-up success indication for cases using bearer control tunnelling, except as identified in item 2.2 above.

3) The requested preconditions are met in the SIP network.

NOTE 5 – For Profile A, the signalling of "preconditions being met" always occurs within the SDP offer in the UPDATE message.

CANCEL or BYE (according to the rule in clause 7.7.1) shall be sent if the Continuity message is not received, i.e., the BICC 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 clause 7.7.1) shall be sent on the SIP side if internal resource reservation was unsuccessful. See clause 7.7.3 for further details.

7.1.1 IAM without calling party number

If no calling party number is received in the incoming IAM message, as a network option, the O-IWU may send an INR message to request the calling party number and not send the INVITE request until receiving an INF message with calling party number. If no calling party number is received in the INF message, O-IWU may reject or continue the call based on local configuration.

For all cases of sending INVITE (A, B, C and D), 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→	INVITE→
Called Party Number	Request-URI (see clauses 7.1.3 and 7.2)
	To (see clause 7.1.3)
Calling Party Number	P-Asserted-Identity (see clause 7.1.4)
	Privacy (see clause 7.1.4)
	From (see clause 7.1.4)
Generic Number (*"additional calling party number"*)	From (see clause 7.1.4)
	Privacy (see clause 7.1.4)
Hop Counter	Max-Forwards (see clause 7.1.5)
TMR/USI	Message Body (application/SDP) (see clause 7.1.2)
ISUP Message	Message Body (application/ISUP) (Note)
Location Number	P-Access-Network-Info
NOTE – Profile C only. See clause 5.4.1.2.	

7.1.2 Coding of SDP media description lines from TMR/USI

The TMR parameter plus the optional User Service Information parameter of the IAM received by the O-IWU indicate the user-requested bearer service characteristics. Their codes should be mapped to the SDP information. [ITU-T Q.1902.3] and [ITU-T Q.763] provide an exhaustive listing of the available codes in the TMR and USI parameters. Generally, any combination of these codes can be mapped into any SDP information as long as transcoding is available.

The O-IWU for profiles A and B shall be capable of encoding the SDP for the AMR codec, which is specified in [IETF RFC 3267]: "RTP payload format and file storage format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) audio codec" the O-IWU shall include the AMR codec transported according to [IETF RFC 4867] in the SDP offer, unless the note below applies. Within the SDP offer, the O-IWU should also provide SDP RR and RS bandwidth modifiers specified in [IETF RFC 3556] to disable RTCP. The O-IWU may include other codecs according to operator policy.

NOTE – If the O-IWU is deployed in an IMS network that by local configuration serves no user equipment that implements the AMR codec, then the AMR codec may be excluded from the SDP offer.

If the O-IWU operates as an international outgoing gateway and if ITU-T G.711 encoding is offered then the following cases apply. These procedures reflect the requirement that transcoding between A-law and µ-law has to occur in a µ-law network only.

– If the call is coming from an A-law PSTN network, the O-IWU shall send an SDP Offer with A-law (PCMA), but not µ-law (PCMU) included in the media description.
– If the call is coming from a µ-law PSTN network, the O-IWU shall send an SDP Offer with both µ-law (PCMU) and A-law (PCMA) included in the media description and PCMU shall take precedence over PCMA.

7.1.2.1 Transcoding not available at the O-IWU

Table 7-2 provides the mapping relations from TMR/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 TMR/USI: BICC/ISUP to SIP

TMR parameter	USI parameter		HLC IE in ATP	m= line			b= line	a= line
TMR codes	**Information Transport Capability**	**User Information Layer 1 Protocol Indicator**	**High Layer Characteristics Identification**	**<media>**	**<transport>**	**<fmt-list>**	**<modifier>:<bandwidth-value>**	**a=rtpmap:<payload type> <encoding name>/<clock rate> [/<encoding parameters>]**
"speech"	*"Speech"*	*"G.711 μ-law"*	Ignore	audio	RTP/AVP	0 (and possibly 8) (Note 1)	AS:64 and RTP/UDP/IP overhead	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 second Dynamic PT) (Note 1)	AS:64 and RTP/UDP/IP overhead	rtpmap:<payload type> PCMU/8000 (and possibly rtpmap:<payload type> PCMA/8000) (Note 1)
"speech"	*"Speech"*	*"G.711 A-law"*	Ignore	audio	RTP/AVP	8	AS:64 and RTP/UDP/IP overhead	rtpmap:8 PCMA/8000
"speech"	*"Speech"*	*"G.711 A-law"*	Ignore	audio	RTP/AVP	Dynamic PT	AS:64 and RTP/UDP/IP overhead	rtpmap:<payload type> PCMA/8000
"3.1 kHz audio"	USI Absent		Ignore	audio	RTP/AVP	0 and/or 8 (Note 1)	AS:64 and RTP/UDP/IP overhead	rtpmap:0 PCMU/8000 and/or rtpmap:8 PCMA/8000 (Note 1)
"3.1 kHz audio"	*"3.1 kHz audio"*	*"G.711 μ-law"*	(Note 3)	audio	RTP/AVP	0 (and possibly 8) (Note 1)	AS:64 and RTP/UDP/IP overhead	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000) (Note 1)
"3.1 kHz audio"	*"3.1 kHz audio"*	*"G.711 A-law"*	(Note 3)	audio	RTP/AVP	8	AS:64 and RTP/UDP/IP overhead	rtpmap:8 PCMA/8000

Table 7-2 – Coding of SDP media description lines from TMR/USI: BICC/ISUP to SIP

TMR parameter	USI parameter		HLC IE in ATP	m= line			b= line	a= line
TMR codes	**Information Transport Capability**	**User Information Layer 1 Protocol Indicator**	**High Layer Characteristics Identification**	**<media>**	**<transport>**	**<fmt-list>**	**<modifier>: <bandwidth-value>**	**a=rtpmap:<payload type> <encoding name>/ <clock rate> [/<encoding parameters>]**
"3.1 kHz audio"	*"3.1 kHz audio"*		*"Facsimile Group 2/3"*	image	udptl	t38	AS:64 and RTP/UDP/IP overhead	Based on [ITU-T T.38].
"3.1 kHz audio"	*"3.1 kHz audio"*		*"Facsimile Group 2/3"*	image	tcptl	t38	AS:64 and RTP/UDP/IP overhead	Based on [ITU-T T.38].
"3.1 kHz audio"	*"3.1 kHz audio"*		*"Facsimile Group 2/3"*	Audio (Note 6)	RTP/AVP	0, 8, or <dynamic-PT>	AS: (64 + RTP/UDP/IP overhead)	rtpmap:0 PCMA/8000 or rtpmap:8 PCMU/8000 or rtpmap:<dynamic-PT> PCMA/8000 or rtpmap:<dynamic-PT> PCMU/8000
"64 kbit/s unrestricted preferred "	*"Speech/ 3.1KHz audio" (Note 4)*	N/A	Ignore	audio	RTP/AVP	9	AS:64 and RTP/UDP/IP overhead	rtpmap:< payload type> CLEARMODE/8000 (Note 2), (Note 5)
"64 kbit/s unrestricted"	*"Unrestricted digital information"*	N/A	Ignore	audio	RTP/AVP	Dynamic PT	AS:64 and RTP/UDP/IP overhead	rtpmap:<payload type> CLEARMODE/8000 (Note 2)
"2 x 64 kbit/s unrestricted"	*"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
"1536 kbit/s unrestricted"	*"Unrestricted digital information"*	N/A	Ignore	FFS	FFS	FFS	FFS	FFS

Table 7-2 – Coding of SDP media description lines from TMR/USI: BICC/ISUP to SIP

TMR parameter	USI parameter		HLC IE in ATP	m= line			b= line	a= line
TMR codes	**Information Transport Capability**	**User Information Layer 1 Protocol Indicator**	**High Layer Characteristics Identification**	**<media>**	**<transport>**	**<fmt-list>**	**<modifier>: <bandwidth-value>**	**a=rtpmap:<payload type> <encoding name>/ <clock rate> [/<encoding parameters>]**
"1920 kbit/s unrestricted"	*"Unrestricted digital information"*	N/A	Ignore	FFS	FFS	FFS	FFS	FFS
"N × 64 kbit/s unrestricted", N from 3 to 29	*"Unrestricted digital information"*	N/A	Ignore	FFS	FFS	FFS	FFS	FFS

NOTE 1 – Both PCMA and PCMU required under the conditions stated in clause 7.1.2.

NOTE 2 – CLEARMODE is specified in [IETF RFC 4040].

NOTE 3 – HLC normally absent in this case. It is possible for HLC to be present with the value "Telephony", although clause 6.3.1 of [b-ITU-T Q.939] indicates that this would normally be accompanied by a value of "Speech" for the Information Transfer Capability element.

NOTE 4 – In this case, the USI Prime parameter will also be present and will indicate "Unrestricted Digital Information with tones/announcements".

NOTE 5 – After the CLEARMODE codec, additional speech codecs such as AMR and/or ITU-T G.722 and/or ITU-T G.711 available via transcoding or reframing should be offered in the same m-line.

NOTE 6 – FAX can either be transported according to [ITU-T T.38] using the "image" media type or as in-band Voice band data over IP using the "audio" media type.

NOTE 7 – As alternative or in addition to the m-line containing the CLEARMODE codec, an IWU supporting the multimedia interworking may add an m-line for speech codecs and an m-line for video codecs.

To avoid transcoding or to support non-speech services, the O-IWU may add media derived from the incoming ISUP information according to Table 7-2. The support of the media listed in Table 7-2 is optional. For Profile B and if the O-IWU supports the PSTN XML body as a network option and adds media derived from the incoming ISUP information according to Table 7-2, the O-IWU shall also map the media related ISUP information into the XML body as shown in Table 7-11.

7.1.3 Request-URI and To header field

The Called Party Number parameter of the IAM and possibly the Address Signals indicators in the Subsequent Number parameter of SAMs contain the forward address information to derive the userinfo component of the INVITE Request-URI.

NOTE – The O-IWU follows existing BICC/ISUP procedures 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 (and Subsequent Number parameters, if any) 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 and shall contain:

– an E.164 International public telecommunication number prefixed by a "+" sign (e.g., tel:+4911231234567), or

– a non-E.164 number (national operator option for service numbers), expressed as a local number as per [IETF RFC 3966].

Table 7-3 –Mapping ISUP Called Party Number to SIP Request-URI and To header field

IAM		**INVITE**	
BICC/ISUP Parameter / field	**Value**	**SIP component**	**Value**
Called Party Number (Note 3)		Request-URI and To header field	display-name (optional) and addr-spec derived from Called Party Number parameter address signals
Nature of Address Indicator (Note 2) (Note 6)	"*national (significant) number*"	Tel URI or SIP URI	Insert "+CC" before the Address signals (Note 1)
	"*international number*"		Insert "+" before the Address signals
	"Network-specific number" or *"reserved for national use"*		according to local policies should either be: – a global number (+CC), if the called party number may be converted into an E.164 address OR, depending on operator's requirements may be converted into – local number (with a phone-context parameter) (Note 4) (Note 5)

Table 7-3 –Mapping ISUP Called Party Number to SIP Request-URI and To header field

IAM	INVITE
NOTE 1 – CC = Country Code of the network in which the O-IWU is located. NOTE 2 – The usage of "Nature of Address Indicator" value "unknown" is allowed but the mapping is not specified in this Recommendation. NOTE 3 – If the address signals received in the ISUP Called Party Number contain a sending terminated signal (hexadecimal digit F), then this shall be discarded or if the O-IWU supports the PSTN XML body as a network option then the PSTN XML sendingCompleteIndication shall be set. NOTE 4 – Mapping between Nature of Address Indicator values and phone-context values is provisioned in the IWU. Setting of value of phone-context is depending on local operator's policies. NOTE 5 – Network-specific number or reserved for national use shall be translated into E.164 format numbers except if the local operator's policy requires keeping it in local format (e.g., for national reasons E.164 numbers cannot be used for such purpose). In the latter case the mapping shall be done as indicated in the table. NOTE 6 – The values "Network routing number in national (significant) number format", "Network routing number in network specific number format" or "Network routing number concatenated with called directory number" are used when number portability is supported. For the mapping see clause 7.1.12.	

7.1.4 P-Asserted-Identity, From and Privacy header fields

Table 7-4 shows the mapping from Calling Party Number and Generic Number to the SIP P-Asserted-Identity, From, and Privacy header fields in the INVITE. Table 7-5 provides details for mapping Generic Number to the From header field. Table 7-6 provides details for the mapping from Calling Party Number to P-Asserted-Identity, while Table 7-7 provides details for the mapping from Calling Party Number to the From header field. Finally, Table 7-8 provides details for mapping from the APRI subfields of Calling Party Number and Generic Number into the Privacy header field.

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

Table 7-4 – Mapping of BICC/ISUP CLI parameters to SIP header fields

<table>
<tr><td rowspan="3"></td><td></td><td colspan="5">Has a Calling Party Number parameter with complete E.164 number, with Screening Indicator = UPVP or NP (See Note 1), and with APRI = "presentation allowed" or "presentation restricted" been received?</td></tr>
<tr><td rowspan="2">Calling Party Number APRI</td><td rowspan="2"></td><td colspan="4">Has a Generic Number ("additional calling party number") with a complete E.164 number, with Screening Indicator = "UPNV", and with APRI = "presentation allowed" been received?</td></tr>
<tr><td>Generic Number APRI</td><td>P-Asserted-Identity header field</td><td>From header field: display-name (optional) and addr-spec</td><td>Privacy header field</td></tr>
<tr><td>N</td><td></td><td>N</td><td></td><td>Header field not included</td><td>unavailable@ unknown.invalid</td><td>Header field not included</td></tr>
<tr><td></td><td></td><td>Y</td><td></td><td>Header field not included</td><td>unavailable@ unknown.invalid</td><td>Header field not included</td></tr>
</table>

N (Note 4)			"presentation restricted "		Derived from the Generic Number parameter address signals (see Table 7-5) (Note 7)	priv-value =: "user" (See Table 7-8).
N (Note 4)		Y	"presentation allowed "	Header field not included	display-name derived from Generic Number (ACgPN) if possible addr-spec derived from Generic Number (ACgPN) address signals or uses network provided value (Note 7)	Header field not included

Table 7-4 – Mapping of BICC/ISUP CLI parameters to SIP header fields

<table>
<tr><td rowspan="3"></td><td></td><td colspan="5">Has a Calling Party Number parameter with complete E.164 number, with Screening Indicator = UPVP or NP (See Note 1), and with APRI = "presentation allowed" or "presentation restricted" been received?</td></tr>
<tr><td rowspan="2">Calling Party Number APRI</td><td rowspan="2"></td><td colspan="4">Has a Generic Number ("additional calling party number") with a complete E.164 number, with Screening Indicator = "UPNV", and with APRI = "presentation allowed" been received?</td></tr>
<tr><td>Generic Number APRI</td><td>P-Asserted-Identity header field</td><td>From header field: display-name (optional) and addr-spec</td><td>Privacy header field</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
</table>

Table 7-4 – Mapping of BICC/ISUP CLI parameters to SIP header fields

		Has a Calling Party Number parameter with complete E.164 number, with Screening Indicator = UPVP or NP (See Note 1), and with APRI = "presentation allowed" or "presentation restricted" been received?				
	Calling Party Number APRI		**Has a Generic Number ("*additional calling party number*") with a complete E.164 number, with Screening Indicator = "UPNV", and with APRI = "presentation allowed" been received?**			
			Generic Number APRI	**P-Asserted-Identity header field**	**From header field: display-name (optional) and addr-spec**	**Privacy header field**
Y (Note 1)	"presentation allowed"	N	-	Derived from Calling Party Number parameter Address Signals (See Table 7-6)	Tel URI or SIP URI derived from Calling Party Number parameter address signals (See Table 7-7) (Note 7)	Privacy header is not included or if included, "id" and "header" are not included (See Table 7-8)
Y (Note 1)	"presentation allowed"	Y	"presentation allowed"	Derived from Calling Party Number parameter Address Signals (See Table 7-6)	Derived from Generic Number (ACgPN) address signals (See Table 7-5) (Note 7)	Privacy header is not included or if included, "id" and "header" and "user" are not included (See Table 7-8)
Y	"presentation allowed"	Y	"presentation restricted "	Derived from Calling Party Number parameter Address Signals (See Table 7-6)	Tel URI or SIP URI derived from Calling Party Number parameter address signals (See Table 7-7) (Note 5) (Note 7)	Privacy header is not included or if included, "id" and "header" are not included (See Table 7-8)
					SIP or SIPS URI with addr-spec of anonymous@anonymous.invalid URI (Note 7)	Privacy header is not included or if included, "id" and "header" are not included (See Table 7-8).

Table 7-4 – Mapping of BICC/ISUP CLI parameters to SIP header fields

		Has a Calling Party Number parameter with complete E.164 number, with Screening Indicator = UPVP or NP (See Note 1), and with APRI = "presentation allowed" or "presentation restricted" been received?				
	Calling Party Number APRI		**Has a Generic Number ("*additional calling party number*") with a complete E.164 number, with Screening Indicator = "UPNV", and with APRI = "presentation allowed" been received?**			
			Generic Number APRI	**P-Asserted-Identity header field**	**From header field: display-name (optional) and addr-spec**	**Privacy header field**
					Derived from the Generic Number parameter address signals (see Table 7-5) (Note 5)	priv-value =: "user". and "header" and "id" are not included (See Table 7-8).
Y	"presentation restricted "	N	-	Derived from Calling Party Number parameter address signals (See Table 7-6)	SIP or SIPS URI with addr-spec of unavailable@hostportion (Note 8) (Note 7)	priv-value =: "id". (See Table 7-8)
Y	"presentation restricted "	Y	"presentation allowed"	Derived from Calling Party Number parameter address signals (See Table 7-6)	Derived from Generic Number (ACgPN) address signals (See Table 7-5) (Note 7)	priv-value =: "id".
Y	"presentation restricted "	Y	"presentation restricted "	Derived from Calling Party Number parameter address signals (See Table 7-6)	SIP or SIPS URI with addr-spec of Anonymous URI (Note 7) (Note 9)	priv-value =: "id".
					Derived from the Generic Number parameter address signals (see Table 7-5) (Note 5) (Note 7)	priv-value =: "id", "user". (See Table 7-8).
Y	"presentation restricted by network" (Note 10)	N	-	Header field not included.	Addr-spec is set to "unavailable@hostportion" (Note 6) (Note 7)	Privacy header is not included or if included, "id" and "header" are not included (See Table 7-8).
Y	"presentation restricted by network" (Note 10)	Y	"presentation allowed"	Header field not included.	Derived from Generic Number (ACgPN) address signals (See Table 7-5) (Note 7)	Privacy header is not included or if included, "id" and "header" are not included (See Table 7-8).

Table 7-4 – Mapping of BICC/ISUP CLI parameters to SIP header fields

<table>
<tr><td rowspan="3"></td><td></td><td colspan="5">Has a Calling Party Number parameter with complete E.164 number, with Screening Indicator = UPVP or NP (See Note 1), and with APRI = "presentation allowed" or "presentation restricted" been received?</td></tr>
<tr><td rowspan="2">Calling Party Number APRI</td><td rowspan="2"></td><td colspan="4">Has a Generic Number ("additional calling party number") with a complete E.164 number, with Screening Indicator = "UPNV", and with APRI = "presentation allowed" been received?</td></tr>
<tr><td>Generic Number APRI</td><td>P-Asserted-Identity header field</td><td>From header field: display-name (optional) and addr-spec</td><td>Privacy header field</td></tr>
<tr><td rowspan="2">Y</td><td rowspan="2">"presentation restricted by network"
(Note 10)</td><td rowspan="2">Y</td><td rowspan="2">"presentation restricted "</td><td rowspan="2">Header field not included.</td><td>SIP or SIPS URI with addr-spec of anonymous@anonymous.invalid URI (Note 7)</td><td>Privacy header is not included or if included, "id" and "header" are not included
(See Table 7-8).</td></tr>
<tr><td>Derived from the Generic Number parameter address signals (see Table 7-5) (Note 5) (Note 7)</td><td>Priv-value = "id" and "header" and "user" are not included
(See Table 7-8).</td></tr>
<tr><td colspan="7">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, as valid as a User Provided Verified and Passed CLI for this purpose.
NOTE 2 – Whether it is possible to derive the display-name from the Generic Number Parameter is FFS.
NOTE 3 – The "From" header may contain an "Anonymous URI". An "Anonymous URI" includes information that does not point to the calling party. [IETF RFC 3261] recommends that the display-name component contains "Anonymous". The Anonymous URI itself should have the value "anonymous@anonymous.invalid".
NOTE 4 – This combination of CgPN and ACgPN is an error case but is shown here to ensure consistent mapping across different implementations.
NOTE 5 – As a network option, the "From" header may be derived from the Generic Number parameter address signals (see Table 6-30). This option is only recommended for use within a trusted domain where an entity such as a TAS is configured to be inserted into the call path that is able to change the "From" Header content to an anonymous user identity (NOTE 7).
NOTE 6 – The setting of the hostportion is according to operator policy.
NOTE 7 – In accordance with [IETF RFC 3261] procedures, a tag shall be added to the "From" header.
NOTE 8 – The "From" header may contain an "Anonymous User Identity". An "Anonymous User Identity" includes information that does not point to the calling party and indicates that the caller has withheld their identity. The encoding of the "Anonymous User Identity" shall be as defined in [b-ETSI TS 123 003].
NOTE 9 – Network option.
NOTE 10 – This is an ETSI specific value described within [b-ETSI EN 300 356-1].</td></tr>
</table>

Table 7-5 – Mapping of Generic Number (*"additional calling party number"*) to SIP From header field

BICC/ISUP Parameter/field	Value	SIP component	Value
Generic Number Number Qualifier Indicator	"*additional calling party number"*	From header field	display-name (optional) and addr-spec
Nature of Address Indicator	"*national (significant) number"*	Addr-spec Tel URI or SIP URI	Add CC (of the country where the IWU is located) to Generic Number Address Signals then map to user portion of URI scheme used. Prefix number with "+".
	"*international number"*		Map complete GenericNumber Address Signals to user portion of URI scheme used. Prefix number with "+".
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	Display-name	display-name may be mapped from Address Signals, if possible and network policy allows it.
		Addr-spec Tel URI or SIP URI	"+" CC NDC SN mapped to user portion of URI scheme used. Prefix number with "+".

Table 7-6 – Mapping of Calling Party Number parameter to SIP P-Asserted-Identity header field

BICC/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 Tel URI or SIP URI	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 Signals	If NOA is "*national (significant) number"*	display-name	display-name may be mapped from Address Signals, if possible and network policy allows it

Table 7-6 – Mapping of Calling Party Number parameter to SIP P-Asserted-Identity header field

BICC/ISUP parameter/field	Value	SIP component	Value
Calling Party Number		P-Asserted-Identity header field	display-name (optional) and addr-spec
	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	addr-spec Tel URI or SIP URI	"+" CC NDC SN mapped to the appropriate global number portion of URI scheme used. Prefix number with "+".

Table 7-7 – Mapping of BICC/ISUP Calling Party Number parameter to SIP From header field

BICC/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. Prefix number with "+".
	"international number"		Map complete CgPN Address Signals to user portion of URI scheme used. Prefix number with "+".
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	display-name	Display-name may be mapped from Address Signals, if possible and network policy allows it.
		addr-spec	"+" CC NDC SN mapped to userinfo portion of URI scheme used. Prefix number with "+".

Table 7-8 – Mapping of BICC/ISUP APRIs into SIP Privacy header field

<table>
<tr><th>BICC/ISUP Parameter/field</th><th>Value</th><th>SIP component</th><th>Value</th></tr>
<tr><td>Calling Party Number</td><td></td><td>Privacy header field</td><td>priv-value</td></tr>
<tr><td rowspan="2">APRI
(See Table 7-4 to determine which APRI to use for this mapping)</td><td>"presentation restricted"</td><td>priv-value</td><td>"id"
("id" included only if the P-Asserted-Identity header is included in the SIP INVITE)</td></tr>
<tr><td>"presentation allowed" or "presentation restricted by network"</td><td>priv-value</td><td>Omit Privacy header
or Privacy header without "id" and "header" if other privacy service is needed)</td></tr>
<tr><td>Generic Number (ACgPN)</td><td></td><td>Privacy header field</td><td>priv-value</td></tr>
<tr><td rowspan="2">APRI</td><td>"presentation restricted"</td><td>Priv-value</td><td>"user"</td></tr>
<tr><td>"presentation allowed"</td><td>Priv-value</td><td>omit Privacy header
or Privacy header without "user" if other privacy service is needed</td></tr>
<tr><td colspan="4">NOTE – When Calling Party Number parameter is received, P-Asserted-Identity header is always derived from it as in Table 7-4.</td></tr>
</table>

7.1.4.1 "cpc" URI Parameter in P-Asserted-Identity Header

For profiles A and B: Table 7-9 shows the mapping of a Calling Party's Category received in an ISUP IAM to a "cpc" URI parameter within tel URI or the userinfo part of SIP URI with user="phone" in the P-Asserted-Identity header. When the Calling Party's Category parameter value "operator, language x" is received the O-IWU shall generate an Accept-Language header field with the value that corresponds to language x.

Table 7-9 – Mapping of the ISUP Calling Party's Category parameter to the CPC parameter

ISUP Parameter	SIP Parameters	
Calling Party's Category	**"cpc" URI parameter in P-Asserted-Identity**	**Accept-Language**
operator, language French	operator	fr
operator, language English	operator	en
operator, language German	operator	de
operator, language Russian	operator	ru
operator, language Spanish	operator	es
ordinary calling subscriber	ordinary	-
test call	test	
payphone	payphone	
calling party's category unknown at this time (national use)	unknown	
mobile terminal located in the home PLMN	mobile-hplmn	

Table 7-9 – Mapping of the ISUP Calling Party's Category parameter to the CPC parameter

ISUP Parameter	SIP Parameters	
mobile terminal located in a visited PLMN	mobile-vplmn	
NOTE 1 – This is a national/regional specific value. Interworking shall only occur when interconnecting with indicated national network. NOTE 2 – In case the calling party''s category contains values that are not in this table then based on operator policy the "cpc" URI parameter may be omitted or may contain national/regional specific value.		

7.1.4.2 "oli" URI Parameter in P-Asserted-Identity Header

For profiles A and B: The ISUP IAM OLI parameter shall be used to set the "oli" URI parameter within tel URI or the userinfo part of SIP URI with user="phone" parameter (as defined in clause 7.2A.12 of ETSI TS 124 229 ([ITU-T Q.3403]) of a P-Asserted-Identity header in the initial INVITE request. In case the ISUP IAM OLI parameter is absent then the P-Asserted-Identity URI "oli" parameter shall be omitted from the initial INVITE request.

7.1.5 Hop Counter (Optional)

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

For profiles A and B 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 7-10, 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 BICC/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.

Table 7-10 – 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.

7.1.6 Coding of encapsulated ISUP IAM parameters in outgoing INVITE (Profile C (SIP-I) only)

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

7.1.6.1 Nature of Connection Indicators

The O-IWU shall increment the satellite indicator in the Nature of Connection Indicators parameter.

7.1.6.2 Propagation Delay Counter

The O-IWU should increase the Propagation Delay Counter parameter by an appropriate value based on available network configuration data that represents the delay of the IP network.

7.1.7 P-Early-Media header

For profiles A and B: If the Transmission Medium Requirement in the received IAM indicated a speech or video call as a network option, the O-IWU inserts a P-Early-Media header field set to "supported" in the sent INVITE request to notify that the authorization of early media is supported according [IETF RFC 5009].

7.1.8 Interworking of PSTN XML elements

For profiles A and B: If the PSTN XML schema is supported, the O-IWU shall map the Access Transport parameter, the User Service Information parameter, the User Teleservice Information and the ST digit of the Called Party Number into the relevant PSTN XML elements as described in Table 7-11.

Table 7-11 – Mapping ISUP ATP parameters into PSTN XML elements

IAM →		INVITE →
ISUP parameter	**Content**	**PSTN XML element**
Access Transport parameter	High layer compatibility	HighLayerCompatibility (Notes 1, 2, 3)
	Low layer compatibility	LowLayerCompatibility (Note 3)
User Service Information		Bearer Capability (Note 3, Note 4
User Teleservice Information	High layer characteristics identification	HighLayerCompatibility (Note 2, Note 3)
Called Party Number	Address signal "ST"	sendingCompleteIndication

NOTE 1 – If two high layer compatibility information elements are received in the ATP of the IAM, they shall be transferred in the same order as received into the PSTN XML body within the INVITE.

NOTE 2 – In the normal case, the High layer compatibility information in the ATP is equal to the High layer characteristics identification in the User Teleservice Information parameter. In the PSTN XML body, no two identical High layer compatibility information shall be present. If an HLC is available both in the ATP and in the User Teleservice information, the HLC from the ATP should be mapped.

NOTE 3 – The O-IWU shall only map this information element if the O-IWU offers media formats which can be transferred by the IM-MGW without transcoding and are derived from the incoming ISUP information according to Table 7-2.

NOTE 4 – See clause 7.1.10.

7.1.9 Progress Indicator interworking

For profiles A and B: If the PSTN XML schema is supported, the O-IWU shall map the Forward Call Indicator and the Progress Indicator Information element covered in the Access Transport parameter into the PSTN XML element included in the sent INVITE request as described in Table 7-12.

Table 7-12 – Mapping Forward Call Indicator parameter and ATP Progress Indicator into PSTN XML elements

IAM→			**INVITE →**
Forward Call Indicators parameter		**Access Transport parameter**	**PSTN XML body with ProgressIndicator with "Progress Description" value No. (Value of PI)**
ISDN User Part Indicator	ISDN Access Indicator		
0 ("ISDN User Part not used all the way")	Value non-significant	Value non-significant	No. 1 (Note 1)
1 ("ISDN User Part used all the way")	0 ("originating access non – ISDN")	Value non-significant	No. 3 (Note 1)
1 ("ISDN User Part used all the way")	1 ("originating access ISDN")	Progress indicator No. (Value of PI)	The PSTN XML ProgressIndicator mapped from Progress indicator received in the ATP (Note 2) <u>and</u> additional ProgressIndicator with "Progress Description" value No. 6 (Note 1, Note 3)
1 ("ISDN User Part used all the way")	1 ("originating access ISDN")	Not present	No. 6 (Note 1)
NOTE 1 – The ProgressIndicator "Coding Standard" parameter shall be set to "00 (ITU-T standardized coding)". The default value for the ProgressIndicator "Location" parameter is "0011 (Transit Network)". NOTE 2 – The entire Progress Indicator, including the "Progress Description", "Coding Standard" and "Location" parameters shall be copied. NOTE 3 – The order of ProgressIndicators within PSTN XML body is irrelevant.			

7.1.10 Mapping of Fallback connection type

For profiles A and B: As a network option if the PSTN XML schema is supported and the O-IWU supports forwarding fallback signalling, the following interworking applies:

When the IAM includes a TMR and a TMR Prime parameter and a USI and USI Prime parameter then the O-IWU shall interwork these parameters as follows:

7.1.10.1 Mapping of User Service Information parameter and TMR Prime

For profiles A and B: The User Service Information parameter is mapped into the **first** PSTN XML BearerCapability element appearing in the 'pstn' XML element as indicated in Table 7-13.

Table 7-13 – Mapping ISUP USI parameter into PSTN XML element

IAM →	INVITE →
User Service Information	PSTN XML element BearerCapability

The TMR Prime parameter is mapped into the 'pstn' InformationTransferCapability XML element of the **first** BearerCapability element as indicated in Table 7-14.

Table 7-14 – Mapping ISUP TMR Prime parameter into PSTN XML element

IAM →	INVITE →	
TMR Prime	**PSTN XML element BearerCapability**	
Speech	InformationTransferCapability	00000
3.1 kHz audio		10000

7.1.10.2 Mapping of User Service Information Prime parameter and TMR

For profiles A and B: The User Service Information Prime parameter is mapped into the **second** PSTN XML BearerCapability element appearing in the 'pstn' XML element as indicated in Table 7-15.

Table 7-15 – Mapping ISUP USI Prime parameter into PSTN XML element

IAM →	INVITE →
User Service Information Prime	PSTN XML element BearerCapability

The TMR parameter is mapped into the 'pstn' InformationTransferCapability XML element of the **second** BearerCapability element as indicated in Table 7-16.

Table 7-16 – Mapping ISUP TMR Prime parameter into PSTN XML element

IAM →	INVITE →	
TMR Prime	**PSTN XML element BearerCapability**	
64 kbit/s preferred	InformationTransferCapability	10001

7.1.10.3 Mapping of TMR Prime and TMR into the m-line of SDP

For profiles A and B: Insert in the SDP a single m-line of "audio". The first stated format is set according the TMR '64 kbit/s preferred' as indicated in Table 7-2.

The second stated codec is set according the TMR Prime with 'Speech' or '3.1 kHz audio' as indicated in Table 7-2.

7.1.11 Mapping of P-Access-Network-Info header

For profiles A and B: If the IAM message includes a location number ISDN User Part parameter, the O-IWU shall include a P-Access-Network-Info header. The P-Access-Network-Info shall be populated as shown in Table 7-17.

Table 7-17 – Mapping Location number parameter into P-Access-Info header

IAM→	**INVITE →**	
Location Number	**P-Access-Network-Info**	
	SIP component	**Value**
Parameter name	not mapped	
Parameter length	not mapped	
Parameter content	access-type	"GSTN"
	gstn-location (Note 4)	Set to the hexadecimal representation of the ISUP parameter content, encoded as a text string between quotes
NOTE 1 – As specified in [ITU-T Q.763], the parameter content includes both the header fields (octets 1 and 2) and the address signals. NOTE 2 – The parameter content includes the Address Presentation Restricted Indicator. This field is not mapped to the Privacy header field. It is upon network operator responsibility to remove the P-Access-Network-Info header field when leaving the trust domain. NOTE 3 – If the Screening Indicator is set to network provided, an np parameter is added to the P-Access-Network-Info header field value. NOTE 4 – Alternatively, as a network option, the value of the Location Number can populate the operator-specific-GI parameter. In this case, the operator-specific-GI is set to the text string between quotes with the sequence of digits found in octet 3 to N (except the filler) starting with the 1st digit. The access-info parameter is set to "operator-specific-GI"		

7.1.12 Coding of the INVITE when Number Portability is supported

For profiles A and B: This clause describes optional coding procedures when Number Portability is supported.

7.1.12.1 Request URI and To header coding

When Number Portability is supported, the method used for signalling of the Called Party E.164 address and the Number Portability Routing Number determines the parameters of the IAM message used to derive the Request URI of the INVITE Request.

The number portability information (rn and npdi) shall not be mapped into the To header field.

[ITU-T Q.769.1] describes three possible addressing methods for signalling of the Called Party E.164 address and Number Portability Routing Number ([ITU-T Q.769.1] uses the terms directory number and network routing number respectively). The choice of these methods is based on network operator and national requirements.

The following clauses describe how the Request URI and To header fields are populated, based on these methods, when a Number Portability Routing Number is available in the IAM.

When the optional Number Portability Routing Number is available and supported these procedures take precedence over procedures for the coding of the Request URI and To header fields described in clause 7.1.3.

When a Number Portability Routing Number is not available, the Request URI and To Header fields are populated as described in clause 7.1.3, with the following addition: If a Number Portability Forward Information parameter is present in the IAM, containing a value of "number portability query done for called number, non-ported called subscriber", a tel URI npdi parameter [IETF RFC 4694] is added.

For the following clauses, the Request URI is a tel URI or SIP URI with "user=phone" and shall contain an International public telecommunication number prefixed by a "+" sign (e.g., tel:+4991115012345).

7.1.12.2 Separate Directory Number Addressing Method

Table 7-18 – Mapping ISUP to SIP Request-URI and To header field with Number Portability Separate Directory Number Addressing Method

IAM→		INVITE
Called Party Number	**Called Directory Number**	**Request-URI and To Header Field**
Address Signal: Nature of Address Indicator: "Network routing number in national (significant) number format" or "National (significant) number" or "Network routing number in network specific number format" as described in [ITU-T Q.769.1] (Note 2)	Address Signal: Nature of Address Indicator: "National (significant) number".	The "telephone-subscriber" is populated from the Called Directory Number as follows: Insert "+CC" before the Address signals (Note 1) The Tel URI rn= parameter is populated from the Called Party Number as follows: Insert "+CC" before the Address signals (Note 1) and is added only to the Request-URI. Use of the local form of the rn= parameter is out of the scope of this Recommendation. Tel URI npdi parameter as defined in [IETF RFC 4694] is added only to the Request-URI.
NOTE 1 – CC = Country Code of the network in which the O-IWU is located. NOTE 2 – If the address signals received in the ISUP Called Party Number contain a sending terminated signal (hexadecimal digit F), then this shall be discarded or if the PSTN XML is supported then the sendingCompleteIndication shall be included.		

7.1.12.3 Concatenated Addressing Method

Table 7-19 – Mapping ISUP to SIP Request-URI and To header field with Number Portability Concatenated Number Addressing Method

IAM→	INVITE
Called Party Number	**Request-URI and To Header Field**
Address Signal: Nature of Address Indicator: "Network routing number concatenated with called directory number" or "National (significant) number" as described in [ITU-T Q.769.1] (Note 2)	The "telephone-subscriber" is populated from the Called Party Number as follows: Remove the prefix representing the Number Portability Routing Number or the prefix prior to the directory number (Note 3). Insert "+CC" before the Address signals (Note 1). The Tel URI rn= parameter is populated from the Called Party Number as follows and is added only to the Request-URI: Use all address digits contained within the Called Party Number or remove the digits that follow the prefix representing the Number Portability Routing Number Insert "+CC" before the Address signals (Note 1) Use of the local form of the rn= parameter is out of the scope of this Recommendation. Tel URI npdi parameter as defined in [IETF RFC 4694] is added only to the Request-URI.
NOTE 1 – CC = Country Code of the network in which the O-IWU is located. NOTE 2 – If the address signals received in the ISUP Called Party Number contain a sending terminated signal (hexadecimal digit F), then this shall be discarded or if the PSTN XML is supported then the sendingCompleteIndication shall be included. NOTE 3 – Based on national policy the whole Number Portability Routing number includes the Called Party Number and a prefix. In such cases only the Prefix has to be removed. Normally the Nature of Address Indicator indicates if the Number Portability Routing Number contains a Called Party Number and a prefix.	

7.1.12.4 Separate Network Routing Number Addressing Method

Table 7-20 – Mapping ISUP to SIP Request-URI and To header field with Number Portability Separate Network Routing Number Addressing Method

<table>
<tr><th colspan="2">IAM→</th><th>INVITE</th></tr>
<tr><th>Network Routing Number</th><th>Called Party Number</th><th>Request-URI and To Header Field</th></tr>
<tr><td>Address Signal:
Nature of Address Indicator:
"Network routing number in national (significant) number format" or "Network routing number in network specific number format" as described in [ITU-T Q.769.1]
(Note 2)</td><td>Address Signal:
Nature of Address Indicator:
"National (significant) number".</td><td>The "telephone-subscriber" is populated from the Called Party Number as follows:
Insert "+CC" before the Address signals (Note 1)
The Tel URI rn= parameter is populated from the Network Routing Number as follows and is added only to the Request-URI:
Insert "+CC" before the Address signals (Note 1)
Use of the local form of the rn= parameter is out of the scope of this Recommendation.
Tel URI npdi parameter as defined in [IETF RFC 4694] is added only to the Request-URI.</td></tr>
<tr><td colspan="3">NOTE 1 – CC = Country Code of the network in which the O-IWU is located.
NOTE 2 – If the address signals received in the ISUP Called Party Number contain a sending terminated signal (hexadecimal digit F), then this shall be discarded or if the PSTN XML is supported then the sendingCompleteIndication shall be included.</td></tr>
</table>

7.1.12.5 Coding of the INVITE for Carrier Routing

This clause describes optional coding procedures for carrier-based routing.

7.1.12.5.1 Mapping of "cic" in REQUEST URI Header

The procedures followed in clause 7.1.3 apply with the following addition.

If the Transit Network Selection parameter, defined according to [ITU-T Q.761], is included in the IAM message the O-IWU, based on network configuration, may send the transit network selection information to the SIP network. In such a case the "cic=" parameter as defined in [IETF RFC 4694] is included in the SIP-Request URI and configured according to the table below.

Table 7-21 – Mapping of ISUP "Transit Network Selection" (TNS) to SIP "Carrier Identification Code" (CIC)

ISUP parameter/field	Value	SIP Component	Value
Transit Network Selection	Digits	Carrier id code in Userinfo of Request URI	"cic=carrier ID code" as defined in [IETF RFC 4694]

7.1.13 PSAP Call-back indication

For profiles A and B: If the O-IWU based on the operator policy has determined that the received IAM message is for the purpose of a PSAP callback, then the O-IWU shall include in the initial INVITE request a Priority header field with a "psap-callback" header field value as specified in [IETF RFC 7090]. The operator policy decision may be based on the PSAP callback indication in the received IAM message and/or any other information made available at the O-IWU.

NOTE – The PSAP callback indication in the received IAM message depends on the national regulatory requirements applicable for the emergency services (e.g., calling party's category parameter indicating priority/emergency call, predefined prefix in front of the called party number) and is outside the scope of this Recommendation.

7.2 Receipt of SAM after INVITE has been sent

If *en bloc* addressing is used towards the SIP network, subsequent SAMs received after the O-IWU has sent the INVITE are ignored.

7.2.1 Overlap procedures upon receipt of SAM

On receipt of a SAM from the BICC/ISUP procedures running at the incoming side of the O-IWU, the O-IWU shall:

1) Stop timer T_{OIW3} (if it is running).

2) T_{OIW2} shall be restarted and the O-IWU shall invoke the appropriate outgoing signalling procedure A, B, C or D as described in clause 7.1, with the following additional procedures:

   a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call.

   b) An SIP INVITE request with incomplete address information will be rejected with a SIP 404 or 484 error response.

   c) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. In the Profile C (SIP-I) case, the IAM which was sent with the original INVITE is also encapsulated in the new INVITE.

   d) The new INVITE shall contain a new SDP offer. The O-IWU may reuse any resources that have already been reserved for this call. This reuse of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question.

   e) All other contents of the new INVITE are interworked from the parameters of the original IAM as per clause 7.1.

If timer T_{OIW2} has expired, subsequent SAMs received after the O-IWU has sent the INVITE are ignored.

7.3 Receipt of 18X response

Table 7-22 provides a summary of the interworking of 18X messages to ISUP messages. For further details please see the reference clause given in each table row.

Table 7-22 – Receipt of 18X response

← ISUP message	← 18X response
ACM or CPG (Note 1)	180 Ringing
ACM or CPG (Note 2) for Profile C (SIP-I) only	183 Session Progress with encapsulated ACM or CPG
NOTE 1 – See clause 7.3.1. NOTE 2 – See clause 7.3.2.	

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

7.3.1 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 O-IWU shall send either the ACM or CPG message as determined by BICC/ISUP procedures related to whether or not an ACM has previously been sent for this call.

For profiles A and B: Based on local knowledge that the call is transited to a PSTN network, the O-IWU may decide not to generate the awaiting answer indication when receiving the 180 Ringing message and backward early media is not authorized according to [IETF RFC 5009].

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

7.3.1.1 Setting for ACM Backward Call Indicators (mandatory) (profiles A and B only)

The table within this clause presents the default values of Backward Call Indicators parameter that are set by the O-IWU when ACM is sent. Other values of the Backward Call Indicators parameter are set according to BICC/ISUP procedures.

The indicators of the BCI parameter, which are set by the O-IWU, are as follows:

Bits	Indicators in BCI parameter
AB	Charge Indicator
DC	Called Party's Status Indicator
FE	Called Party's Category Indicator
HG	End-to-end Method Indicator
I	Interworking Indicator
J	End-to-end Information Indicator
K	ISDN User Part/BICC Indicator
L	Holding Indicator (national use)
M	ISDN Access Indicator
N	Echo Control Device Indicator

For profiles A and B, Called Party's Status Indicator (Bit DC) is set to "*subscriber free*".

For profiles A and B, the default settings are shown in Table 7-23.

Table 7-23 –Backward Call Indicators settings

Parameter	Bits	Codes	Meaning
Charge Indicator	AB	1 0	*"charge"*
Called Party's Category Indicator	FE	0 0	*"no indication"*
End-to-end Method Indicator	HG	0 0	*"no end-to-end method available"*
Interworking Indicator	I	1	*"interworking encountered"*
		0	*"no interworking encountered"* Note 1
End-to-end Information Indicator	J	0	*"no end-to-end information available"*
ISDN User part/BICC Indicator	K	0	*"ISDN user part/BICC not used all the way"*
		1	"ISDN user part/BICC used all the way" Note 1
Holding Indicator (national use)	L	0	*"holding not requested"*
ISDN Access Indicator	M	0	*"terminating access non-ISDN"*
		1	"terminating access ISDN" Note 1
Echo Control Device Indicator	N	1	*"incoming echo control device included"* Note 2
		0	*"incoming echo control device not included"* Note 3
NOTE 1 – As a network operator option, the value is used for TMR = 64 kbit/s unrestricted. This avoids the sending of a Progress Indicator with progress information 0 0 0 0 0 1 0 "Destination access is non-ISDN", so the call will not be released for that reason by an ISDN terminal. NOTE 2 – For speech calls, e.g., TMR is "3.1KHz audio". NOTE 3 – For known data calls, e.g., TMR "64 kbit/s unrestricted" or HLC "Facsimile Group 2/3".			

When a 180 (Ringing) response is received with the Alert-Info header field at an O-IWU supporting capabilities associated with the Alert-Info header field an O-IWU may instruct the MGW to play out early media available at the associated URL to the PSTN leg of the communication.

7.3.1.1.1 Interworking of PSTN XML body

For profiles A and B: If the PSTN XML body is supported as a network option, the Backward Call Indicators parameters derived as shown in Table 7-24 shall take precedence over the above Backward Call Indicators parameter setting.

Table 7-24 – Derivation of Backward Call Indicators from PSTN XML body

←ACM	← 180 Ringing or 183 Session Progress
Backward Call Indicators parameter **Optional Backward Call Indicators parameter**	**PSTN XML body with Progress Indicator with "Coding Standard" value "00 (ITU-T standardized coding)" and with "Progress Description" No (Value of PI)**
Backward Call Indicators parameter ISDN User Part Indicator 0 "ISDN User Part not used all the way"	No. 1 ("Call is not end-to-end ISDN: further call progress information may be available in-band")

Table 7-24 – Derivation of Backward Call Indicators from PSTN XML body

←ACM	← 180 Ringing or 183 Session Progress
Backward Call Indicators parameter ISDN User Part Indicator 0 "ISDN User Part used all the way" ISDN Access Indicator 0 "Terminating access non-ISDN"	No. 2 ("Destination address is non-ISDN")
Backward Call Indicators parameter Interworking Indicator 0 "no interworking encountered" ISDN User Part Indicator 1 "ISDN User Part used all the way" ISDN Access Indicator 1 "Terminating access ISDN"	No. 7 ("Terminating user ISDN")
Optional Backward Call Indicators parameter In-band Information Indicator 1 "in-band information or an appropriate pattern is now available"	No. 8 ("In-band information or an appropriate pattern is now available")

7.3.1.2 Settings for Event Information (mandatory) in CPG (profiles A and B only)

The table within this clause 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 parameter are set according to BICC/ISUP procedures.

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

The code in Table 7-25 shall be set by the O-IWU in the Event Information parameter on receipt of 180 Ringing.

Table 7-25 – Coding of Event Indicator for profiles A and B

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

For profiles A and B: If the 180 Ringing does not contain a P-Early-Media header or the P-Early-Media header field does not authorize the backward early media, the O-IWU instructs the MGM to generate the awaiting answer indication. The coding of the Event Indicator is shown in Table 7-25.

If the 180 Ringing contains a P-Early-Media header authorize backward early media, the O-IWU instructs the MGW to pass through the media path.

Based on local knowledge that the call is transited to a PSTN network, the O-IWU may decide not to generate the awaiting answer indication when receiving the 180 Ringing message and backward early media is not authorized according to [IETF RFC 5009].

If a Backward Call Indicator is present, the setting is shown in Table 7-23. The Called Party's Status Indicator (Bit DC) is set to "*subscriber free*".

7.3.2 Receipt of 183 Session Progress

For profiles A and B: If 183 Session Progress is received without any encapsulated ISUP message, on receipt of the first 183 Session Progress that includes a P-Early-Media header field authorizing backward early media an ACM is sent. The interworking as described in Table 7-23 applies with exception that the Called Party's Status Indicator is set to *"no indication"*. In addition an Optional Backward Call Indicators parameter is present set to *"In-band information or an appropriate pattern is now available"*. Bit A = 1 "in-band information or an appropriate pattern is now available" shall be set if 183 Session Progress response is received and according to [IETF RFC 5009] backward early media is authorized.

Table 7-26 – Sending criteria of Optional Backward Call Indicators parameter

←ACM	← 183 Session Progress
Optional Backward Call Indicators parameter In-band Information Indicator "in-band information or an appropriate pattern is now available"	P-Early-Media header authorizing backward early media

For profiles A and B: As a network option if the PSTN XML body is supported the interworking according to Table 7-24 apply.

For profiles A and B: If the Address Complete Message (ACM) has already been sent, the O-IWU shall send the Call Progress message (CPG) if 183 Session Progress includes the first P-Early-Media header field authorizing backward early media. The coding of the Event Indicator is shown in Table 7-27.

Table 7-27 – Coding of Event Indicator

Bits	Codes	Meaning
G F E D C B A	0 0 0 0 0 1 1	*"in-band information or an appropriate pattern is now available"*

The O-IWU instructs the MGW to pass through the media path.

For profiles A and B: The Backward Call Indicators parameter is optional in the CPG message and shall only be included if any indicators have changed from those previously sent. If a Backward Call indicator is present and if the PSTN XML body is supported as a network option, the Backward Call indicators parameters are derived as shown in Table 7-24 shall take precedence as described in Table 7-23. The Called Party's Status Indicator (Bit DC) is set to *"no indication*".

For profiles A and B: As a network option, reception of 183 containing a SIP reason header with a Q.850 Cause Value an Cause parameter is present, the cause value derived from the Reason header. The coding of the Event Indicator is shown in Table 7-27.

If no P-Early-Media header or no Reason header is present, no BICC/ISUP message is sent backward and BICC/ISUP procedures should continue.

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

7.3.3 Receipt of 181 Call is Being Forwarded (Profiles A and B only)

On receipt of the first 181 Call is Being Forwarded backward early media is not authorized, an ACM is sent and the Call Diversion Information parameter is present. The Backward Call indicator is set according to Table 7-23.

If the 181 Call is Being Forwarded contains a P-Early-Media header authorizes backward early media, an ACM is sent and the Call Diversion Information parameter and an Optional Backward Call indicator set to ("In-band information or an appropriate pattern is now available" is present as shown in Table 7-29.

If the Address Complete Message (ACM) has already been sent and the 181 Call is Being Forwarded contains a P-Early-Media header authorizes backward early media O-IWU shall send the Call Progress message (CPG). The coding of the Event Indicator is shown in Table 7-28.

Table 7-28 – Coding of Event Indicator

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

An Optional Backward Call indicator is set to: Bit A = 1 "in-band information or an appropriate pattern is now available" shall be set if 181 Call is Being Forwarded response is received and according to [IETF RFC 5009] the backward early media is authorized.

Table 7-29 – Sending criteria of Optional Backward Call Indicators parameter

←ACM/CPG	← 181 Call is Being Forwarded
Optional Backward Call Indicators parameter In-band Information Indicator in-band information or an appropriate pattern is now available	P-Early-Media header authorizing backward early media

The Backward Call Indicators parameter is optional in the CPG message and shall only be included if any indicators have changed from those previously sent. If a Backward Call Indicator is present the indicator values are set according to Table 7-23.

The O-IWU instructs the MGW to pass through the media path.

7.3.4 Handling of P-Early-Media header

For profiles A and B: If the O-IWU receives a 18x response with a P-Early-Media header field that changes the authorization of early media, the O-IWU terminates the sending of the awaiting answer indication if the header field authorizes backward early media, and initiates the sending of the awaiting answer indication if the header field removes authorization of backward early media and if the O-IWU has received the 180 Ringing response.

7.3.5 Access Transport parameter, Transmission Medium Used parameter

For profiles A and B: If the O-IWU supports the PSTN XML body as a network option and if a PSTN XML body is received within the 180 ringing or 183 session progress, the O-IWU shall store the received PSTN XML elements, replacing any previously stored PSTN XML elements on that dialogue.

NOTE – Multiple 18x responses can be received, both within a single dialogue and in multiple dialogues. The PSTN XML bodies are stored on a per-dialogue basis to be mapped to the ATP/TMU parameters on receipt of the 200 OK (see clause 7.5.1.2).

7.3.6 Progress Indicator

For profiles A and B: If the O-IWU supports the PSTN XML body as a network option and receives it in the 180 or 183, the O-IWU shall store a "ProgressIndicator" element from the PSTN XML body on a per dialogue basis and shall add additionally map it into a Progress Indicator in the ACM as shown in Table 7-30.

Table 7-30 – Mapping of the Progress Indicator in PSTN XML body into ATP in the ACM

←ACM	←180/183
Access Transport parameter	**PSTN XML body with Progress Indicator with "Coding Standard" value "00 (ITU-T standardized coding)" and with "Progress Description" No. (Value of PI)**
Progress Indicator (Note)	Progress Indicator No. 1 / 2
Progress Indicator (Note)	Progress Indicator No. 8
NOTE – The entire Progress Indicator, including the "Progress Description", "Coding Standard" and "Location" parameters shall be copied.	

For profiles A and B: If the O-IWU supports the PSTN XML body as a network option and receives it in the 180 or 183, any "ProgressIndicator" element in the PSTN XML body shall be stored on a per-dialogue basis as well as mapped as shown in Tables 7-31 and 7-32.

Table 7-31 – Mapping of the Progress Indicator in PSTN XML body into ATP in the CPG

←CPG	←180/183
Access Transport parameter	**PSTN XML body with Progress Indicator X**
Progress Indicator (Note 1, Note 3)	Progress Indicator No. 1 / 2
Progress Indicator (Note 2, Note 3)	Progress Indicator No. 4
Progress Indicator No. 4 (Note 2, Note 4)	Progress Indicator No. 7
NOTE 1 – Values 1 ("call is not end-to-end ISDN: further call progress information may be available in-band") or 2 ("destination address is non-ISDN") shall be sent if Value 4 ("Call has returned to the ISDN") has been sent since value 1 or 2 was previously sent or if no value 1 or 2 was previously sent. NOTE 2 – Value 4 ("Call has returned to the ISDN") shall be sent if value 1 ("call is not end-to-end ISDN: further call progress information may be available in-band") or 2 ("destination address is non-ISDN") was sent previously and no value 4 has been signalled since. NOTE 3 – The entire Progress Indicator, including the "Progress Description", "Coding Standard" and "Location" parameters shall be copied. NOTE 4 – The Progress Indicator "Coding Standard" parameter shall be set to "00 (ITU-T standardized coding)". The default value for the Progress Indicator "Location" parameter is "0100 (Public Network serving remote user)".	

Table 7-32 – Mapping of Progress Indicator in PSTN XML body into Event Indicator

←CPG	←180/183
Event Indicator	**PSTN XML body with Progress Indicator with "Coding Standard" value "00 (ITU-T standardized coding)" and with Progress Description" value No. X**
"In-band information or appropriate pattern is now available"	No. 8 "In-band information or appropriate pattern is now available"

7.4 Expiry of timers and sending of early ACM

When either timer T_{OIW1} (in the case of calls converted to *en bloc* at the outgoing SIP interface) or timer T_{OIW2} expires, the O-IWU shall return ACM. In the case that the continuity check is performed (ISUP) or COT is expected (BICC), the O-IWU shall withhold sending ACM until a successful continuity indication has been received. For Profile A, 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 indicators of the BCI parameter shall be set as described in clause 7.3.1.1.

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) and the O-IWU shall send an Answer Message (ANM) or Connect message (CON) as determined by BICC/ISUP procedures. The O-IWU shall stop any existing awaiting answer indication (e.g., ringing tone).

The O-IWU shall not progress any further early dialogues to established dialogues. Therefore, upon receipt of a subsequent final 200 (OK) response for any further dialogue for an INVITE request (e.g., due to forking), the O-IWU shall:

1) acknowledge the response with an ACK request; and

2) send a BYE request to this dialogue in order to terminate it.

For Profile C (SIP-I), if 200 OK INVITE is received with encapsulated CON or ANM message, the O-IWU shall determine the appropriate backward BICC/ISUP message and parameters based on the encapsulated ISUP message and existing BICC/ISUP signalling state.

7.5.1 Coding of the ANM

Upon receipt of the first 200 OK (INVITE), if the Address Complete Message (ACM) has already been sent, the O-IWU shall send the Answer Message (ANM) to the preceding exchange.

For profiles A and B: If Backwards Call Indicators are included in the ANM, then the coding of these parameters shall be as described in clause 7.3.1.1. The Backward Call Indicators parameter is optional in the ANM message and shall only be included if any indicators have changed from those previously sent.

7.5.1.1 Access Transport parameter

For profiles A and B: If the O-IWU supports the PSTN XML body as a network option and if a PSTN XML body is received within the 200 OK (INVITE) or has been previously stored from a 18x message, the O-IWU shall map the most recently received information for the established dialogue (i.e., the dialogue for which the first 200 OK has been received) into the ANM as shown in Table 7-33 except Progress Indicator value No. 3 or No. 8.

Table 7-33 – Mapping of PSTN XML elements into ISUP parameters

← ANM/CON		← 200 OK / stored information from previous 18x
ISUP Parameter	**Content**	**PSTN XML (Note 9)**
Access Transport parameter	Progress indicator (Note 5, Note 8)	ProgressIndicator with "Coding Standard" value "00 (ITU-T standardized coding)" and with "Progress Description" value No. 1 / 2
	Progress indicator with "Progress Description" value No. 4 (Note 4, Note 6, Note 9)	ProgressIndicator with "Coding Standard" value "00 (ITU-T standardized coding)" and with "Progress Description" value No. 7
	Progress indicator (Note 6, Note 8)	ProgressIndicator with "Coding Standard" value "00 (ITU-T standardized coding)" and with "Progress Description" value No. 4
	Progress indicator (Note 7, Note 8)	ProgressIndicator with "Coding Standard" value "00 (ITU-T standardized coding)" and with "Progress Description" value No. 5
	High layer compatibility (Note 1)	HighLayerCompatibility
	Bearer Capability	BearerCapability (Note 2)
	Bearer Capability ("UDI-TA")	BearerCapability ("UDI-TA") (Note 3)

Table 7-33 – Mapping of PSTN XML elements into ISUP parameters

<table>
<tr><th colspan="2">← ANM/CON</th><th>← 200 OK / stored information from previous 18x</th></tr>
<tr><th>ISUP Parameter</th><th>Content</th><th>PSTN XML (Note 9)</th></tr>
<tr><td colspan="3">NOTE 1 – This information element shall only be mapped if the O-IWU transfers media types listed in Table 7-4 without transcoding.
NOTE 2 – Applicable if the O-IWU has not propagated UDI fallback signalling according to clause 7.1.10.
NOTE 3 – Applicable if the O-IWU has propagated UDI fallback signalling according to clause 7.1.10. Only the value "UDI-TA" within the PSTN XML BC shall be mapped. Other values within the PSTN XML BC are mapped to TMU as described in clause 7.3.5.
NOTE 4 – ProgressIndicator No. 7 is not mapped into the ISUP ATP. However, it may be mapped into PI=4.
NOTE 5 – Values 1 ("call is not end-to-end ISDN: further call progress information may be available in-band") or 2 ("destination address is non-ISDN") shall be sent if Value 4 ("Call has returned to the ISDN") has been sent since value 1 or 2 was previously sent or if no value 1 or 2 was previously sent.
NOTE 6 – Value 4 ("Call has returned to the ISDN") shall be sent if value 1 ("call is not end-to-end ISDN: further call progress information may be available in-band") or 2 ("destination address is non-ISDN") was sent previously and no value 4 has been signalled since.
NOTE 7 – This value indicates a bearer service change and is present with an associated BearerCapability and indicates that fallback has occurred (i.e., TMR and TMR Prime present in IAM and the destination ISDN user has accepted the BearerCapability equating to TMR Prime).
NOTE 8 – The entire Progress Indicator, including the "Progress Description", "Coding Standard" and "Location" parameters shall be copied.
NOTE 9 – The Progress Indicator "Coding Standard" parameter shall be set to "00 (ITU-T standardized coding)". The default value for the Progress Indicator "Location" parameter is "0100 (Public Network serving remote user)".</td></tr>
</table>

7.5.1.2 Transmission Medium Used parameter (TMU)

For profiles A and B: The procedures in the present clause shall only apply if the O-IWU supports the PSTN XML body, and has propagated UDI-TA fallback signalling according to clause 7.1.10.

If a Bearer Capability element within a PSTN XML body is received within the first 200 OK(INVITE) or has been previously stored from a 18x message relating to the now established dialogue (i.e., the dialogue for which the first 200 OK has been received), the O-IWU shall map the most recently received information (if any) into a TMU within the ANM as shown in Table 7-34. If the most recently received PSTN XML BearerCapability is "UDI-TA", it shall be mapped into an ISUP Access Transport Parameter Bearer Capability (see clause 7.5.1.1).

NOTE – The TMU is only included if both the TMR and TMR Prime were received in the ISUP IAM and fallback has occurred in the succeeding network.

Table 7-34 – Mapping to TMU parameter

← ANM/CON	← 200 OK / stored information from previous 18x
TMU	**PSTN XML BearerCapability**
TMU = "Speech"	PSTN XML BearerCapability = "Speech"
TMU= "3.1 kHz audio"	PSTN XML BearerCapability = "3.1 kHz audio"

Table 7-34 – Mapping to TMU parameter

← ANM/CON	← 200 OK / stored information from previous 18x
No mapping (fallback has not occurred)	PSTN XML BearerCapability = "UDI-TA"
TMU = "3.1 kHz audio"	PSTN XML BearerCapability not present

7.5.2 Setting of Backward Call Indicators in the CON message (profiles A and B only)

The Called Party's Status Indicator (Bit DC) of BCI parameter is set to *"no indication"*. The other indicators of the BCI parameter shall be set as described in clause 7.3.1.1.

7.5.2.1 Access Transport parameter

For profiles A and B: The O-IWU shall apply the same procedure as described for the ANM in clause 7.5.1.1

7.5.2.2 Transmission Medium Used parameter

For profiles A and B: The O-IWU shall apply the same procedure as described for the ANM in clause 7.5.1.2

7.5.3 Receipt of a reINVITE request

For profiles A and B: When a reINVITE request is received from the network containing a Call-Info header field the O-IWU may instruct the MGW to send media available at the associated URL to the PSTN leg of the communication.

7.6 Through connection of BICC/ISUP bearer path

Through connection of bearer path is applicable to Type 1 or Type 3 gateway only.

For profiles A and B, through connection at the O-IWU shall follow the ITU-T Q.764 procedures for the destination exchange if this functionality is not available at the ASN. If the ASN supports the ITU-T Q.764 procedures for through connection at a destination exchange, the O-IWU shall follow the procedures specified for Profile C (SIP-I).

For Profile C (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 set-up 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 profiles A and B, the following conditions result in ringing tone being played from the O-IWU:

1) 180 Ringing received and the P-Early-Media header is not present (Profile A and B); and
2) ISUP procedures indicate that ringing tone can be applied; and
3) the local arrangements assign the role of destination exchange to the O-IWU rather than the associated SIP entity.

NOTE 1 – It is possible that ringing tone or a progress announcement is already being played as a result of T_{OIW1} or T_{OIW2} expiry. See clause 7.4.

NOTE 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 a BICC or ISUP REL message:

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: The O-IWU shall hold the REL message until a SIP response has been received. At that point, it shall take action 3 or 4 as appropriate.

3) 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.

4) 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 an early dialogue only, CANCEL shall be used instead.

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

A Reason header field containing the received (Q.850) Cause Value of the REL message shall be added to the CANCEL or BYE request. The mapping of the Cause Indicators parameter to the Reason header is shown in Table 6-49 (see clause 6.11.2).

For profiles A and B: If the O-IWU supports the PSTN XML body as a network option, the O-IWU shall map the contained information into a PSTN XML body within the BYE or CANCEL as shown in Table 6-51.

7.7.2 Receipt of backward BYE

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

On receipt of SIP BYE, the O-IWU shall invoke the BICC Release sending procedure ([ITU-T Q.1902.4]) on the BICC side.

In the case of Profile C (SIP-I), the encapsulated REL shall be passed to ISUP/BICC procedures without modification.

For Profile A or B

If a Reason header field with Q.850 Cause Value is included in the BYE, then the Cause Value shall be mapped to the ISUP Cause Value field in the ISUP REL. The mapping of the Reason header to the Cause Indicators parameter is shown in Table 6-46 (see clause 6.11.1). Table 7-35 shows the coding of the Cause Value in the REL message if it is not available from the Reason header field.

Table 7-35 – Release from SIP side at O-IWU

←REL Cause Indicators parameter	←SIP message
Cause Value No. 16 ("*normal call clearing*")	BYE

If the O-IWU supports the PSTN XML body as a network option and if a PSTN XML body is received within the BYE, the O-IWU shall map the contained information into the Access Transport parameter of the REL as shown in clause 7.5.1.1.

7.7.3 Autonomous release at O-IWU

Table 7-36 shows the trigger events at the O-IWU and the release initiated by the O-IWU when the call is traversing from BICC/ISUP to SIP.

If, after answer, BICC/ISUP procedures result in autonomous REL message from the O-IWU then a BYE shall be sent on the SIP side if the ACK has been sent before. A CANCEL method shall be sent before 200 OK (INVITE) has been received.

A Reason header field containing the (Q.850) Cause Value of the REL message sent by the O-IWU shall be added to the SIP Message (BYE or CANCEL) to be sent by the SIP side of the O-IWU.

NOTE – The IWU shall send the ACK method before it sends the BYE, if 200 OK (INVITE) is received.

Table 7-36 – Autonomous Release at O-IWU

REL ← **Cause Indicators parameter**	**Trigger event**	**→ SIP**
As determined by BICC/ISUP procedure.	COT received with the Continuity Indicators parameter set to "*continuity check failed*" (ISUP only).	Send CANCEL or BYE according to the rule described in clause 7.7.1. Note 1
	The BICC/ISUP timer T8 expires.	Send CANCEL according to the rule described in clause 7.7.1.
REL with cause value 47 (resource unavailable, unspecified).	Internal resource reservation unsuccessful	As determined by SIP procedure
As determined by BICC/ISUP procedure.	BICC/ISUP procedures result in generation of autonomous REL on BICC/ISUP side.	CANCEL or BYE according to the rule described in clause 7.7.1.
Depending on the SIP release reason.	SIP procedures result in a decision to release the call.	As determined by SIP procedure.
NOTE 1 – A Reason header field containing the (Q.850) Cause Value 41 Temporary Failure shall be added to the CANCEL request to be sent by the SIP side of the O-IWU.		

7.7.4 Receipt of RSC, GRS or CGB (ISUP)

Table 7-37 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*".

– For the RSC message, the circuit identified by the CIC is affected.

  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.

– For the GRS message, the affected circuits are identified by the CIC and the Range subfield of the Range and Status parameter.

– For the CGB message, the affected circuits are identified by the CIC and the Range and Status parameter.

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

A Reason header field containing the (Q.850) Cause Value of the REL message sent by the O-IWU shall be added to the SIP message (BYE or CANCEL) to be sent by the SIP side of the O-IWU.

In the Profile C (SIP-I) case the RSC, GRS or CGB ISUP messages shall not be encapsulated, but if a BYE request is sent, it shall encapsulate the REL message that would be sent towards a forward ISUP node.

Table 7-37 – 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 RSC or GRS (BICC)

Table 7-38 shows the message sent by the O-IWU upon receipt of a BICC RSC message or GRS message.

– For the RSC message, the circuit identified by the CIC is affected.

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

– For the GRS message, the affected circuits are identified by the CIC and the Range subfield of the Range and Status parameter.

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

A Reason header field containing the (Q.850) Cause Value of the REL message sent by the O-IWU shall be added to the SIP message (BYE or CANCEL) to be sent by the SIP side of the O-IWU. In the Profile C (SIP-I) case the RSC or GRS messages shall not be encapsulated, but if a BYE request is sent, it shall encapsulate the REL message that would be sent towards a forward ISUP node.

Table 7-38 – Receipt of RSC or GRS (BICC) at O-IWU

Message received from BICC →	SIP →
Reset Circuit/CIC message (RSC)	CANCEL or BYE
Circuit/CIC Group Reset message (GRS)	CANCEL or BYE

7.7.6 Receipt of 4XX, 5XX, 6XX responses to INVITE

If a Reason header is included in a 4XX, 5XX, 6XX, then the Cause Value of the Reason header shall 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 6-46 (see clause 6.11.1). 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 7-39.

For Profile C, if an encapsulated REL is received it shall be passed to BICC/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 clause specify additional SIP side behaviour 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 BICC/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 BICC/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 behaviour of the O-IWU is described within the referred to section. However, the table indicates the "eventual" behaviour 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.

Table 7-39 – Receipt of 4XX, 5XX or 6XX at O-IWU

← REL (Cause Value)	← 4XX/5XX/6XX SIP message	Remarks
111 (Protocol error, unspecified)	400 Bad Request	
127 Interworking	401 Unauthorized	(Note 1)
127 Interworking	402 Payment Required	
79 (Service or option not implemented, unspecified)	403 Forbidden	
1 Unallocated number	404 Not Found	
127 Interworking	405 Method Not Allowed	
127 Interworking	406 Not Acceptable	
127 Interworking	407 Proxy authentication required	(Note 1)
102 Recovery on timer expiry	408 Request Timeout	
22 Number changed (without diagnostic)	410 Gone	
127 Interworking	413 Request Entity too long	(Note 1)
111 Protocol error, unspecified	414 Request-uri too long	(Note 1)
127 Interworking	415 Unsupported Media type	(Note 1)
111 Protocol error, unspecified	416 Unsupported URI scheme	(Note 1)
79 Service or option not implemented, unspecified	417 Unknown Resource-Priority	
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	423 Interval Too Brief	
21 Call rejected	433 Anonymity Disallowed	(Note 4)
127 Interworking, unspecified	440 Max-Breadth Exceeded	
20 Subscriber absent	480 Temporarily Unavailable	
127 Interworking	481 Call/Transaction does not exist	
127 Interworking	482 Loop Detected	

Table 7-39 – Receipt of 4XX, 5XX or 6XX at O-IWU

← REL (Cause Value)	← 4XX/5XX/6XX SIP message	Remarks
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 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	493 Undecipherable	
127 Interworking	500 Server Internal error	
79 Service or option not implemented, unspecified	501 Not implemented	
27 Destination out of order	502 Bad Gateway	
41 Temporary failure	503 Service Unavailable	(Note 1)
102 Recovery on timer expiry	504 Server timeout	
127 Interworking	505 Version not supported	(Note 1)
95 Invalid message, unspecified	513 Message too large	(Note 1)
127 Interworking	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. NOTE 2 – This response does not terminate a SIP dialogue, but only a specific transaction within it. NOTE 3 – No mapping if the O-IWU previously issued a CANCEL request for the INVITE. NOTE 4 – Anonymity Disallowed, [IETF RFC 5079] refers.		

For profiles A and B: If the O-IWU supports the PSTN XML body as a network option and if a PSTN XML body is received within the 4xx/5xx/6xx, the O-IWU shall map the contained information into the Access Transport parameter of the REL as shown in clause 7.5.1.1.

For profiles A and B: When a 4xx, 5xx or 6xx SIP response to an INVITE request is received from the network containing an Error-Info header field, the O-IWU, supporting the capabilities associated with the Error-Info header field, may instruct the MGW to play out media available at the associated URL towards PSTN.

7.7.6.1 Special handling of 484 Address Incomplete response when T_{OIW3} is in use

On receipt of a 484 Address Incomplete response for the current INVITE (i.e., there are no other pending INVITE transactions for this call), if the O-IWU is configured to propagate overlap signalling into the SIP network, the O-IWU shall not send a REL message immediately and shall instead start timer T_{OIW3}. The REL message shall only be sent if T_{OIW3} expires. If the O-IWU is not configured to propagate overlap signalling into the SIP network, then the timer shall not be started and the REL shall be sent immediately to the BICC/ISUP network.

At the receipt of a SAM, a SIP 1xx provisional response or a SIP 200 OK (INVITE), the O-IWU shall stop T_{OIW2} and T_{OIW3}.

The O-IWU shall send a REL message with Cause Value 28 towards the BICC/ISUP network if T_{OIW3} expires.

7.7.6.2 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.

7.7.6.2.1 580 Precondition Failure response to an INVITE

Release with Cause Value, as indicated in Table 7-39, is sent immediately to the BICC/ISUP network.

7.7.6.2.2 580 Precondition Failure response to an UPDATE within an early dialogue

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

7.7.6.3 Receipt of SIP redirect (3xx) response

For profiles A and B: When receiving a SIP response with a response code 3xx, the default behaviour of the O-IWU is to release the call with a cause code value 127 (Interworking unspecified).

NOTE – The O-IWU may also decide for example to redirect the call towards the URIs in the Contact header field of the response as an operator option, but such handling is outside of the scope of the present document.

7.7.6.4 Encoding of the INFO

For profiles A and B: Table 7-40 provides a summary of how the header fields and MIME body within the outgoing INFO messages are populated when in-dialogue SIP INFO requests are used for overlap dialling.

Table 7-40 – Interworked contents of the INFO message

SAM→	INFO→
Digits	SubsequentDigit: <digit> (Note)
	Content-Type: application/session-info Content-Disposition: signal; handling= optional
NOTE – MIME body digit(s): 0 - 9 / "A" / "B" / "C" / "D" / "E" / "F".	

7.8 Timers at O-IWU

Table 7-41 defines the interworking timers introduced in clause 7.

Table 7-41 – Interworking timers

Symbol	Timeout value	Cause for initiation	Normal termination	At expiry	Reference
T_{OIW1}	4-6 seconds (default of 4 seconds)	On receipt of an IAM or SAM after the minimum number of digits required for routing the call has been received, if the end of address signalling has not been determined.	At the receipt of fresh address information.	Send the initial INVITE, return an ACM.	7.1, 7.4 (Note 1)
T_{OIW2}	4-20 seconds (default of 4 seconds)	Sending of INVITE unless the ACM has already been sent.	On receipt of 484 Address Incomplete for the current INVITE, 180 Ringing, 183 Session Progress with encapsulated ACM, or 404 Not Found or 484 Address Incomplete for an INVITE transaction for which T_{OIW3} is running, or 200 OK INVITE	Send early ACM.	7.1, 7.2.1, 7.3.1, 7.4, 7.5 (Note 2)
T_{OIW3}	4-6 seconds (default of 4 seconds)	On receipt of 484 Address Incomplete for the current INVITE if there are no other pending INVITE transactions for this call.	At the receipt of fresh address information.	Send REL with Cause Value 28 to the BICC/ISUP side.	7.2.1, 7.7.6.1 (Note 3)
NOTE 1 – This timer is used for ISUP overlap to SIP *en bloc* conversion. NOTE 2 – This timer is used to send an early ACM if a delay is encountered in receiving a response from the subsequent SIP network. NOTE 3 – This timer is known as the "SIP dialogue protection timer". This timer is only used where the O-IWU is configured to propagate ISUP overlap signalling into the SIP network.					

Annex A

BICC specific interworking for basic call

(This annex forms an integral part of this Recommendation.)

A.1 Introduction

This annex contains additional interworking to/from SIP which are particular to the BICC protocol.

A.2 Interworking BICC to/from SIP with common media bearer technology and BICC supports "Bearer Control Tunnelling"

If both BICC and SIP networks use the same media bearer technology, there is no media intermediary and the BICC side uses bearer control tunnelling then the following procedures apply.

For BICC CS-2, the only defined Bearer Control Protocol carried by the Bearer Control Tunnelling mechanism is IPBCP [ITU-T Q.1990]. However, the procedures below apply equally to any future Bearer Control Protocol for which interworking with SDP and the SDP offer/answer procedures is defined.

A.2.1 Bearer Control Interworking

A Bearer Control Interworking function is assumed to exist which performs interworking between Bearer Control information (in the BICC Bearer Control Tunnelling information element) and SDP message bodies (in SIP messages). For IPBCP, the procedures for this interworking function are defined in clause A.3.1.

A.2.1.1 Interworking from SDP offers to BICC Bearer Control Tunnelling information

On receipt of a SIP message containing an SDP offer, the Bearer Control Interworking function is used to generate a Bearer Control Protocol Data Unit for inclusion in a BICC message. The particular BICC message used depends on the procedures defined below.

The procedures of [IETF RFC 3264] and [IETF RFC 3261] are used to determine the SIP message that should contain the SDP answer corresponding to this offer. Sending of this message is delayed until a BICC message has been received containing a Bearer Control Protocol Data Unit as described in clause A.2.1.3.

A.2.1.2 Interworking from SDP answers to BICC Bearer Control Tunnelling information

On receipt of a SIP message containing an SDP answer, the Bearer Control Interworking function is used to generate a Bearer Control Protocol Data Unit for inclusion in a BICC message. The particular BICC message used depends on the procedures defined below.

A.2.1.3 Interworking from BICC Bearer Control Tunnelling information to SDP

On receipt of a BICC message containing a Bearer Control Protocol Data Unit, the Bearer Control Interworking Function is used to generate an SDP offer or answer for inclusion within a SIP message.

If the SDP is an SDP offer, then the particular SIP message used depends on the procedures defined below.

If the SDP is an SDP answer, then the SIP message sent is as identified in clause A.2.1.1.

A.2.2 Message mapping procedures

A.2.2.1 SIP to BICC

A.2.2.1.1 Initial INVITE

On receipt of the INVITE, the I-IWU determines the Bearer Setup Procedure to be used on the BICC side. This depends on whether the INVITE contains an SDP offer:

If the INVITE contains an SDP offer, then the I-IWU uses the "Per call bearer set-up using bearer control tunnelling – fast forwards" procedures defined in [ITU-T Q.1902.4]. The INVITE is mapped to an IAM as described in clause 7.1.

If the INVITE does not contain an SDP offer, then the I-IWU uses the "Per call bearer set-up using bearer control tunnelling – backwards" procedures defined in [ITU-T Q.1902.4]. The INVITE is mapped to an IAM as described in clause 7.1.

A.2.2.1.2 APM

Subsequently, an APM message is received according to the ITU-T Q.1902.4 procedures. This is mapped to a SIP 183 Session Progress response to the initial INVITE.

A.2.2.1.3 PRACK

On receipt of a PRACK message, responding to the 183 Session Progress response sent in clause A.2.2.1.2, containing SDP, the I-IWU shall send an APM message on the BICC side.

A.2.2.1.4 Further APM messages

On receipt of further APM messages on the BICC side, containing Bearer Control Tunnelling information which maps to an SDP offer, the I-IWU shall send an UPDATE request on the SIP side.

A.2.2.1.5 UPDATE requests

On receipt of an UPDATE request on the SIP side, containing SDP, the I-IWU shall send an APM message on the BICC side.

A.2.2.1.6 200 OK UPDATE response

On receipt of a 200 OK UPDATE message, in response to the UPDATE request sent as a result of clause A.2.2.1.4, containing SDP, the I-IWU shall send an APM message on the BICC side.

A.2.2.2 BICC to SIP

A.2.2.2.1 Initial IAM

On receipt of an IAM, the O-IWU action depends on the Bearer Setup Procedure requested.

A.2.2.2.1.1 Fast Forwards set-up

In this case, the IAM contains Bearer Control Tunnelling information which maps to an SDP offer. An INVITE is sent containing this SDP offer.

A.2.2.2.1.2 Backwards

In this case, the IAM does not contain Bearer Control Tunnelling information. An INVITE is sent without SDP.

A.2.2.2.1.3 Delayed Forwards

In this case, the IAM does not contain Bearer Control Tunnelling information. An APM is returned according to the ITU-T Q.1902.4 procedures.

Subsequently, an APM message is received containing Bearer Control Tunnelling information, which maps to an SDP offer. An INVITE is sent containing this SDP offer.

A.2.2.2.2 Provisional response to INVITE

A provisional response to the INVITE may be received containing SDP which maps to a Bearer Control Protocol Data Unit. This is included as Bearer Control Tunnelling data within an APM message.

A.2.2.2.3 Subsequent APMs

On receipt of an APM message containing Bearer Control Tunnelling information, this information is mapped to an SDP offer or answer. In the case of an SDP offer, this is sent in an UPDATE message. In the case of an SDP answer, the procedures of clause A.2.1.3 determine the SIP message to send.

A.2.3 Preconditions

Preconditions refer to the mechanisms used to determine when bearer set-up is complete, including completion of any procedures within the bearer network not visible to the IWF.

Preconditions are handled on the SIP side using the mechanisms of [IETF RFC 3312] which are based on attributes within the SDP.

Preconditions are handled on the BICC side using the continuity mechanism as described in [ITU-T Q.1902.4] to delay continuation of call set-up until all preconditions to call set-up have been met.

Note that BICC provides mechanisms to indicate the existence and completion of preconditions from the O-ISN to the T-ISN, but not in the reverse direction: it is assumed that there are no (pre-ACM) procedures at the O-ISN that need to be delayed pending the completion of actions at the T-ISN.

The Bearer Control Interworking Function is responsible for processing precondition indications within the SDP and indicating to the BICC procedures when the above BICC mechanisms are required. The following indications may be passed from the Bearer Control Interworking Function to the BICC protocol procedures:

– precondition required;

– precondition met.

Similarly, when the BICC mechanism requires preconditions to be signalled, a request is made to the Bearer Control Interworking Function to add the appropriate indications to SDP. The following indications may be passed from the BICC protocol procedures to the Bearer Control Interworking Function:

- precondition required;
- precondition met.

A.2.3.1 Interworking preconditions

A.2.3.1.1 SIP to BICC

A.2.3.1.1.1 Fast Forwards set-up

On receipt of the indication precondition required from the Bearer Control Interworking Function, the Continuity Indicator in the IAM shall be set to "*COT to be expected*". Subsequently, on receipt of the indication precondition met from the Bearer Control Interworking Function (and on the determination that all preconditions local to the BICC side are also met), a COT message with Continuity Indicator set to "*Continuity*" shall be sent.

A.2.3.1.2 BICC to SIP

A.2.3.1.2.1 Fast Forwards set-up

If the indication "*COT to be expected*" is received in an IAM, then the indication precondition required is sent to the Bearer Control Interworking Function along with the Bearer Control Tunnelling Information in the IAM.

Subsequently, on receipt at the O-IWF of a COT message indicating "*continuity*", then the indication precondition met is sent to the Bearer Control Interworking Function.

A.2.3.1.2.2 Backwards set-up

No action is taken on receipt of the indications preconditions required and preconditions met.

A.2.3.1.2.3 Delayed Forwards

If the indication "*COT to be expected*" is received in the IAM, then the indication precondition required is sent to the Bearer Control Interworking Function along with the Bearer Control Tunnelling Information received in the subsequent APM.

Subsequently, on receipt of a COT message indicating "*Continuity*", then the indication precondition met is sent to the Bearer Control Interworking Function.

A.3 Bearer Control Interworking Function

A.3.1 IPBCP/SDP Bearer Control Interworking Function (BC-IWF)

This clause defines the procedures associated with a Bearer Control Interworking Function (BC-IWF) which interworks IPBCP to/from SDP. In all cases, the BC-IWF is a call stateful device. This is particularly important in enabling the BC-IWF to manipulate precondition information it receives within SDP offers/answers and IPBCP messages.

The IPBCP/SDP Bearer Control Interworking function shall behave as follows:

A.3.1.1 SDP to IPBCP

A.3.1.1.1 Receipt of SDP offer

On receipt of an SDP offer (as determined by the procedures within [IETF RFC 3264]), the BC-IWF shall send a REQUEST message on the IPBCP side. The REQUEST message contents shall be formatted as per the procedures in clause 6 of [ITU-T Q.1970]. Any SDP fields that cannot be directly carried within the SDP allowed within the IPBCP REQUEST message shall not be sent to the BICC side. In addition, if the SDP offer contained any precondition media level attributes indicating that preconditions to session establishment are present on the SIP side of the call, these shall be removed from the SDP sent to the IPBCP side. Instead, a preconditions required indication (as defined by the procedures in clause A.2.3) is sent to the BC-IWF. Subsequently, the procedures outlined in clause A.2.3.1.1 shall be followed with respect to the setting of indicators within the BICC IAM. Furthermore, if the SDP offer instead resulted in the BC-IWF receiving a preconditions met indication (as a result of the precondition SDP indicating that all mandatory preconditions had been met), then the BC-IWF shall correlate receipt of this indication with receipt of a preconditions required indication in a previous offer for this call and the procedures outlined within clause A.2.3.1.1, with respect to preconditions met, shall be followed.

A.3.1.1.2 Receipt of SDP answer

i) IPBCP has previously sent a REQUEST message for which it has not yet received an answer.

On receipt of an SDP answer (as determined by the procedures within [IETF RFC 3264]), the BC-IWF shall send an ACCEPTED message to the IPBCP side. The ACCEPTED message contents shall be formatted as per the procedures of clause 6 of [ITU-T Q.1970]. With the exception of media level attributes describing preconditions, if the SDP field is allowed to be included in the ACCEPTED message, it shall be included. If the SDP received in the answer indicates a change in status of the preconditions from any previous SDP received at the I-IWF, then this change in precondition status shall be reported to the BC-IWF using precondition indications as defined in clause A.2.3.

If the SDP answer is received, and the port number of the media stream that was being offered in the SDP offer is set to 0, then the BC-IWF shall send a REJECTED message to the IPBCP side. The REJECTED message contents shall be formatted as per the procedures of clause 6 of [ITU-T Q.1970]. With the exception of media level attributes describing preconditions, if the SDP field is allowed to be included in the REJECTED message, it shall be included.

ii) IPBCP has not previously sent a REQUEST message or has sent a REQUEST message for which an answer has been received.

On receipt of an SDP answer (as determined by the procedures within [IETF RFC 3264]), the BC-IWF shall not send any message to the IPBCP side.

A.3.1.2 IPBCP to SDP

A.3.1.2.1 Receipt of REQUEST message

On receipt of an IPBCP REQUEST message, the BC-IWF shall construct and send an SDP offer in the first SIP message sent as a result of the interworking procedures defined in this Recommendation, and as per the procedures relating to the sending of SDP offers in SIP defined within [IETF RFC 3264] and [IETF RFC 3261]. The SDP fields contained within the IPBCP REQUEST message shall be included within the SDP offer. If the BC-IWF receives a preconditions required indication, then the BC-IWF shall ensure that the SDP offer sent from the BC-IWF contains a "local" precondition (in the language of [IETF RFC 3312]). The current status of this "local" precondition shall have a strength tag of "none" and a direction tag of "none". The desired status of the local precondition shall be set to a strength of "mandatory" and a direction value of "sendrecv". Additionally, the BC-IWF shall insert a corresponding remote precondition with a desired status of strength-tag = none and direction-tag = none. The BC-IWF is responsible for storing the state of all preconditions during the duration of the call.

If, in the period between sending this offer and sending the last offer, the BC-IWF receives a precondition met indication, then the BC-IWF shall correlate receipt of this precondition status information with the value of the "local" precondition tag which it inserted on receipt of the precondition required indication received in a previous IPBCP REQUEST message. The BC-IWF shall set the current status of this precondition equal to the desired status before sending out the SDP offer containing the updated current status.

A.3.1.2.2 Receipt of ACCEPTED message

On receipt of an IPBCP ACCEPTED message, the BC-IWF shall construct and send an SDP answer in the first SIP message sent as a result of the interworking procedures defined in this Recommendation, and as per the procedures relating to the sending of SDP answers defined within [IETF RFC 3264] and [IETF RFC 3261]. The SDP fields contained within the IPBCP ACCEPTED message shall be included within the SDP answer. Additionally, the BC-IWF shall include any SDP relating to the status of the preconditions SDP sent within the SDP offer that was interworked to the REQUEST message responsible for generating this ACCEPTED message. In particular, if the BC-IWF has received a preconditions required indication in the SDP offer which generated the

REQUEST message responsible for this ACCEPTED message, then the BC-IWF shall add in precondition SDP to update the current status (and desired status if necessary) of the preconditions. The procedures used to respond to the SDP received in the previous SDP offer, correlated with this answer, are fully described in [IETF RFC 3312].

A.3.1.2.3 Receipt of CONFUSED message

On receipt of the CONFUSED message, the BC-IWF shall follow the procedures outlined within [ITU-T Q.1970].

A.3.1.2.4 Receipt of REJECTED message

On receipt of the REJECTED message, the BC-IWF shall send an SDP answer in the first available SIP message. The SDP answer shall be constructed using the SDP fields present in the REJECTED message however, the BC-IWF shall set the port number for the media stream to the value 0.

Annex B

Interworking for ISDN supplementary services

(This annex forms an integral part of this Recommendation.)

This annex describes service interworking of ISDN supplementary services between SIP and BICC/ISUP. The support of these supplementary services is optional. If the supplementary services are supported, the procedures described within this clause shall be applied.

Except where otherwise stated, services in Profile C (SIP-I) operation use the parameters of the (de)encapsulated ISUP, and no other interworking is required. Accordingly, the service interworking descriptions below are only for profiles A and B operation unless Profile C (SIP-I) is specifically indicated.

B.1 Interworking of CLIP/CLIR supplementary service to SIP networks

Profiles A and B

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 clauses 6.1.3.6 and 7.1.4. This interworking is essentially the same as for basic call and differs only in that if the CLIR service is invoked, the Address Presentation Restricted 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 signalling 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 clauses 6.1.3.6 and 7.1.4.

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

Profile C (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 clause 6.1.3.6 and processing by BICC/ISUP procedures is the same as the value contained in the encapsulated ISUP, no additional interworking is needed beyond the use of ISUP encapsulation. In the contrary case the Calling Party Sub-address is deleted from the ATP.

B.2 Interworking of COLP/COLR supplementary service to SIP networks

B.2.1 General

The COLP/COLR services are only to be interworked between trusted nodes, that is before passing any COLP/COLR information over the SIP-BICC/ISUP boundary the IWU shall satisfy itself that the nodes on the BICC/ISUP side to which the information is to be passed are trusted.

B.2.2 Incoming Call Interworking from SIP to BICC/ISUP at the I-IWU

B.2.2.1 General

For profiles A and B: If the IWU supports interworking of the COLP/COLR supplementary service the clauses B.2.2.1 to B.2.2.2 shall apply.

B.2.2.1 INVITE to IAM interworking (SIP to ISUP/BICC calls)

In the case of SIP to ISUP/BICC calls the I-IWU may invoke the COLP service as an operator option by setting the "Connected Line Identity Request indicator" field of the "Optional Forward Call Indicators" parameter of the IAM to "requested".

NOTE – This implies that all outgoing calls will invoke the COLP service.

B.2.2.2 ANM/CON to 200 OK (INVITE)

Tables B.2-1 and B.2-2 specify the interworking required in the case when the COLP has been automatically requested on behalf of the originating SIP node. The tables also indicate the interworkings required if the COLP service has been invoked and the called party has or has not invoked the COLR service.

Table B.2-1 – Mapping to P-Asserted-Identity and Privacy Header Fields

SIP Component	Setting
P-Asserted-Identity	See Table B.2-2
Privacy	See Table B.2-3

Table B.2-2 – Mapping of Connected Number parameter to SIP P-Asserted-Identity header fields

<table>
<tr><th>BICC/ISUP parameter / field</th><th>Value</th><th>SIP component</th><th>Value</th></tr>
<tr><td>Connected Number</td><td></td><td>P-Asserted-Identity header field</td><td></td></tr>
<tr><td rowspan="2">Nature of Address Indicator</td><td>"national (significant) number"</td><td rowspan="2">Tel URI or SIP URI
(Note 1)</td><td>Add CC (of the country where the IWU is located) to Connected PN address signals to construct E.164 number in URI. Prefix number with "+".</td></tr>
<tr><td>"international number"</td><td>Map complete Connected address signals to construct E.164 number in URI. Prefix number with "+".</td></tr>
</table>

Table B.2-2 – Mapping of Connected Number parameter to SIP P-Asserted-Identity header fields

BICC/ISUP parameter / field	Value	SIP component	Value
Connected Number		P-Asserted-Identity header field	
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	Tel URI or SIP URI (Note 1)	CC+NDC+SN as E.164 number in URI. Prefix number with "+".
NOTE 1 –A tel URI or a SIP URI with "user=phone" is used according to operator policy.			

Table B.2-3 – Mapping of BICC/ISUP APRIs into SIP privacy header fields

BICC/ISUP parameter / field	Value	SIP component	Value
Connected Number		Privacy header field	priv-value
APRI	*"presentation restricted"*	Priv-value	"id" ("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 "header" if other privacy service is needed

B.2.2.3 TIP/TIR interworking

For profiles A and B: If the IWU supports the interworking of the TIP/TIR supplementary service this subclause shall apply.

For the mapping of the INVITE request to IAM:

– The bit H Connected Line Identity Request Indicator of the Optional Forward Call Indicators parameter in the IAM shall be set to "requested".

If a received ISUP ANM includes an ISUP Generic Number ("additional connected number") parameter, then the

I-IWU shall send a 200 (OK) response (to the INVITE request) including an option tag "from-change". If the initial INVITE was received and the Supported header field contains the "from-change" tag, the 200 (OK) response is followed by an UPDATE request, containing the "additional connected number" copied into the From header field as shown in Table B.2-4.

The To header field of the UPDATE request is derived from the P-Asserted-Identity header field received within the initial INVITE request.

Table B.2-4 – Mapping of ANM Generic Number ("additional connected number") to SIP From header field in a SIP UPDATE request

ISUP Parameter/field	Value	SIP component	Value
Generic Number Number Qualifier Indicator	"*additional connected 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 Generic Number Address Signals then map to user portion of URI scheme used
	"international number"		Map complete Generic Number Address Signals used prefixed with a "+" to user portion of URI scheme used
Address Presentation Restriction Indicator (APRI)	"*presentation allowed*"		No Privacy header field or not "*user*"
	"*presentation restricted*"		"*user*"
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	Display-name (optional)	display-name shall be mapped from Address Signals, if network policy allows it
		Addr-spec	"+" CC NDC SN mapped to user portion of URI scheme used

A received connected number in an ANM shall be mapped to the P-Asserted-Identity header field as shown in Table B.2-3 of the UPDATE request.

B.2.3 Outgoing Call Interworking from BICC/ISUP to SIP at O-IWU

B.2.3.1 General

For profiles A and B: If the IWU supports interworking of the COLP/COLR supplementary service clauses B.2.3.2 to B.2.3.4 shall apply.

B.2.3.2 IAM to INVITE interworking (ISUP to SIP calls)

The O-IWU determines that the COLP service has been requested by the calling party by parsing the "Optional Forward Call Indicators" field of the incoming IAM. If the "Connected Line Identity Request Indicator" is set to "requested" then the BICC/ISUP to SIP interworking node shall ensure that any backwards "connected party" information is interworked to the appropriate parameters of the ISUP ANM or CON message sent backwards to the calling party as detailed within this subclause.

The O-IWU has to store the status of the "Connected Line Identity Request Indicator".

B.2.3.3 1xx to ANM or CON interworking

If the P-Asserted-Identity header field is included within a 1xx SIP response, the identity shall be stored within the O-IWU together with information about the SIP dialogue of the 1xx SIP response and shall be included within the ANM or CON message if required by the procedures described in clause B.2.3.4. In accordance with ISUP procedures, a connected number shall not be included within the ACM message. The mapping of the of the P-Asserted-Identity and Privacy header fields is shown in Tables B.2-2 and B.2-4.

B.2.3.4 200 OK (INVITE) to ANM/CON interworking

Tables B.2-5 and B.2-6 specify the interworking required in the case when the calling party has invoked the COLP service. The tables also indicate the interworking procedures required if the calling party has invoked the COLP service and the called party has or has not invoked the COLR service.

If no P-Asserted-Identity header field is provided within the 200 OK (INVITE) message, the stored information previously received in the last provisional 1xx response of the same SIP dialogue shall be used.

NOTE – Due to forking, other P-Asserted-Identities might have been received in different SIP dialogues.

If the Calling Party has requested the COLP service (as indicated by the stored request status) but the 200 OK (INVITE) and previous 1xx provisional responses do not include a P-Asserted-Identity header field, the O-IWU shall set up a network provided Connected Number with an Address not Available indication.

If the P-Asserted-Identity is available then the Connected number has to be set up with the screening indication network provided. The mapping of the P-Asserted-Identity and Privacy (if available) is shown in Table B.2-6.

Table B.2-5 – Connected Number parameter mapping

← ANM/CON	← 200 OK INVITE
Connected Number (Network Provided)	P-Asserted-ID
Address Presentation Restricted Indicator	Privacy Value Field

Table B.2-6 – Mapping of P-Asserted-Identity and privacy headers to the ISUP/BICC Connected Number parameter

SIP component	Value	BICC/ISUP parameter / field	Value
P-Asserted-Identity header field (NOTE)	E.164 number	Connected Number	
		Number Incomplete Indicator	"Complete"
		Numbering Plan Indicator	"ISDN/Telephony (E.164)"
		Nature of Address Indicator	If CC encoded in the URI is equal to the CC of the country where IWU is located AND the next BICC/ISUP node is located in the same country then set to "national (significant) number" else set to "international number"

Table B.2-6 – Mapping of P-Asserted-Identity and privacy headers to the ISUP/BICC Connected Number parameter

SIP component	Value	BICC/ISUP parameter / field	Value
		Address Presentation Restricted Indicator (APRI)	Depends on priv-value in Privacy header.
		Screening Indicator	Network Provided
Addr-spec	"CC" "NDC" "SN" from the URI	Address signal	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 is not present		APRI	Presentation allowed
Privacy header field	priv-value	APRI	"Address Presentation Restricted Indicator"
priv-value	"header"	APRI	Presentation restricted
	"user"	APRI	Presentation restricted
	"none"	APRI	Presentation allowed
	"id"	APRI	Presentation restricted
NOTE – It is possible that a P-Asserted-Identity header field includes both a tel URI and a SIP URI. In this case, either the tel URI or the SIP URI with user="phone" and a specific host portion, as selected by operator policy, may be used.			

B.2.3.5 TIP/TIR interworking

For profiles A and B: If the IWU supports the interworking of the TIP/TIR supplementary service this subclause shall apply.

For the mapping of IAM to the INVITE request:

If an Optional Forward Call Indicators parameter in the IAM is received where the bit H Connected Line Identity Request Indicator is set to "requested", then the option tag "from-change" shall be add to the Supported header field. See Table B.2-7.

Table B.2-7 – Mapping of ISUP IAM to SIP INVITE request

ISUP Parameter	Derived value of parameter field	Source SIP header field and component	Source Component value
Optional Forward Call Indicator	*Connected line identity request indicator* is set to "requested"	Supported	"from-change"

NOTE – The presence of "from-change" enables the reception of Generic Number with Number Qualifier parameter field set to additional connected, if available. As per [ITU-T Q.3617] the presence of "from-change" tag is not a criterion for TIP service.

If a provisional or final response including the option tag "from-change" is received, then the O-IWU shall:

– if a 200 (OK) response to the INVITE request is received, start timer T_{TIR1}; and

– store the 200 (OK) response, without interworking it.

Otherwise, the 200 (OK) response (to the INVITE request) shall be mapped as described in clause 7.5. The timer T_{TIR1} is described in clause B.20.

If an UPDATE request is received containing a changed From header field before the timer TTIR1 expired, then the O-IWU shall:

– stop timer TTIR1;

– map the From header field received in the UPDATE request to the Generic number in the ANM as shown in Table B.2-8 and Table B.2-9;

– if the UPDATE request includes a P-Asserted-Identity header field that is different from the one within the stored 200 (OK) response, the latest received P-Asserted-Identity header field shall be mapped to the connected number as described in Table B.2-6; and

– map the parameters needed to be mapped of the stored 200 (OK) response to an ANM as described in clause B.2.3.4, modified by the changed mapping steps of the From and P-Asserted-Identity header fields.

When T_{TIR1} expires, then the stored 200 (OK) response (to the INVITE request) response shall be mapped as described in clause B.2.3.4.

Table B.2-8 – Mapping of SIP UPDATE request to ISUP ANM/CON

ANM/CON	UPDATE
Generic number	From header field

Table B.2-9 – Mapping of SIP From header field to ISUP Generic Number ("additional connected number") parameter

Source SIP header field and component	Source component value	Generic Number parameter field	Derived value of parameter field
–	–	Number Qualifier Indicator	"*additional connected number*"
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*"
–	–	Number Incomplete Indicator	"*complete*"
–	–	Numbering Plan Indicator	"*ISDN (Telephony) numbering plan (Recommendation E.164)*"
Privacy, priv-value component	Privacy header field absent	Address Presentation Restricted Indicator (APRI)	"*presentation allowed*"
	"*none*"		"*presentation allowed*"
	"*header*"		"*presentation allowed*"
	"*user*"		"*presentation restricted*"
	"id"		"*presentation allowed*"

Table B.2-9 – Mapping of SIP From header field to ISUP Generic Number ("additional connected number") parameter

Source SIP header field and component	Source component value	Generic Number parameter field	Derived value of parameter field
–	–	Screening Indicator	"*user provided, not verified*"
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 number*" then set to CC + NDC + SN

B.2.4 Profile C (SIP-I)

No additional interworking beyond the use of ISUP encapsulation is required.

B.3 Interworking of Direct-Dialling-In (DDI) supplementary service to SIP networks

A direct dialling-in call is a basic call and no additional treatment is required by the IWU.

Profile C (SIP-I)

No additional interworking beyond the use of ISUP encapsulation.

B.4 Interworking of Malicious Call Identification (MCID) supplementary service to SIP networks

If the IMS supplementary service Malicious Communication Identification (MCID) is not supported: the IWU shall act in accordance with the procedures described in clause 7.7 of [ITU-T Q.731.7] under the subclause "Interactions with other networks".

B.4.1 Malicious Communication Identification (MCID)

B.4.1.1 General

For profiles A and B: The protocol specification of the Malicious Communication Identification supplementary service is described in [ITU-T Q.3624]. The XML MCID body used in related SIP messages is also specified in [ITU-T Q.3624].

B.4.1.2 Interworking at the O-IWU

B.4.1.2.1 General

If the IWU supports the interworking of the MCID service the O-IWU shall map a SIP INFO request ("legacy" mode of usage of the INFO method as defined in [IETF RFC 6086] containing a XML mcid body with MCID XML Request schema to an Identification Request (IDR) message and an Identification response (IRS) message to a SIP INFO request containing a XML mcid body with MCID XML Response schema in accordance with Table B.4-1.

The IDR message shall be generated upon receipt of the SIP INFO request containing a XML mcid body with MCID XML Request schema.

The SIP INFO request containing a XML mcid body with MCID XML Response schema shall be generated upon receipt of the IRS message.

Table B.4-1 – Mapping between ISUP IDR and IRS and SIP messages

ISUP Message	SIP Message
IDR	INFO containing a XML mcid body with MCID XML Request schema
IRS	INFO containing a XML mcid body with MCID XML Response schema

B.4.1.2.2 Interworking of the MCID XML Request schema with the ISUP MCID Request Indicators

If the IWU supports the interworking of the MCID service O-IWU shall map the codes in the MCID XML elements to the MCID Request Indicator and Holding Indicator parameter fields in accordance with Table B.4-2.

Table B.4-2 – Mapping between ISUP MCID request and holding indicators and MCID XML elements

ISUP Parameter		XML Element
bit A:	MCID Request Indicator	McidRequestIndicator
0	MCID not requested	type=0
1	MCID requested	type=1
bit B:	Holding Indicator (national use)	HoldingIndicator
0	holding not requested	type=0
1	holding requested	type=1

B.4.1.2.3 Interworking of the ISUP MCID Response Indicators with the MCID XML Response schema

If the IWU supports the interworking of the MCID service the O-IWU shall map the codes in the MCID Response Indicator and Hold Provided Indicator parameter field to the MCID XML elements in accordance with Table B.4-3.

Table B.4-3 – Mapping between ISUP MCID Response and Hold Provided Indicators and MCID XML elements

ISUP Parameter		XML Element
bit A:	MCID Response Indicator	McidResponseIndicator
0	MCID not included	type=0
1	MCID included	type=1
bit B:	*Hold provided indicator (national use)*	HoldingProvidedIndicator
0	holding not provided	type=0
1	holding provided	type=1

B.4.1.2.4 Interworking of the ISUP Calling Party Number in an Identification Response with the OrigPartyIdentity within the MCID XML Response schema

If the O-IWU supports the interworking of the MCID service and receives an ISUP Identification Response containing a Calling Party Number with the screening indicator set to "*user provided,* *verified and passed*" or "*network provided*", the O-IWU shall map the Calling Party Number to the MCID XML Response schema OrigPartyIdentity element applying the same mapping procedures as described in Table 7-6 for the mapping into the SIP P-Asserted-Identity header and shall map the Calling Party Number APRI to the MCID XML Response schema OrigPartyPresentationRestriction element. If the Calling Party Number APRI has a value of "*presentation allowed*" then the MCID XML Response schema OrigPartyPresentationRestriction element shall be set to "*false*", otherwise it shall be set to "*true*".

B.4.1.2.5 Interworking of the ISUP Generic Number in an Identification Response with the GenericNumber within the MCID XML Response schema

If the O-IWU supports the interworking of the MCID service and receives an ISUP Identification Response containing a Generic Number with the screening indicator set to "*user provided, verified and passed*", or "*user provided, not verified*", or "*network provided*", the O-IWU shall map the Generic Number to the MCID XML Response schema GenericNumber element applying the same mapping procedures as described in Table 7-5 for the mapping into the SIP From header and shall map the Generic Number APRI to the MCID XML Response schema GenericNumberPresentationRestriction element. If the Generic Number APRI has a value of "*presentation allowed*" then the MCID XML Response schema GenericNumberPresentationRestriction element shall be set to "*false*", otherwise it shall be set to "*true*".

B.4.1.3 Interworking at the I-IWU

B.4.1.3.1 General

For profiles A and B: If the IWU supports the interworking of the MCID service the I-IWU shall map an Identification Request (IDR) message to a SIP INFO request ("legacy" mode of usage of the INFO method as defined in [IETF RFC 6086] containing a XML mcid body with MCID XML Request schema and a SIP INFO request containing a XML mcid body with MCID XML Response schema to an Identification response (IRS) message in accordance with Table B.4-1.

B.4.1.3.2 Interworking of identification Request

The SIP INFO request containing a XML mcid body with MCID XML Request schema shall be generated upon receipt of the IDR message. The I-IWU shall map the codes in the MCID request indicator and holding indicator parameter fields to the MCID XML elements in accordance with Table B.4-2.

B.4.1.3.3 Interworking of identification Response

The IRS message shall be generated upon receipt of the SIP INFO request containing a XML mcid body with MCID XML Response schema. The I-IWU shall map the codes in the MCID XML elements to the MCID response indicator and hold provided indicator parameter fields in accordance with Table B.4-3.

If the received MCID XML Response schema contains an OrigPartyIdentity element, the I-IWU shall map the OrigPartyIdentity to the Calling Party Number within the IRS applying the same mapping procedure as described in Table 6-13 for the mapping from the SIP P-Asserted-Identity header, with the exception that the I-IWU shall map the MCID XML Response schema OrigPartyPresentationRestriction element to the Calling Party Number APRI. If the MCID XML Response schema OrigPartyPresentationRestriction element has the value "*true*", the Calling Party Number APRI shall be set to "*presentation restricted*", and otherwise the Calling Party Number APRI shall be set to "*presentation allowed*".

If the received MCID XML Response schema contains an GenericNumber element, the I-IWU shall map the GenericNumber to the Generic Number within the IRS applying the same mapping procedure as described in Table 6-15 for the mapping from the From header, with the exception that the I-IWU shall map the MCID XML Response schema GenericNumber PresentationRestriction element to the Generic Number APRI. If the MCID XML Response schema GenericNumberPresentationRestriction element has the value "*true*", the Generic Number APRI shall be set to "*presentation restricted*", and otherwise the Generic Number APRI shall be set to "*presentation allowed*".

B.4.2 Profile C (SIP-I)

All parameters can be taken from the encapsulated ISUP MIME as usual. However, the IP bearer cannot be held after the release of the call.

B.5 Interworking of Sub-addressing (SUB) supplementary service to SIP networks

B.5.1 General

The ISDN subaddress information in ISUP is transported within the Access Transport parameter. The coding of the subaddress parameter within the Access Transport parameter is described within [ITU-T Q.931]. The isdn-subaddress parameter "isub" carried within a tel or SIP URI is defined within [IETF RFC 3966]. The isdn-subaddress encoding type parameter "isub-encoding" carried within a tel or SIP URI is defined within [IETF RFC 4715].

B.5.1.1 Interworking at I-IWU

For profiles A and B: If the interworking of the "isup" parameter of the To header into the ISDN subaddress is supported the procedures described in this subclause shall apply.

The mapping in Table B.5-1 of the isdn-subaddress parameter received within a tel or SIP URI of the initial INVITE request to the ISUP Access Transport parameter encapsulating the subaddress information (calling party subaddress and/or called party subaddress information elements) to be sent within the IAM message shall be applied.

The mapping in Table B.5-2 of the subaddress information (connected subaddress information element) received within an ANM message containing the ISUP Access Transport parameter to the isdn-subaddress parameters "isub" and "isub-encoding" of a tel or SIP URI to be sent within a 200 OK (INVITE) shall be applied.

Table B.5-1 – Mapping of the isdn-subaddress received in an initial INVITE to the subaddress information sent in the IAM

<table>
<tr><th colspan="3">SIP Message INVITE</th><th colspan="3">ISUP Message IAM</th></tr>
<tr><th>Source SIP header field and component</th><th colspan="2">Source component value</th><th>ISUP Parameter field</th><th colspan="2">Derived value of parameter field</th></tr>
<tr><td rowspan="4">To header field including the isdn-subaddress
(Note)</td><td rowspan="4">"isub=" 1*uric
"uric" containing the subaddress digits</td><td>isub-encoding not present</td><td rowspan="4">Access Transport Parameter</td><td rowspan="4">called party subaddress</td><td rowspan="2">Type of subaddress = "NSAP" (000)</td></tr>
<tr><td>"isub-encoding=nsap-ia5"</td></tr>
<tr><td>"isub-encoding=nsap-bcd"</td><td>Type of subaddress = "NSAP" (000)</td></tr>
<tr><td>"isub-encoding=nsap"</td><td>Type of subaddress = "NSAP" (000)</td></tr>
</table>

Table B.5-1 – Mapping of the isdn-subaddress received in an initial INVITE to the subaddress information sent in the IAM

<table>
<tr><th colspan="3">SIP Message INVITE</th><th colspan="3">ISUP Message IAM</th></tr>
<tr><td></td><td colspan="2">"isub=" 1*uric ("uric" containing the subaddress digits) and isub-encoding does not contain nsap value</td><td colspan="3">No mapping</td></tr>
<tr><td rowspan="5">P-Asserted-Identity header Field including the isdn-subaddress</td><td rowspan="4">";isub=" 1*uric
"uric" containing the subaddress digits</td><td>isub-encoding not present</td><td rowspan="4">Access Transport Parameter</td><td rowspan="4">calling party subaddress</td><td rowspan="2">Type of subaddress = "NSAP" (000)</td></tr>
<tr><td>"isub-encoding=nsap-ia5"</td></tr>
<tr><td>"isub-encoding=nsap-bcd"</td><td>Type of subaddress = "NSAP" (000)</td></tr>
<tr><td>"isub-encoding=nsap"</td><td>Type of subaddress = "NSAP" (000)</td></tr>
<tr><td colspan="2">"isub=" 1*uric ("uric" containing the subaddress digits) and isub-encoding does not contain nsap value</td><td colspan="3">No mapping</td></tr>
<tr><td colspan="6">NOTE – As an operator option, an isdn-subaddress within the Request-URI may also be mapped into the ISUP Access Transport parameter.</td></tr>
</table>

Table B.5-2 – Mapping of the subaddress information received in an ANM to the isdn subaddress sent in the 200 OK (INVITE)

ISUP Message ANM		SIP Message 200 (OK)	
ISUP Parameter field	**Source component value**	**Source SIP header field and component**	**Derived value of parameter field**
Access Transport parameter	connected subaddress and Type of subaddress = "NSAP" (000)	P-Asserted-Identity including the isdn-subaddress	";isub=" 1*uric and "isub-encoding=nsap-ia5" The subaddress digits included into the "uric" shall be derived from the Access Transport Parameter
	connected subaddress and Type of subaddress ≠ "NSAP" (000)	No mapping	

B.5.1.2 Interworking at O-IWU

For profiles A and B: If the interworking of the ISDN subaddress into the "isup" parameter of the To header is supported the procedures described in this subclause shall apply.

The mapping in Table B.5-3 of the subaddress information received in the Access Transport parameter (calling party subaddress and/or called party subaddress information elements) of the IAM message to the isdn-subaddress parameters "isub" and "isub-encoding" of a tel or SIP URI to be sent within the initial INVITE request shall be applied.

The mapping in Table B.5-4 of the isdn-subaddress information received within a tel or SIP URI of a 200 OK (INVITE) to the subaddress information of the ISUP Access Transport parameter (connected subaddress information element) to be sent within the ANM message shall be applied.

Table B.5-3 – Mapping of the subaddress information received in an IAM to the isdn-subaddress sent in the INVITE

<table>
<tr><th colspan="2">ISUP IAM Message</th><th colspan="2">SIP INVITE Message</th></tr>
<tr><th>ISUP Parameter field</th><th>Source component value</th><th>Source SIP header field and component</th><th>Derived value of parameter field</th></tr>
<tr><td rowspan="4">Access Transport parameter</td><td>called party subaddress and Type of subaddress = "NSAP" (000)</td><td>To header field including the isdn-subaddress, and, as an operator option, Request URI</td><td>";isub=" 1*uric and "isub-encoding=nsap-ia5"
The subaddress digits included into the "uric" shall be derived from the Access Transport Parameter</td></tr>
<tr><td>called party subaddress and Type of subaddress ≠ "NSAP" (000)</td><td colspan="2">No mapping</td></tr>
<tr><td>calling party subaddress and Type of subaddress = "NSAP" (000)</td><td>P-Asserted-Identity header field including the isdn-subaddress</td><td>";isub=" 1*uric and "isub-encoding=nsap-ia5"
The subaddress digits included into the "uric" shall be derived from the Access Transport Parameter</td></tr>
<tr><td>calling party Subaddress and Type of Subaddress ≠ "NSAP" (000)</td><td colspan="2">No mapping</td></tr>
</table>

Table B.5-4 – Mapping of the isdn subaddress received in a 200OK to the subaddress information sent in the ANM

<table>
<tr><th colspan="3">SIP Message 200 (OK)</th><th colspan="3">ISUP Message ANM</th></tr>
<tr><th>Source SIP header field and component</th><th colspan="2">Source component value</th><th>ISUP Parameter field</th><th colspan="2">Derived value of parameter field</th></tr>
<tr><td rowspan="3">P-Asserted-Identity header Field
including the isdn-subaddress</td><td rowspan="3">"isub=" 1*uric

"uric" containing the subaddress digits</td><td>isub-encoding not present</td><td rowspan="3">Access Transport parameter</td><td rowspan="3">connected subaddress</td><td rowspan="2">Type of subaddress = "NSAP" (000)</td></tr>
<tr><td>"isub-encoding=nsap-ia5"</td></tr>
<tr><td>"isub-encoding=nsap-bcd"</td><td>Type of subaddress = "NSAP" (000)</td></tr>
</table>

Table B.5-4 – Mapping of the isdn subaddress received in a 200OK to the subaddress information sent in the ANM

SIP Message 200 (OK)			ISUP Message ANM		
		"isub-encoding=nsap"			Type of subaddress = "NSAP" (000)
	"isub=" 1*uric ("uric" containing the subaddress digits) and isub-encoding does not contain nsap value		No mapping		

B.5.2 Profile C (SIP-I)

At the O-IWU: the service shall be supported by encapsulation.

At the I-IWU: If the address within the Called Party Number after application of the interworking rules in clause 6.1.3.6 and processing by BICC/ISUP procedures is the same as the value contained in the encapsulated ISUP, no additional interworking is needed beyond the use of ISUP encapsulation. In the contrary case the Called Party Sub-address is deleted from the ATP.

B.6 Interworking of Call Forwarding Busy (CFB)/Call Forwarding No Reply (CFNR)/Call Forwarding Unconditional (CFU)/ Call Deflection (CD) supplementary services to SIP networks

If the IMS Communication Diversion (CDIV) supplementary service is not supported, the IWU shall act in accordance with the procedures described within clause 2.7 of [ITU-T Q.732.2-5], under the heading "Interactions with other networks".

B.6.1 Communication Diversion (CDIV)

B.6.1.1 General

For profiles A and B: If the IMS Communication Diversion (CDIV) supplementary service is not supported the following subclauses apply.

The protocol specification of the Communication Diversion supplementary service is described in [ITU-T Q.3620]. The mapping of Communication Diversion supplementary service with Call Diversion services PSTN/ISDN supplementary service including the mapping of the optional History-Info header field as defined in [IETF RFC 7044] is described.

The hi-target-param parameter set to "mp" as defined in [IETF RFC 7044] indicates that the target of the Request-URI was changed.

In case of interworking with networks which do not provide any notifications of the communication diversion or communication redirection information (e.g., redirection counter) in the signalling system, the communication continues according to the basic call procedures.

In case of interworking with networks not supporting [IETF RFC 7044] the "mp" header field parameter may not appear.

B.6.1.2 Interworking at the O-IWU

B.6.1.2.1 General

For profiles A and B: For the mapping of IAM to the INVITE request no additional procedures beyond the basic call and interworking procedures are needed unless Call forwarding within the ISUP network appeared.

With regard to the backward messages the following mapping is valid.

Table B.6.1-1 – 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 B.6.1-5
CPG indicating call forwarding (see NOTE)	181 (Call Is Being Forwarded) response	See Table B.6.1-6
ACM indicating ringing	180 (Ringing) response	See Table B.6.1-7
CPG indicating Alerting (see NOTE)	180 (Ringing) response	See Table B.6.1-8
ANM	200 (OK) response	See Table B.6.1-9
CON	200 (OK) response (Neither a 181 (Call Is Being Forwarded) response nor a 180 (Ringing) response was received)	See Table B.6.1-10
NOTE – A CPG will be sent if an ACM was already sent.		

Table B.6.1-2 – Mapping of History-Info header field to ISUP Redirection number

Source SIP header field and component	Source Component value	Redirection number	Derived value of parameter field
hi-targeted-to-uri of the last History-Info hi-entry containing a "cause" URI parameter, as defined in [IETF RFC 4458]. (Note 2) The global number portion of the hi-targeted-to-uri is 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 O-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.
NOTE 1 – If the SIP URI does not contain "user=phone", mapping to the redirection number is impossible, therefore no need to generate Redirection number and Redirection number restriction indicator (per Table B.6.1-3), Notification subscription options cannot be set as "presentation allowed with redirection number". NOTE 2 – The hi-target-param parameter set to "mp" as defined in [IETF RFC 7044] indicates that the target of the Request-URI was changed and appears in this hi-targeted-to-uri. In case of interworking with networks not supporting [IETF RFC 7044] the "mp" header field parameter may not appear.			

Table B.6.1-3 – Mapping of History-Info header field to ISUP Redirection number restriction

Source SIP header field and component	Source Component value	Redirection number restriction	Derived value of parameter field
Privacy "headers" component of the hi-targeted-to-uri or/and Privacy header field	"*history*" or "*session*" or "*header*"	Presentation Restricted Indicator	"*Presentation restricted*"
	Privacy "headers" component of the hi-targeted-to-uri and Privacy header field absent or "*none*"		"*Presentation allowed*" or absent

Table B.6.1-4 – Mapping of hi-targeted-to-uri to ISUP Call Diversion Information

Source SIP header field and component	Source Component value	Call Diversion Information	Derived value of parameter field
Privacy "headers" component of the hi-targeted-to-uri **or/and** Privacy header field		Notification subscription options	If the priv-value "*history*" or "*session*" or "*header*" is received within the Privacy header field or the priv-value "*history*" is received within the "headers" component of the hi-targeted-to-uri representing the diverting URI(s) and within the hi-targeted-to-uri representing diverted-to URI then "*presentation not allowed*" shall be set Otherwise, if the priv-value "*history*" is received only within the "headers" component of the hi-targeted-to-uri representing the diverted-to URI then "*presentation allowed without redirection number*" shall be set. (Note 1, Note 2) Otherwise, "*presentation allowed with redirection number*" shall be set.
"cause" URI parameter, as defined in [IETF RFC 4458] of the last History-Info hi-entry containing hi-targeted-to-uri with "cause" URI parameter. (Note 3)	**Cause value**	Call diversion information	**Redirecting Reason**
	404		*Unknown*
	302		*Unconditional*
	486		*User busy*
	408		*No reply*
	480		*Deflection immediate*
	503		*Mobile subscriber not reachable*
	487		*Deflection during alerting*
NOTE 1 – diverting URI corresponds to the hi-targeted-to-uri of the hi-entry containing a hi-index value that matches the "mp" header field parameter value of the diverted-to URI. If the diverted-to URI does not contain the "mp" header field parameter, the diverting URI corresponds to the hi-targeted-to-uri of the hi-entry before the last hi-entry containing "cause" URI parameter.			

Table B.6.1-4 – Mapping of hi-targeted-to-uri to ISUP Call Diversion Information

Source SIP header field and component	Source Component value	Call Diversion Information	Derived value of parameter field
NOTE 2 – diverted-to URI corresponds to the hi-targeted-to-uri of the last hi-entry containing "cause" URI parameter and is mapped to the Redirection number, see Table B.6.1-2. NOTE 3 – The hi-target-param parameter set to "mp" as defined in [IETF RFC 7044] indicates that the target of the Request-URI was changed and appears in this hi-targeted-to-uri. In case of interworking with networks not supporting [IETF RFC 7044] the "mp" header field parameter may not appear.			

Table B.6.1-5 – Mapping of 181 (Call Is Being Forwarded) → ACM if no ACM was sent before

Source SIP header field and component	Source Component value	ISUP Parameter	Derived value of parameter field
181 (Call Is Being Forwarded)		ACM	
		Generic notification indicators	*Call is diverting*
History-Info header field	See Table B.6.1-2	Redirection number	See Table B.6.1-2
Privacy "headers" component of the hi-targeted-to-uri or/and Privacy header field	See Table B.6.1-3	Redirection number restriction	See Table B.6.1-3
Privacy "headers" component of the hi-targeted-to-uri or/and Privacy header field	See Table B.6.1-4	Call diversion information *Notification subscription options*	See Table B.6.1-4
hi-targeted-to-uri; "cause" URI parameter as defined in IETF RFC 4458 of the last History-Info hi-entry containing such "cause" URI parameter. (Note)	See Table B.6.1-4	Call diversion information	Redirecting Reason See Table B.6.1-4
NOTE – The hi-target-param parameter set to "mp" as defined in [IETF RFC 7044] indicates that the target of the Request-URI was changed and appears in this hi-targeted-to-uri. In case of interworking with networks not supporting [IETF RFC 7044] the "mp" header field parameter may not appear.			

Table B.6.1-6 – 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	
		Generic notification indicators	*Call is diverting*

Table B.6.1-6 – 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
"cause" URI parameter, as defined in [IETF RFC 4458] of the last History-Info hi-entry containing hi-targeted-to-uri with "cause" URI parameter. (Note 2)	486	Event indicator	*CFB* (national use)
	408 (see Note 1)		*CFNR* (national use)
	302		*CFU* (national use)
	Any other value, or if appropriate national use value CFB, CFNR or CFU is not used in a network. Or if no agreement exists between operators to use theses values, or if no hi-targeted-to-uri with "cause" URI parameter is contained in the SIP 181.		*PROGRESS*
History-Info header field	See Table B.6.1-2	Redirection number	See Table B.6.1-2
Privacy "headers" component of the hi-targeted-to-uri or/and Privacy header field	See Table B.6.1-3	Redirection number restriction	See Table B.6.1-3
Privacy "headers" component of the hi-targeted-to-uri or/and Privacy header field	See Table B.6.1-4	Call diversion information *Notification subscription options*	See Table B.6.1-4
hi-targeted-to-uri; "cause" URI parameter, as defined in IETF RFC 4458 of the last History-Info hi-entry containing such "cause" URI parameter. (NOTE 2)	See Table B.6.1-4	Call diversion information *Redirecting Reason*	See Table B.6.1-4
NOTE 1 – This appears in the cases of CFNR or CDa. NOTE 2 – The hi-target-param parameter set to "mp" as defined in [IETF RFC 7044] indicates that the target of the Request-URI was changed and appears in this hi-targeted-to-uri. In case of interworking with networks not supporting [IETF RFC 7044] the "mp" header field parameter may not appear.			

Table B.6.1-7 – Mapping of 180 (Ringing) → ACM if no ACM was sent before

Source SIP header field and component	Source Component value	ISUP Parameter	Derived value of parameter field
180 (Ringing) response		ACM	
History-Info header field	If hi-targeted-to-uri of at least one History-Info hi-entry contains a "cause" URI	Generic notification indicators	*Call is diverting*

Table B.6.1-7 – Mapping of 180 (Ringing) → ACM if no ACM was sent before

Source SIP header field and component	Source Component value	ISUP Parameter	Derived value of parameter field
	parameter, as defined in [IETF RFC 4458].		
History-Info header field	See Table B.6.1-2	Redirection number (Note 1)	See Table B.6.1-2
Privacy "headers" component of the hi-targeted-to-uri or/and Privacy header field	See Table B.6.1-3	Redirection number restriction (Note 1)	See Table B.6.1-3
Privacy "headers" component of the hi-targeted-to-uri or/and Privacy header field	See Table B.6.1-4	Call diversion information *Notification subscription options* (Note 1)	See Table B.6.1-4
hi-targeted-to-uri; "cause" URI parameter, as defined in [IETF RFC 4458] of the last History-Info hi-entry containing such "cause" URI parameter. (NOTE 2)	See Table B.6.1-4	Call diversion information *Redirecting Reason* (Note 1)	See Table B.6.1-4
NOTE 1 – Parameter shall only be supplied if hi-targeted-to-uri of at least one History-Info hi-entry contains a "cause" URI parameter, as defined in [IETF RFC 4458]. NOTE 2 – The hi-target-param parameter set to "mp" as defined in [IETF RFC 7044] indicates that the target of the Request-URI was changed and appears in this hi-targeted-to-uri. In case of interworking with networks not supporting [IETF RFC 7044] the "mp" header field parameter may not appear.			

The mapping described within Table B.6.1-7 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 the backward direction.

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

Table B.6.1-8 – Mapping of 180 (Ringing) → CPG if ACM was already sent

Source SIP header field and component	Source Component value	ISUP Parameter	Derived value of parameter field
180 (Ringing) response		CPG	
History-Info header field	If hi-targeted-to-uri of at least one History-Info hi-entry contains a "cause" URI parameter, as defined in [IETF RFC 4458].	Generic notification indicators	*Call is diverting*
		Event Indicator	*ALERTING*
History-Info header field	See Table B.6.1-2	Redirection number	See Table B.6.1-2

Table B.6.1-8 – Mapping of 180 (Ringing) → CPG if ACM was already sent

Source SIP header field and component	Source Component value	ISUP Parameter	Derived value of parameter field
		(Note 1)	
Privacy "headers" component of the hi-targeted-to-uri or/and Privacy header field	See Table B.6.1-3	Redirection number restriction (Note 1)	See Table B.6.1-3
Privacy "headers" component of the hi-targeted-to-uri or/and Privacy header field	See Table B.6.1-4	Call diversion information *Notification subscription options* (Note 1)	See Table B.6.1-4
hi-targeted-to-uri; "cause" URI parameter, as defined in [IETF RFC 4458] of the last History-Info hi-entry containing such "cause" URI parameter. (NOTE 2)	See Table B.6.1-4	Call diversion information *Redirecting Reason* (Note 1)	See Table B.6.1-4
NOTE 1 – Parameter shall only be supplied if hi-targeted-to-uri of at least one History-Info hi-entry contains a "cause" URI parameter, as defined in [IETF RFC 4458]. NOTE 2 – The hi-target-param parameter set to "mp" as defined in [IETF RFC 7044] indicates that the target of the Request-URI was changed and appears in this hi-targeted-to-uri. In case of interworking with networks not supporting [IETF RFC 7044] the "mp" header field parameter may not appear.			

The mapping in Table B.6.1-8 appears when a 181 previously was mapped to an ACM. Therefore the state machine of the IWU knows that a CDIV is in Progress.

Table B.6.1-9 – Mapping of 200 (OK) response

Source SIP header field and component	Source Component value	ISUP Parameter	Derived value of parameter field
200 (OK) response		ANM/CON	
History-Info header field	See Table B.6.1-2	Redirection number (Note)	See Table B.6.1-2
Privacy "headers" component of the hi-targeted-to-uri or/and Privacy header field	See Table B.6.1-3	Redirection number restriction	See Table B.6.1-3
NOTE – The Redirection number shall only be supplied if a 200 (OK) response is mapped to an ANM message.			

B.6.1.2.2 Call forwarding within the ISUP Network appeared

The following scenario shows if a Call Forwarding appears in the ISUP/PSTN and the diverted-to number is within the SIP network. Table B.6.1-10 should be seen as an example.

For the mapping 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 redirecting number at the O-IWU it can be necessary 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" URI parameter and a hi-index and an mp-param, as described within Table B.6.1-10.

Table B.6.1-10 – 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 created hi-entry IF Redirection counter exceeds 1 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 Address Signals to construct E.164 number in URI.
	"international number"		Map complete Redirecting number Address Signals to E.164 number in URI.
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	Addr-spec "+" CC NDC SN mapped to userinfo portion of SIP URI. (Note 5) Add "user=phone".
Redirecting Number	APRI	Privacy header field that corresponds to the penultimate hi-targeted-to-uri entry in the History-Info header	Priv-value
	"*presentation restricted*"		"*history*"
	"*presentation allowed*"		Privacy header field absent or "*none*" (Note 3)
Redirection Information	Redirecting indicator	Privacy header field that corresponds to the penultimate hi-targeted-to-uri entry in the History-Info header	Priv-value
	Call diverted		"*none*" (Note 4)
	Call diverted, all redirection info presentation restricted		"*history*"
Redirection Information	Redirection counter 1	hi-index and mp-param (Note 7)	Number of diversions are sown due to the number of hi-index entries Index for original called number = 1 Address Signals (CdPN)

Table B.6.1-10 – Mapping of IAM to SIP INVITE request

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
			Number = 1.1 and addition "mp=1"
	2		Index for original called number = 1 Index for Redirecting number with Index = 1.1 and addition of "mp=1" Address Signals (CdPN) Number = 1.1.1 and addition of "mp=1.1"
	N		Index for original called number = 1 Placeholder History entry with Index = 1.1 and addition of "mp=1" … Fill up … Index for Redirecting Number with = 1.[(N-1)*".1"] and addition of "mp" set to the hi-index value of the hi-targeted-to-uri that precede. Index for Address Signals (CdPN) = 1.N* ".1" (e.g., N=3 → 1.1.1.1) and addition of mp=1.[(N-1)*].1
Redirection Information	**Redirecting Reason and** **Original Redirection Reason** (Note 1)	hi-targeted-to-uri; "cause" URI parameter, as defined in [IETF RFC 4458]. The Redirecting Reason shall be mapped to the last hi-targeted-to-uri. If the redirection counter is 2 or higher, the Original Redirection Reason shall be mapped to the second hi-targeted-to-uri. If the redirection counter is 3 or higher, for each hi-targeted-to-uri following a placeholder History entry the value "404" shall be taken (Note 2)	**cause value**
	unknown		404
	unconditional		302
	User Busy		486
	No reply		408
	Deflection during alerting		487
	Deflection immediate response		480
	Mobile subscriber not reachable		503
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

Table B.6.1-10 – Mapping of IAM to SIP INVITE request

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
Original Called 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 (NOTE 6)
Original Called 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 Redirection Reason contains only the "unknown" parameter.

NOTE 2 – For all History entries except the first one a "cause" URI parameter as defined in [IETF RFC 4458] has to be included.

NOTE 3 – If the Redirecting Indicator has the value "Call diverted, all redirection info presentation restricted", the privacy value "history" shall be set.

NOTE 4 – If the Redirecting Number APRI has the value "presentation restricted", the privacy value "history" shall be set.

NOTE 5 – Used URI scheme shall be SIP URI. The "cause" URI parameter cannot be added if hi-targeted-to-uri is a tel URI.

NOTE 6 – The used URI scheme can be tel URI or SIP URI since the first hi-targeted-to-uri entry of the History-Info header field does not contain the "cause" URI parameter.

NOTE 7 – The hi-target-param defined in [IETF RFC 7044] defines the mp-param "mp" as a header field parameter that contains the value of the hi-index in the hi-entry with a hi-targeted-to-uri that reflects the Request-URI that was retargeted, thus identifying the "mapped from" target. Since the hi-entries are created based on the redirection counter to reflect the diverting/diverted-to entries, the hi-target-param "mp" shall be present in each entry except the first one.

B.6.1.3 Interworking at the I-IWU

For profiles A and B:

Table B.6.1-11 – Mapping of SIP to ISUP messages

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

Table B.6.1-12 – 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
In History-Info SIP header field, hi-targeted-to-uri in hi-entry with a hi-index that matches the mp-param value of the		Redirecting number	

Table B.6.1-12 – 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
last hi-entry containing a mp-param and a "cause" URI parameter, as defined in [IETF RFC 4458]. (Note 1) (Note 3)			
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) numbe*r" 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 or/and priv-value component of the hi-entry in History-Info header field specified in this table (NOTE 2)		APRI	If the priv-value "*history*" or "*session*" or "*header*" is received within the Privacy header field or if the priv-value "*history*" is received within the "headers" component of the hi-targeted-to-uri in hi-entry before last hi-entry containing a "cause" URI parameter then "*presentation restricted*" shall be set. Otherwise, "*presentation allowed"* shall be set.
NOTE 1 – If the SIP URI does not contain "user=phone", mapping to the 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. NOTE 3 – The hi-target-param parameter set to "mp" as defined in [IETF RFC 7044] indicates that the target of the Request-URI was changed and appears in the hi-targeted-to-uri. In case of interworking with networks not supporting [IETF RFC 7044] the "mp" header field parameter may not appear. If the "mp" header field parameter is missing in the last hi-entry containing a "cause" URI parameter as defined in [IETF RFC 4458], the hi-entry to use is the entry just before.			

Table B.6.1-13 – Mapping of History-Info header to ISUP Redirection Information

Source SIP header field and component	Source Component value	Redirection Information	Derived value of parameter field
Privacy header field and priv-value of hi-entry with	"*history*" or "*session*" or "*header*"	Redirecting Indicator	*Call diverted, all redirection info presentation restricted*

Table B.6.1-13 – Mapping of History-Info header to ISUP Redirection Information

Source SIP header field and component	Source Component value	Redirection Information	Derived value of parameter field
a hi-index that matches the mp-param value of the last hi-entry containing a mp-param and a "cause" URI parameter as defined in IETF RFC 4458 of the History-Info header field. (Note 2)	for the Privacy header field or for the hi-targeted-to-uri entry		
	Privacy header field and the privacy component of the hi-targeted-to-uri entry either absent or "*none*"		*Call diverted*
		Original redirection reason	*Unknown*
Cause value in the last hi-targeted-to-uri containing a "cause" URI parameter as defined in [IETF RFC 4458] (NOTE 2)	**cause value**	Redirecting Reason	**Redirecting Reason**
	404		*Unknown/not available*
	302		*Unconditional*
	486		*User busy*
	408		*No reply*
	480		*Deflection immediate response*
	487		*Deflection during alerting*
	503		*Mobile subscriber not reachable*
Hi-index		Redirection counter	Number of History entries containing a "cause" URI parameter with value as listed in the "cause" URI parameter row in this table (Note 1)
NOTE 1 – If the determined number of redirection in SIP exceeds the ISUP maximum parameter value, the IWU shall set the Redirection counter to its maximum value. For instance, in ISUP [ITU-T Q.763], the Redirection counter parameter cannot exceed 5. NOTE 2 – The hi-target-param parameter set to "mp" as defined in [IETF RFC 7044] indicates that the target of the Request-URI was changed and appears in the hi-targeted-to-uri. In case of interworking with networks not supporting [IETF RFC 7044] the "mp" header field parameter may not appear. If the "mp" header field parameter is missing in the last hi-entry containing a "cause" URI parameter as defined in [IETF RFC 4458], the hi-entry to use is the entry just before.			

Table B.6.1-14 – 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 (ITU-T E.164)*"

Table B.6.1-14 – 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
hi-targeted-to-uri of hi-entry with a hi-index that matches the mp-param value of the 1st hi-targeted-to-uri containing a mp-param and a "cause" URI parameter, as defined in [IETF RFC 4458]; the global number portion of the URI, is assumed to be in form "+" CC + NDC + SN (Note 1) (Note 3)	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 does not contain "user=phone", mapping to the 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.

NOTE 3 – The hi-target-param parameter set to "mp" as defined in [IETF RFC 7044] indicates that the target of the Request-URI was changed and appears in the hi-targeted-to-uri. In case of interworking with networks not supporting [IETF RFC 7044] the "mp" header field parameter may not appear. If the "mp" header field parameter is missing in the last hi-entry containing a "cause" URI parameter as defined in [IETF RFC 4458], the hi-entry to use is the entry just before.

Table B.6.1-15 – Mapping of INVITE to IAM

INVITE		IAM	
History-Info header field	See Table B.6.-12	Redirecting Number	See Table B.6.1-12
History-Info header field	See Table B.6.1-13	Redirection Information	See Table B.6.1-13
History-Info header field	See Table B.6.1-14	Original Called Number	See Table B.6.1-14

Table B.6.-16 – Mapping of ISUP to SIP Messages

←Message sent to SIP	←Message Received from BICC/ISUP	
181 (Being forwarded)	ACM no indication with Redirection number and call diversion information (CFU, CFB, Cdi)	See Table B.6.1-18
180 (Ringing)	ACM indicating ringing, oBCi: Call diversion may occur (CFNR, Cda)	See clause 6.5.1.1
181 (Being forwarded)	CPG indicating progress or subsequent diversion indicated in the CPG with Redirection number and call diversion information (CFNR, Cda)	See Table B.6.1-19
180 (Ringing)	CPG indicating ringing and Redirection number restriction parameter	See Table B.6.1-20
200 (OK)	ANM and Redirection number restriction parameter	See Table B.6.1-21

Table B.6.1-17 – Mapping of ISUP Redirection Number Restriction to History-Info header field

Redirection Number Restriction	Derived value of parameter field	SIP component	Value
Presentation restricted indicator	"*Presentation restricted*"	privacy "headers" component of the hi-targeted-to-uri	"*History*"
	"*Presentation allowed*" or absent AND any previous received notification subscription option was NOT "*presentation not allowed*" AND was NOT "*presentation allowed without redirection number*"		Privacy header field absent or "*none*"

Table B.6.1-18 – Mapping of ACM → 181 (Call Is Being Forwarded) response

ISUP Parameter	Derived value of parameter field	SIP component	Value
Generic Notification indicators	Call is diverting		
Redirection number		History-Info header field with one hi-entry and the hi-target-param set to "mp"	hi-targeted-to-uri:
Nature of Address Indicator:	"*national (significant) number"*	hi-targeted-to-uri	Add CC (of the country where the IWU is located) to Redirection number Address Signals to construct E.164 number in URI.

Table B.6.1-18 – Mapping of ACM → 181 (Call Is Being Forwarded) response

ISUP Parameter	Derived value of parameter field	SIP component	Value
	"international number"		Map complete Redirection number Address Signals to E.164 number in URI.
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	Addr-spec "+" CC NDC SN mapped to userinfo portion of SIP URI. (Note 2) Add "user=phone".
Call Diversion Information	**Redirecting Reason**	IETF RFC 4458 "cause" URI parameter in the hi-targeted-to-uri (Note 1)	**cause value**
	Unknown/not available		404
	Unconditional		302
	User busy		486
	No reply		408
	Deflection immediate response		480
	Deflection during alerting		487
	Mobile subscriber not reachable		503
	Notification subscription option	privacy "headers" component of the hi-targeted-to-uri (Note 1)	**Roles**
	unknown		Escaped Privacy value is set according to the rules of [ITU-T Q.3620] clause 4.5.2.6.4 item c
	presentation not allowed		A 181 Being Forwarded shall **not** be sent
	presentation allowed with redirection number		Escaped Privacy value is set according to the rules of [ITU-T Q.3620] clause 4.5.2.6.4 item c
	presentation allowed without redirection number		Escaped Privacy value is set according to the rules of [ITU-T Q.3620] clause 4.5.2.6.4 item c
NOTE 1 – Needs to be stored for a possible inclusion into subsequent messages. NOTE 2 – Used URI scheme shall be SIP URI. The "cause" URI parameter cannot be added if hi-targeted-to-uri is a tel URI.			

Table B.6.1-19 – Mapping of CPG → 181 (Call Is Being Forwarded) response

ISUP Parameter	Derived value of parameter field	SIP component	Value
Event Indicator	Progress		
Generic Notification Indicators	Call is diverting		
Redirection Number		History-Info header field with one hi-entry and the hi-target-param set to "mp"	hi-targeted-to-uri:
Nature of Address Indicator	"*national (significant) number*"	hi-targeted-to-uri	Add CC (of the country where the IWU is located) to Redirection number Address Signals to construct E.164 number in URI.
	"international number"	hi-targeted-to-uri	Map complete Redirection number Address Signals to E.164 number in URI.
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	Addr-spec "+" CC NDC SN mapped to userinfo portion of SIP URI. (Note 2) Add "user=phone".
Call Diversion Information	**Redirecting Reason**	IETF RFC 4458 "cause" URI parameter in the hi-targeted-to-uri (Note 1)	**cause value**
	Unknown/not available		404
	Unconditional		302
	User busy		486
	No reply		408
	Deflection immediate response		480
	Deflection during alerting		487
	Mobile subscriber not reachable		503
	Notification subscription option	privacy "headers" component of the hi-targeted-to-uri (Note 1)	**Roles**
	unknown		Escaped Privacy value is set according to the rules of [ITU-T Q.3620] clause 4.5.2.6.4 items c
	presentation not allowed		A 181 Being Forwarded shall not be sent

Table B.6.1-19 – Mapping of CPG → 181 (Call Is Being Forwarded) response

ISUP Parameter	Derived value of parameter field	SIP component	Value
	presentation allowed with redirection number		Escaped Privacy value is set according to the rules of [ITU-T Q.3620] clause 4.5.2.6.4 items c
	presentation allowed without redirection number		Escaped Privacy value is set according to the rules of [ITU-T Q.3620] clause 4.5.2.6.4 items c
NOTE 1 – Needs to be stored for a possible inclusion into subsequent messages. NOTE 2 – Used URI scheme shall be SIP URI. The "cause" URI parameter cannot be added if hi-targeted-to-uri is a tel URI.			

Table B.6.1-20 addresses two separate conditions: the CPG is received from the diverting exchange in which case the Call diversion information is included; and the CPG is received from the diverted-to exchange in which case the Call diversion information is not included. Interworking for both conditions is shown.

Table B.6.1-20 – Mapping of CPG → 180 (Ringing) response

ISUP Parameter	Derived value of parameter field	SIP component	Value
Event Indicator	Alerting		
Redirection Number		History-Info header field with one hi-entry and the hi-target-param set to "mp"	See Table B.6.1-18
Call Diversion Information	**Redirecting Reason**	[IETF RFC 4458] "cause" URI parameter in the hi-targeted-to-uri (Note 1)	**cause value**
	Unknown/not available		404
	Unconditional		302
	User busy		486
	No reply		408
	Deflection immediate response		480
	Deflection during alerting		487
	Mobile subscriber not reachable		503
	Notification subscription option	privacy "headers" component of the hi-targeted-to-uri (Note 1)	**Roles**
	unknown		Escaped Privacy value is set according to the rules of [ITU-T Q.3620] clause 4.5.2.6.4 item c

Table B.6.1-20 – Mapping of CPG → 180 (Ringing) response

ISUP Parameter	Derived value of parameter field	SIP component	Value
	presentation not allowed		The 180 Ringing response shall be sent without the History-Info header field included
	presentation allowed with redirection number		Escaped Privacy value is set according to the rules of [ITU-T Q.3620] clause 4.5.2.6.4 item c
	presentation allowed without redirection number		Escaped Privacy value is set according to the rules of [ITU-T Q.3620] clause 4.5.2.6.4 item c
If no Call Diversion Information parameter is present		IETF RFC 4458 "cause" URI parameter in the hi-targeted-to-uri	Value stored from a previous received ACM or CPG. See Tables B.6.2-8 and B.6.2-9
		privacy "headers" component of the hi-targeted-to-uri	Value stored from a previous received ACM or CPG. See Tables B.6.2-8 and B.6.2-9
Redirection Number Restriction (NOTE 2)			See Table B.6.1-17

NOTE 1 – Needs to be stored for a possible inclusion into subsequent messages.

NOTE 2 – This parameter may appear without the Call Diversion Information parameter and Redirection number. In such cases, the O-IWU shall send the previously sent and stored History-Info header field and set the priv-value in the History-Info header field as described in Table B.6.-17.

Table B.6.1-21 – Mapping of ANM → 200 (OK) response (to INVITE request)

ISUP Parameter	Derived value of parameter field	SIP component	Value
Redirection Number		History-Info header field with one hi-entry and the hi-target-param set to "mp"	See Table B.6.1-18
		[IETF RFC 4458] "cause" URI parameter in the hi-targeted-to-uri	cause value= as stored from a previous received ACM or CPG. See Tables B.6.2-8 and B.6.2-9
Redirection Number Restriction (Note)			See Table B.6.1-17

NOTE – This parameter may appear without the Call Diversion Information parameter and Redirection number. In such cases, the O-IWU shall send the previously sent and stored History-Info header field and set the priv-value in the History-Info header field as described in Table B.6.1-17.

B.6.2 Profile C (SIP-I)

Call forwarding in the PSTN requires no additional interworking beyond the use of ISUP encapsulation.

B.7 Interworking of Explicit Call Transfer (ECT) supplementary service to SIP networks

Profiles A and B

When the IWU receives a FAC message with Generic notification indicator coded as "Call transfer active" or "call transfer alerting" and a CPG with Generic notification indicator coded as "Remote hold" was received previously for the current communication, the action described in Table B.7-1 applies. In all other cases the actions of the IWU at the ISUP/BICC side are described in [ITU-T Q.732.7] under the clause "Interactions with other networks".

Table B.7-1 – Mapping between ISUP and SIP for the Explicit Communication Transfer supplementary service

ISUP message	Mapping
FAC with a "call transfer, active" or "call transfer, alerting" Generic notification indicator	As described for CPG message with a "remote retrieval" Generic notification indicator in clause 7.4.10.2

Profile C (SIP-I)

No additional interworking beyond the use of ISUP encapsulation.

B.8 Interworking of Call Waiting (CW) supplementary service to SIP networks

B.8.1 General

For profiles A and B: If the IWU supports the interworking of the IMS Communication Waiting service the clauses B.8.2 to B.8.3 applies.

The protocol specification of the Communication Waiting supplementary service is described in [ITU-T Q.3622].

B.8.2 Interworking at the I-IWU

With regard to the backward messages of the Call Waiting PSTN/ISDN supplementary service, the following mapping is valid:

Table B.8-1 – Mapping of ISUP messages to SIP Massages

→Message Received from BICC/ISUP	→Message sent to SIP
ACM or CPG with generic notification indicator "Call is a waiting call" (Note 1).	180 Ringing with an Alert-Info header field set to "urn:alert:service:call-waiting" (Note 2).
NOTE 1 – The coding shall be in accordance with [ITU-T Q.733]. NOTE 2 – The coding shall be in accordance with [IETF RFC 7462].	

B.8.3 Interworking at the O-IWU

With regard to the backward messages of the Communication Waiting service, the following mapping is valid:

Table B.8-2 – Mapping of SIP messages to ISUP messages

←Message sent to ISUP	←Message Received from SIP
ACM or CPG with generic notification indicator "Call is a waiting call" (Note 1).	180 Ringing with an Alert-Info header field set to "urn:alert:service:call-waiting" (Note 2).
NOTE 1 –The coding shall be in accordance with [ITU-T Q733]. NOTE 2 – The coding shall be in accordance with [IETF RFC 7462].	

B.8.3 No support of Communication Waiting (CW) service

If the IWU does not support the interworking of the IMS Communication Waiting (CW) service the IWU shall act in accordance with the procedures described within clause 1.7 of [ITU-T Q.733.1], under the heading "Interactions with other networks".

B.8.4 Profile C (SIP-I)

No additional interworking beyond the use of ISUP encapsulation is required.

B.9 Interworking of Call Hold (HOLD) supplementary service to SIP networks

Profiles A and B

Call Hold is defined as an ISUP supplementary service within [ITU-T Q.733.2].

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 Generic Notification Indicator parameter is used to send the appropriate notification towards the remote party.

The following notification descriptions are used:

– *"remote hold"*;
– *"remote retrieval"*.

The Event Indicator is set to *"progress"*.

The same service is also available within SIP networks and is defined in [IETF 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 B.9-1.

Table B.9-1 – 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 retrieval"*	↔	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"*	↔ (Note 1)	UPDATE with the attribute line "a=sendonly" or "a=inactive" for the offered media stream (see above) (Note 2)
before answer	CPG with *"remote retrieval "*	↔ (Note 1)	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 1 – For the "before answer" scenarios, mapping applies only for hold requests sent by the calling party to the called party as the called party cannot put the calling party on hold before answer.

NOTE 2 – If an additional early dialogue is established during the "*remote hold*" condition the IWU shall send an UPDATE request containing an SDP offer with "*sendonly*" or "*inactive*" media on the new early dialogue, as described in [IETF RFC 3264].

B.9.1 Additional interworking requirements

For profiles A and B: When an IWU receives a CPG message with a *"remote hold"* Generic notification indicator and the media on the IMS side are "*sendrecv*" or "*recvonly*", the IWU shall forward the hold request by sending an UPDATE request on the early dialogue which was last established containing an SDP offer with "*sendonly*" if the stream was previously "*sendrecv*" or "*inactive*" if the stream was previously "*recvonly*" media, as described in [IETF RFC 3264].

If an additional early dialogue is established during the "*remote hold*" condition the IWU shall send an UPDATE request containing an SDP offer with "*sendonly*" or "*inactive*" media on the new early dialogue, as described in [IETF RFC 3264].

If an UPDATE request with an SDP offer is received on one of the early dialogues for a call in the "*remote hold*" condition the IWU shall send an appropriate SDP answer followed by a new UPDATE request including SDP with "*sendonly*" or "*inactive*" media on the dialogue, as described in [IETF RFC 3264].

If an IWU receives a 200 OK (INVITE) response on an early dialogue for which the call is in a "*remote hold*" condition the IWU shall send an UPDATE or re-INVITE request containing an SDP offer with "*sendonly*" or "*inactive*" media on the dialogue where 200 OK (INVITE) was received, as described in [IETF RFC 3264]. If received SDP is indicating "*sendonly*" and the Contact header field of the remote party contained an "*isfocus*" header field parameter, defined in [IETF RFC 3840], the IWU shall order the MGW to disconnect the media towards the IMS.

If the IWU receives a CPG with Generic Notification Indicator "*remote retrieval*" and there is an early dialogue on IMS side then a SIP UPDATE request (indicating call retrieval) shall be sent if the call hold service had been invoked on the early dialogue before. For each subsequent early dialogue for which the IWU receives an 18x response or an UPDATE request with an SDP offer, the IWU shall send SIP UPDATE indicating call retrieval after a possible SDP answer to the SDP offer, if that dialogue had received a call hold indication before.

If the IWU receives a CPG with Generic Notification Indicator "*remote retrieval*" and there is a confirmed dialogue on IMS side then a SIP re-INVITE shall be sent for this dialogue only if the call hold service had been invoked for this dialogue before. If the media path towards the IMS was disconnected due to an "*isfocus*" header field parameter, defined in [IETF RFC 3840], in the remote party Contact header field, media shall be resumed.

When an IWU receives a CPG message with a "*remote retrieval*" Generic Notification Indicator and the media on the IMS side are "*sendonly*" or "*inactive*", the IWU shall forward the resume request by sending an UPDATE or re-INVITE message containing an SDP offer with "*sendrecv*" if the stream was previously "*sendonly*" or "*recvonly*" if the stream was previously "*inactive*" media, as described in [IETF RFC 3264]. If the media path towards the IMS was disconnected due to an "*isfocus*" header field parameter, defined in [IETF RFC 3840], in the remote party Contact header field, media shall be resumed.

If the IWU receives a CPG with "*remote hold*" or "*remote retrieval*" before answer, it shall forward the request using an UPDATE message. If the IWU receives a CPG with "*remote hold*" or "*remote retrieval*" after answer, it should forward the request using re-INVITE.

The interworking does not impact the user plane with the following exceptions:

– the IWU provides modified SDP RR and RS bandwidth modifiers within the UPDATE or re-INVITE messages;

– the Contact header of the IMS remote party contains the "*isfocus*" header field parameter; or

– If the IWU provides modified SDP RR and RS bandwidth modifiers to the IMS side, the IWU shall also provide modified SDP RR and RS bandwidths to the MGW, as described in the TS [ITU-T Q.3629] clause 9.2.10. If the Contact header of the IMS remote party contains the "*isfocus*" header field parameter, defined in [IETF RFC 3840], media shall not be provided towards the IMS network.

Profile C (SIP-I)

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 B.9-2.

Table B.9-2 – 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 retrieval"* CPG with *"remote retrieval"* 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	→ (Note) ←	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 retrieval"* CPG with *"remote retrieval"* extracted from the body of the SIP message	→ (Note) ←	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.
NOTE – For the "before answer" scenarios, mapping applies only for hold requests sent by the calling party to the called party as the called party cannot put the calling party on hold before answer.

NOTE – The Interworking of the Call Hold (HOLD) Supplementary service between BICC and SIP networks is for further study since BICC CS-2 does not support media suspension.

B.10 Interworking of Completion of Calls to Busy Subscriber (CCBS) and Completion of Calls on No Reply (CCNR) supplementary service to SIP networks

B.10.1 General

If the interworking of the Completion of Communications to Busy Subscriber and Completion of Communications by No Reply supplementary is supported the procedures described in clauses B.10.2 to B.10.3 shall apply. If not, in accordance with the procedures described within [ITU-T Q.733.3], the service shall be terminated at the IWU.

The protocol specification of the Completion of Communications to Busy Subscriber and Completion of Communications by No Reply supplementary services is described in [ITU-T Q.3625].

SIP SUBSCRIBE and NOTIFY methods shall be used in accordance with the procedures defined in [IETF RFC 6665].

B.10.2 Interworking at the I-IWU

For profiles A and B: If the I-IWU supports the interworking of CCBS/CCNR supplementary services, the I-IWU shall map between the SIP and ISUP messages in accordance with Table B.10-1.

Table B.10-1 – Mapping of ISUP and SIP messages

SIP Message	Parameter	ISUP Message	Parameter
← 180 Ringing	Call-Info header field with purpose header field parameter set to "call-completion" and "m" header field parameter set to "NR" (no reply) (Note 1)	← ACM	Called party's status indicator set to "Subscriber free" and CCNR possible indicator set to "CCNR possible" (Note 2)
← 180 Ringing	Call-Info header field with purpose header field parameter set to "call-completion and "m" header field parameter set to "NR" (no reply) (Note 1)	← CPG	Event indicator set to "Alerting" and CCNR possible indicator set to "CCNR possible" (Note 2)
← 486 Busy here	Call-Info header field with purpose header field parameter set to "call-completion" and "m" header field parameter set to "BS" (busy subscriber) (Note 1)	← REL	Cause Indicator cause #17 with Diagnostic (CCBS indicator set to "CCBS possible") (Note 3)
→ INVITE	Request URI contains "m" SIP URI parameter or Call-Info header field contains "purpose" header field parameter set to "call-completion" and "m" header field parameter. (Note 1)	→ IAM	CCSS parameter set to "CCSS call" (Note 2) (Note 4)
NOTE 1 – The coding shall be in accordance with [IETF RFC 6910]. NOTE 2 – The coding shall be in accordance with [ITU-T Q.763]. NOTE 3 – The coding shall be in accordance with [ITU-T Q.850]. NOTE 4 – CCSS parameter set to the value "CCSS call" is included in the IAM if Request-URI contains the SIP URI parameter "m" i.e., creation of the CCSS parameter does not depend on the received value of the SIP URI parameter "m".			

If the I-IWU supports the interworking of CCBS/CCNR supplementary services, the I-IWU shall map between the SIP and TCAP messages in accordance with Tables B.10-2 and B.10-3.

Table B.10-2 – Mapping of SIP messages to TCAP messages

SIP Message	Parameter	TCAP Message	Parameter
SUBSCRIBE with m-parameter in Request URI set to "BS" or containing Call-Info header field with "purpose" parameter set to "call-completion" and m-parameter set to "BS" (Note 1)	Request URI (Note 5)	TC-Begin CCBS REQUEST (invoke)	CalledPartyNumber (Note 3)
	P-Asserted-Identity		CallingPartyNumber (Note 4)
			RetainSupported (Note 2)

Table B.10-2 – Mapping of SIP messages to TCAP messages

SIP Message	Parameter	TCAP Message	Parameter
SUBSCRIBE with m-parameter in Request URI set to "NR" or containing Call-Info header field with "purpose" parameter set to "call-completion" and m-parameter set to "NR" (Note 1).	Request URI (Note 5)	TC-Begin CCNR REQUEST (invoke)	CalledPartyNumber (Note 3)
	P-Asserted-Identity		CallingPartyNumber (Note 4)
			RetainSupported (Note 2)
PUBLISH with m-parameter in Request URI or m-parameter in Call-Info header field set to "BS" and body containing PIDF basic status set to "closed".		TC-Cont CCBS SUSPEND	
PUBLISH with m-parameter in Request URI or m-parameter in Call-Info header field set to "BS" and body containing PIDF basic status set to "open".		TC-Cont CCBS RESUME	
PUBLISH with m-parameter in Request URI or m-parameter in Call-Info header field set to "NR" and body containing PIDF basic status set to "closed".		TC-Cont CCBS SUSPEND	
PUBLISH with m-parameter in Request URI or m-parameter in Call-Info header field set to "NR" and body containing PIDF basic status set to "open".		TC-Cont CCBS RESUME	
SUBSCRIBE with m-parameter in Request URI or m-parameter in Call-Info header field set to "BS" and with Expires header set to "zero".		TC-End CCBS CANCEL	
SUBSCRIBE with m-parameter in Request URI or m-parameter in Call-Info header field set to "NR" and with Expires header set to "zero".		TC-End CCBS CANCEL	

NOTE 1 – Expires header defines subscription duration / CC service duration timer (CC-T3).

NOTE 2 – Parameter is set by default, as retention option is supported in IMS by default.

NOTE 3 – Mapping of the Request URI header field to the CalledPartyNumber is done according to Table 6-2 in clause 6.1.3.1.

NOTE 4 – Mapping of the P-Asserted Identity header field to the CallingPartyNumber is done according to Table 6-13 in clause 6.1.3.6.1.

NOTE 5 – If URI of the IWU was returned in the Call-Info header field in the 180 Ringing or 486 Busy here messages described in Table B.10-1, then the IWU needs to remember the Request URI of the original INVITE for mapping to the CalledPartyNumber.

Table B.10-3 – Mapping of TCAP messages to SIP messages

TCAP Message	Parameter	SIP Message	Parameter
TC-Cont CCBS REQUEST (return result)	RetainSupported	NOTIFY with cc-state parameter set to "queued"	cc-service-retention
TC-End CCBS/CCNR REQUEST (error result)	ShortTermDenial	480 Temporarily unavailable	

Table B.10-3 – Mapping of TCAP messages to SIP messages

TCAP Message	Parameter	SIP Message	Parameter
TC-End CCBS/CCNR REQUEST (error result)	LongTermDenial	403 Forbidden	
TC-End CCBS CANCEL		NOTIFY with the "reason" Subscription-State header field parameter set to "noresource"	
TC-Cont REMOTE USER FREE		NOTIFY with cc-state set to "ready" (Note)	
NOTE – This does not terminate the subscribtion AS-AS.			

B.10.3 Interworking at the O-IWU

For profiles A and B: If the O-IWU supports the interworking of CCBS/CCNR supplementary services, the O-IWU shall map between the ISUP and SIP messages in accordance with Table B.10-4.

Table B.10-4 – Mapping of SIP and ISUP messages

<table>
<tr><th>ISUP Message</th><th>Parameter</th><th>SIP Message</th><th>Parameter</th></tr>
<tr><td>← ACM</td><td>Called party's Status Indicator set to "Subscriber free" and
CCNR Possible Indicator set to "CCNR possible"
(Note 2)</td><td rowspan="2">← 180 Ringing</td><td rowspan="2">Call-Info header field with purpose header field parameter set to "call-completion"
(Note 1)</td></tr>
<tr><td>← CPG</td><td>Event Indicator set to "Alerting" and
CCNR possible indicator set to "CCNR possible"
(NOTE 2) (Note 4)</td></tr>
<tr><td>← REL</td><td>Cause Indicator
cause #17 or #34 with Diagnostic (CCBS indicator set to "CCBS possible")
(Note 3)</td><td>← 486 Busy here</td><td>Call-Info header field with purpose header field parameter set to "call-completion"
(Note 1)</td></tr>
<tr><td>→ IAM</td><td>CCSS parameter set to "CCSS call"
(Note 2)</td><td>→ INVITE</td><td>Request URI contains m-parameter and a Call-Info header field field, with purpose header field parameter set to "call-completion", and an m-parameter
(Note 1) (Note 5)</td></tr>
<tr><td colspan="4">NOTE 1 –: The coding shall be in accordance with [IETF RFC 6910].
NOTE 2 – The coding shall be in accordance with [ITU-T Q.763].
NOTE 3 – The coding shall be in accordance with [ITU-T Q.850].
NOTE 4 – A CPG will be sent if an ACM was already sent.
NOTE 5 – Based on the operator policy the "m" SIP URI parameter in the Request-URI and m-parameter in Call-Info header field is set to the value "BS" or "NR".</td></tr>
</table>

If the O-IWU supports the interworking of CCBS/CCNR supplementary services, the O-IWU shall map between the TCAP and SIP messages in accordance with Table B.10-5.

Table B.10-5 – Mapping of TCAP messages to SIP messages

TCAP Message	Parameter	SIP Message	Parameter
TC-Begin CCBS REQUEST (invoke)	CalledPartyNumber	SUBSCRIBE with m-parameter in Request URI and m-parameter in Call-Info header field set to "BS" (NOTE 1)	To header (Note 2)
			Request-URI (Note 2)
	CallingPartyNumber		From header (Note 3)
			P-Asserted-Identity (Note 4)
TC-Begin CCNR REQUEST (invoke)	CalledPartyNumber	SUBSCRIBE with m-parameter in Request URI and m-parameter in Call-Info header field set to "NR" (NOTE 1)	To header (Note 2)
			Request-URI (Note 2)
	CallingPartyNumber		From header (Note 3)
			P-Asserted-Identity (Note 4)
TC-Cont CCBS SUSPEND		PUBLISH with m-parameter in Request URI and m-parameter in Call-Info header field set to "BS" and body containing PIDF basic status set to "closed".	
		PUBLISH with m-parameter in Request URI and m-parameter in Call-Info header field set to "NR" and body containing PIDF basic status set to "closed".	
TC-Cont CCBS RESUME		PUBLISH with m-parameter in Request URI and m-parameter in Call-Info header field set to "BS" and body containing PIDF basic status set to "open".	
		PUBLISH with m-parameter in Request URI and m-parameter in Call-Info header field set to "NR" and body containing PIDF basic status set to "open".	
TC-End CCBS CANCEL		SUBSCRIBE with m-parameter in Request URI and m-parameter in Call-Info header field set to "BS" and with Expires header set to "0"	
		SUBSCRIBE with m-parameter in Request URI and m-parameter in Call-Info header field set to "NR" and with Expires header set to "0"	

NOTE 1 – Expires header defines subscription duration / CC service duration timer (CC-T3).

NOTE 2 – For the mapping of the CalledPartyNumber to the To header and Request-URI see Table 7-3 in clause 7.1.3.

NOTE 3 – For the mapping of the CallingPartyNumber to the From header see Table 7-7 in clause 7.1.4. If no CallingPartyNumber is available, the From header shall be set to a SIP URI with addr-spec of "unavailable@ unknown.invalid".

NOTE 4 – For the mapping of the CallingPartyNumber to the P-Asserted-Identity header see Table 7-6 in clause 7.1.4. If no CallingPartyNumber is available, a P-Asserted-Identity header shall not be inserted.

Table B.10-6 – Mapping of SIP messages to TCAP messages

SIP Message	Parameter	TCAP Message	Parameter
NOTIFY with cc-state parameter set to "queued"	cc-service-retention	TC Cont CCBS/CCNR REQUEST (return result)	RetainSupported
480 Temporarily unavailable		TC-End CCBS/CCNR REQUEST (error result)	ShortTermDenial
403 Forbidden		TC-End CCBS/CCNR REQUEST (error result)	LongTermDenial
NOTIFY with the Subscription-State header field set to "terminated" and the "reason" parameter set to "noresource"		TC-End CCBS CANCEL	
NOTIFY with cc-state set to "ready" (Note)		TC-Cont REMOTE USER FREE	
NOTE – This does not terminate the subscription AS-AS.			

B.10.4 Profile C (SIP-I)

No additional interworking beyond the use of ISUP encapsulation and SCCP connectivity between originating and terminating ISDN networks required.

B.11 Interworking of Terminal Portability (TP) supplementary service to SIP networks

Profiles A and B

Terminal Portability is defined as an ISUP supplementary service within [ITU-T Q.733.4].

For the Terminal Portability supplementary service, the Suspend and Resume messages containing the Suspend/Resume indicators set to "ISDN subscriber initiated" are used.

The Suspend message indicates a temporary cessation of communication without releasing the call. It can only be accepted during the conversation/data phase. A Resume message indicates a request to recommence communication.

Although there is no similar service in SIP networks, it is appropriate to map the flows for an ISUP Terminal Portability supplementary service onto the flows for Call Hold in SIP networks in order to request media suspension at the remote SIP user agent. A Suspend message containing the Suspend/Resume indicators set to "*ISDN subscriber initiated*" shall be treated like a CPG with "*remote hold*" in Table B.9-1. A Resume message containing the Suspend/Resume indicators set to "*ISDN subscriber initiated*" shall be treated like a CPG with "*remote retrieval*" in Table B.9-1.

Profile C (SIP-I)

Interworking is via the encapsulated SUS and RES messages. No additional interworking is required.

NOTE – The Interworking of Terminal Portability (TP) Supplementary service between BICC and SIP networks is for further study since BICC CS-2 does not support media suspension.

B.12 Interworking of Conference Calling (CONF) supplementary service to SIP networks

B.12.1 No support of conference Call

If the Conference Call (CONF) supplementary service according to [ITU-T Q.3621] is not supported, the IWU shall act in accordance with the procedures described within clause 2.7 of [ITU-T Q.734.1], under the heading "Interactions with other networks".

B.12.2 Support of conference Call

B.12.2.1 General

For profiles A and B: The protocol description of the CONF supplementary service is described in [ITU-T Q.3621]. In this clause the interworking from the conference event package in accordance with [IETF RFC 4575] to the messages of the PSTN/ISDN CONF supplementary service is described. Note that an interworking from the PSTN/ISDN to the IMS is out of the scope of this Recommendation.

B.12.2.2 Subscribing for the conference event package

For profiles A and B: Based on local policy, the IWU may subscribe for the conference event package on behalf of the PSTN/ISDN participant after the participant joins or is added to a conference.

When the conference event package option is implemented, and one of the following events occurs at the IWU:

– a 200 (OK) response is received as a response to an initial INVITE request originated by the IWU, where the Contact header field contains an "isfocus" parameter; or

– an ACK message is received which acknowledges a 200 (OK) response to the initial INVITE request, and the initial INVITE request is originated by the conferencing AS and contains an "isfocus" parameter in the Contact header field;

then the following steps shall be performed:

1) a SUBSCRIBE request shall be created according to [IETF RFC 4575], using the updated procedures from [IETF RFC 6665];

2) the Request URI is set to the Contact address of the conferencing AS;

3) the P-Asserted-Identity header field, the From header field and the Privacy header field are set with the same value as:
   - the P-Asserted-Identity header field, the From header field and the Privacy header field in the initial INVITE request originated by the IWU; or
   - the P-Asserted-Identity header field, the To header field and the Privacy header field in a 1xx or 2xx response sent by the IWU to the initial INVITE request from the conferencing AS.

B.12.2.3 Interworking the notification

NOTE – There is a need to differentiate between the procedures of interworking for a full and a partial type of notification.

For profiles A and B: When a full type of notification is received a check is made of the content. If the changes with respect to a previous version of the notification have not been sent on to the PSTN/ISDN for this session, the IWU shall perform an ISUP interaction towards the PSTN/ISDN. If the changes with respect to a previous version of the notification have been sent to the PSTN/ISDN for this session, the IWU shall not perform an ISUP interaction towards the PSTN/ISDN.

When a partial notification is received then it is assumed that a value of a received notification has changed, so the IWU performs an ISUP interaction towards the PSTN/ISDN, as follows:

– Conference established:

  Upon receipt of a conference information document with the <conference-state-type> element *active* set to "true", the IWU shall send a CPG message to the PSTN/ISDN with a notification "*conference established*".

– Participant added:

  Upon receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* set to "connected", and it was not set to "on-hold" before and the Contact URI in the element *entity* is not the address of the served PSTN/ISDN participant, the IWU shall send a CPG message to the PSTN/ISDN with a notification "*other party added*".

– Served PSTN/ISDN participant isolated:

Upon receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* set to "on-hold", and it was set to "connected" before and the Contact URI in the element *entity* is the address of the served PSTN/ISDN participant, the IWU shall send a CPG message to the PSTN/ISDN with a notification "*isolated*".

– Other participant isolated:

Upon receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* set to "on-hold", and it was set to "connected" before and the Contact URI in the element *entity* is not the address of the served PSTN/ISDN participant, the IWU shall send a CPG message to the PSTN/ISDN with a notification "*other party isolated*".

– Served PSTN/ISDN participant reattached:

Upon receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* set to "connected", and it was set to "on-hold" before and the Contact URI in the element *entity* is the address of the served PSTN/ISDN participant, the IWU shall send a CPG message to the PSTN/ISDN with a notification "*reattached*".

– Other participant reattached:

Upon receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* set to "connected", and it was set to "on-hold" before and the Contact URI in the element *entity* is not the address of the served PSTN/ISDN participant, the IWU shall send a CPG message to the PSTN/ISDN with a notification "*other party reattached*".

– Other party disconnected:

Upon receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* set to "disconnected", and the element *joining-method of joining-type* is not set to "focus-owner", the IWU shall send a CPG message to the PSTN/ISDN with a notification "*other party disconnected*".

B.12.3 Profile C (SIP-I)

No additional interworking beyond the use of ISUP encapsulation is required.

B.13 Interworking of Three-Party Service (3PTY) supplementary service to SIP networks

For profiles A and B

The default behaviour of the IWU at the ISUP/BICC side is described in [ITU-T Q.734] under the clause "Interactions with other networks". In addition, the IWU shall apply the interworking from ISUP to SIP described in Table B.13-1.

Alternatively, the IWU may in addition apply the interworking to the Conference supplementary service described in clause B.12.2.

Table B.13-1 – Mapping between ISUP and SIP for the Conference Calling (CONF) and Three-Party Service (3PTY) supplementary service

ISUP message	Mapping
CPG with a "Conference established" Generic Notification Indicator	As described for CPG message with a "remote retrieval" Generic Notification Indicator in Clause 7.4.10.2
CPG with a "Conference disconnected" Generic Notification Indicator	As described for CPG message with a "remote retrieval" Generic Notification Indicator in clause 7.4.10.2
CPG with an "isolated" Generic Notification Indicator	As described for CPG message with a "remote hold" Generic Notification Indicator in clause 7.4.10.2
CPG with a "reattached" Generic Notification Indicator	As described for CPG message with a "remote retrieval" Generic Notification Indicator in clause 7.4.10.2

Profile C (SIP-I)

No additional interworking beyond the use of ISUP encapsulation is required.

B.14 Interworking of Closed User Group (CUG) supplementary service to SIP networks

B.14.1 General

The protocol specification of the Closed User Group supplementary service is described in [ITU-T Q.3627].

If the interworking of the Closed User Group supplementary service is not supported: The IWU shall act in accordance with the procedures described within clause 1.5.2.4.2 of [ITU-T Q.735.1], under the heading "Exceptional procedures".

B.14.2 Interworking at the I-IWU

For profiles A and B: If the I-IWU supports the interworking of the CUG supplementary service, the I-IWU shall map between the SIP and ISUP messages in accordance with Table B.14-1.

Table B.14-1 – Mapping of SIP messages to ISUP messages

SIP Message	ISUP Message
INVITE request containing a XML cug body with CUG XML schema	IAM containing the Closed User Group Interlock Code parameter and the Closed User Group Call Indicator of the Optional Forward Call Indicator parameter

If the IWU supports the interworking of the CUG supplementary service, the I-IWU shall interwork the CUG XML schema with the ISUP Closed User Group Interlock Code parameter and the *Closed user group call indicator* of the Optional Forward Call Indicator parameter in accordance with Tables B.14-2 and B.14-3.

Table B.14-2 – Mapping of the SIP XML CUG Element to the ISUP Closed User Group Interlock Code parameter

CUG XML Element	Source component value	ISUP Closed user group interlock code Parameter	derived value of parameter field
networkIndicator	networkIdentityType = 4 hexbinary coded digits (NOTE)	"Network Identity"	Octet 1 & Octet 2 including 4 binary coded digits derived from XML Network Identity
cugInterlockBinaryCode	sixteenbitType = 16 bit coded value	"Binary Code"	Octet 3 & Octet 4 including a 16 bit Binary Code derived from the XML Binary Code
NOTE – ISUP Closed User Group Interlock Code parameter Octet 1 contains "Network Identity" (NI) digits 1 & 2 and Octet 2 contains "Network Identity" digits 3 & 4.The networkidentityType is filled with "Octet1 & Octet 2" = "NI digit 1, NI digit 2, NI digit 3, NI digit 4". Example: Digit 1=0, Digit 2=4, Digit 3=9, Digit 4=0 so the networkidentityType is encoded with "0490".			

Table B.14-3 – Mapping of the SIP XML CUG Element to the ISUP Closed User Group Call Indicator included in the optional Forward Call Indicator parameter

CUG XML Element	Source component value	ISUP "Optional Forward Call Indicator" Parameter	derived value of parameter field
cugCommunicationIndicator	Type=00	"closed user group call" indicator	non-CUG call
	Type=01		spare
	Type=10		closed user group call, outgoing access allowed
	Type=11		closed user group call, outgoing access not allowed

If the I-IWU supports the interworking of the CUG supplementary service, then if an INVITE request with the MIME including a cug XML element is received and the terminating network is not supporting CUG, the I-IWU shall behave as shown in Table B.14-4.

Table B.14-4 – Action at the I-IWU with a PSTN/ISDN network without CUG capability

cugCommunicationIndicator in INVITE request	Action at the I-IWU
Type=11 (CUG without outgoing access)	Release the communication with 403
Type=10 (CUG with outgoing access)	Treat the communication as an ordinary call (Note)
Non-CUG	Treat the communication as an ordinary call
NOTE – The cugCommunicationIndicator shall not be mapped or if appropriate the CUG call indicator of the optional forward call indicator shall be set to non-CUG call.	

B.14.3 Interworking at the O-IWU

For profiles A and B: If the IWU supports the interworking of CUG supplementary service, the O-IWU shall map between the SIP and ISUP messages in accordance with Table B.14-5.

Table B.14-5 – Mapping of ISUP messages to SIP messages

ISUP Message	SIP Message
IAM containing the Closed User Group Interlock Code parameter and the Closed User Group Call Indicator of the Optional Forward Call Indicator parameter	INVITE request containing a XML cug body with CUG XML schema

If the IWU supports the interworking of the CUG supplementary service, the IWU shall interwork the CUG XML schema with the ISUP Closed User Group Interlock Code parameter and the *Closed user group call indicator* of the Optional Forward Call Indicator parameter in accordance with Tables B.14-6 and B.14-7.

Table B.14-6 – Mapping of the ISUP Closed User Group Interlock Code to SIP XML CUG element

ISUP Closed User Group Interlock Code parameter	Source component value	CUG XML Element	derived value of parameter field
"Network Identity"	Octet 1 & Octet 2 including 4 binary coded digits	networkIndicator	networkIdentityType = 4 hexbinary coded digits derived from Network Identity (Note)
"Binary Code"	Octet 3 & Octet 4 including a 16 bit Binary Code	cugInterlockBinaryCode	sixteenbitType = 16 bit coded value derived from Binary Code
NOTE – ISUP Closed user group interlock code Parameter Octet 1 contains "Network Identity" (NI) digits 1 & 2 and Octet 2 contains "Network Identity" digits 3 & 4.The networkidentityType shall be filled with "Octet1 & Octet 2" = "NI digit 1, NI digit 2, NI digit 3, NI digit 4". Example: Digit 1=0, Digit 2=4, Digit 3=9, Digit 4=0 so the networkidentityType is encoded with "0490".			

Table B.14-7 – Mapping of the ISUP Closed User Group Call Indicator to SIP XML CUG element

ISUP Optional Forward Call Indicator parameter	Source component value	CUG XML Element	derived value of parameter field
Closed User Group Call Indicator	non-CUG call	cugCommunicationIndicator	Type=00
	spare		Type=01
	closed user group call, outgoing access allowed		Type=10
	closed user group call, outgoing access not allowed		Type=11

If the IWU supports the interworking of the CUG supplementary service, but the IMS is not supporting CUG, the procedures described in [ITU-T Q.735.1] shall apply if an INVITE request with the MIME body including a cug XML element is sent and the O-IWU supports CUG supplementary service.

B.14.4 Profile C (SIP-I)

No additional interworking beyond the use of ISUP encapsulation is required.

B.15 Interworking of Multi-Level Precedence and Preemption (MLPP) supplementary service to SIP networks

Profiles A and B

The IWU shall act in accordance with the procedures described within clause 3.7 of [ITU-T Q.735.3], under the heading "Interactions with other networks".

Profile C (SIP-I)

No additional interworking beyond the use of ISUP encapsulation is required.

B.16 Interworking of Global Virtual Network Service (GVNS) supplementary service to SIP networks

Profiles A and B

The IWU shall act in accordance with the procedures described within clause 6.7 of [ITU-T Q.735.6], under the heading "Interactions with other networks".

Profile C (SIP-I)

No additional interworking beyond the use of ISUP encapsulation is required.

B.17 Interworking of International Telecommunication Charge Card (ITCC) supplementary service to SIP networks

Profiles A and B

The IWU shall act in accordance with the procedures described within clause 1.7 of [ITU-T Q.736.1], under the heading "Interactions with other networks".

Profile C (SIP-I)

SCCP connectivity between originating and terminating ISDN networks is needed. This connectivity could be available as a bypass to the SIP network.

All parameters can be taken from the encapsulated ISUP MIME.

Interworking of ITCC without SCCP by-pass is FFS.

B.18 Interworking of Reverse Charging (REV) supplementary service to SIP networks

Profiles A and B

The IWU shall act in accordance with the procedures described within clause 3.7 of [ITU-T Q.736.3], under the heading "Interactions with other networks".

Profile C (SIP-I)

No additional interworking beyond the use of ISUP encapsulation is required.

B.19 Interworking of User-to-User Signalling (UUS) supplementary service to SIP networks

Profiles A and B

B.19.1 General

If the interworking of the User-to-User header field is defined within [IETF RFC 7433] is not supported: The IWU shall act in accordance with the procedures described within [ITU-T Q.737.1], under the heading "Interactions with other networks".

The coding of the User-user information element is described within [ITU-T Q.931]. The User-user information element is carried within the ISDN User Part parameter user-to-user information, as defined in [ITU-T Q.763]. The User-to-User header field is defined within [IETF RFC 7433]. A package for interworking user-to-user information with the ISDN is defined by [IETF RFC 7434].

B.19.2 User-to-user information Interworking from SIP to ISUP

For profiles A and B: On the receipt of a User-to-User header field with the "*purpose*" header field parameter set to "*isdn-uui*", or a User-to-User header field without a "*purpose*" parameter, with "*encoding*" header field parameter set to "*hex*" or without an "*encoding*" parameter, with "*content*" header field parameter set to "*isdn-uui*" or without a "*content*" parameter, that is valid as defined by [IETF RFC 7434], the IWU shall map the content of the "*uui-data*" field to the "*protocol discriminator*" and "*user information*" parameters of the User-user information element.

The "*length of user-user contents*" parameter shall be set by the IWU according to the normal procedures.

The IWU maps the messages transporting the user-to-user information according to the normal interworking procedures (see Table B.19-1).

Table B.19-1 – Mapping of the User-to-User header field to the ISUP User-to-user Information parameter

SIP parameter →		→ ISUP parameter	
SIP header field	**Source component value**	**ISUP parameter name**	**ISUP parameter field**
User-to-User	uui-data	User-to-user	Protocol discriminator and user information

B.19.3 User-to-user information Interworking from ISUP to SIP

For profiles A and B: On the receipt of the user-to-user information parameter the IWU shall map the protocol discriminator and user information parameter fields to the uui-data field of the User-to-User header field (see Table B.19-2).

If sent, the "*purpose*", "*content*" and "*encoding*" header field parameters are not mapped and are set in accordance with [IETF RFC 7434].

The IWU maps the messages transporting the User-to-user Information parameters according to the normal interworking procedures.

Table B.19-2 – Mapping of the ISUP User-to-user Information parameter to the User-to-User header field

→ ISUP parameter		→ SIP parameter	
ISUP parameter name	**ISUP parameter field**	**SIP header field**	**Source component value**
User-to-user	Protocol discriminator and user information	User-to-User	uui-data (Note)
NOTE – The IWU shall always send uui-data as a token (see [IETF RFC 7433]). The letters used for the hex digits shall always be capital form.			

B.19.4 User-to-User Signalling (UUS) service 1 (explicit)

The actions of the IWU at the ISUP/BICC side are described in [ITU-T Q.737.1] under the clause "Interaction with other networks".

B.19.5 User-to-User Signalling (UUS) service 2

The actions of the IWU at the ISUP/BICC side are described in [ITU-T Q.737.1] under the clause "Interaction with other networks".

B.19.6 User-to-User Signalling (UUS) service 3

The actions of the IWU at the ISUP/BICC side are described in [ITU-T Q.737.1] under the clause "Interaction with other networks".

B.19.7 Profile C (SIP-I)

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.

B.20 Supplementary timers (For profiles A and B)

Table B.20-1 – TIR timer definition

Symbol	Timeout value	Cause for initiation	Normal termination	At expiry
TTIR1	100 – 2000 milliseconds (default 100 milliseconds)	On receipt of provisional or final response including the option tag "from-change"	At the receipt of an UPDATE	Map the received 200OK to an ANM

Annex C

(This annex forms an integral part of this Recommendation.)

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

C.1 SIP/SIP-I references (normative)

C.1.1 SIP/SIP-I signalling references and profile

C.1.1.1 References

See also clause C.2.

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

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

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

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

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

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

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

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

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

[IETF RFC 3323] IETF RFC 3323 (2002), *A Privacy Mechanism for the Session Initiation Protocol (SIP).*

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

[IETF RFC 3515] IETF RFC 3515 (April 2003), *The Session Initiation Protocol (SIP) Refer Method*.

[IETF RFC 3840] IETF RFC 3840 (August 2004), *Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)*.

[IETF RFC 3966] IETF RFC 3966, *The tel URI for Telephone Numbers*.

[IETF RFC 4040] IETF RFC 4040, *RTP Payload Format for a 64 kbit/s Transparent Call*"

[IETF RFC 4458] IETF RFC 4458 (April 2006), *Session Initiation Protocol (SIP) URIs for Applications such as Voicemail and Interactive Voice Response (IVR)*.

[IETF RFC 4575] IETF RFC 4575 (August 2006), *A Session Initiation Protocol (SIP) Event Package for Conference State*.

[IETF RFC 4694] IETF RFC 4694, *Number portability parameters for the "tel" URI*.

[IETF RFC 4715] IETF RFC 4715 (November 2006), *The Integrated Services Digital Network (ISDN) Subaddress Encoding Type for tel URI*.

[IETF RFC 4967] IETF RFC 4867 (May 2007), *RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs*.

[IETF RFC 5009] IETF RFC 5009, *Private Header (P-Header) Extension to the Session Initiation Protocol (SIP) for Authorization of Early Media*.

[IETF RFC 5079] IETF RFC 5079 *(December 2007), Rejecting Anonymous Requests in the Session Initiation Protocol (SIP).*

[IETF RFC 6086] IETF RFC 6086 (January 2001), *Session Initiation Protocol (SIP) INFO Method and Package Framework*.

[IETF RFC 6665] IETF RFC 6665 (July 2012), *SIP-Specific Event Notification*.

[IETF RFC 6910] IETF RFC 6910 (April 2013), *Completion of Calls for the Session Initiation Protocol (SIP).*

[IETF RFC 7044] IETF RFC 7044 (February 2014), *An Extension to the Session Initiation Protocol (SIP) for Request History Information*.

[IETF RFC 7433] IETF RFC 7433 (January 2015), *A Mechanism for Transporting User-to-User Call Control Information in SIP*.

[IETF RFC 7434] IETF RFC 7434 (January 2015), *Interworking ISDN Call Control User Information with SIP*.

[IETF RFC 7462] IET RFC 7462 (March 2015), *URNs for the Alert-Info Header Field of the Session Initiation Protocol (SIP).*

[IETF RFC 7090] IETF RFC 7090 (2014), *Public Safety Answering Point (PSAP) Callback*.

C.1.1.2 SIP/SIP-I signalling profiles

Reference	Profile	Profile C
RFC 2046 *Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types*	Supported	Supported
RFC 2327 *SDP: Session Description Protocol*	Supported	Supported
RFC 2806 *URLs for Telephone Calls*	Supported	Supported
RFC 2976 *The SIP INFO Method*	Not Supported	Supported
RFC 3204 *MIME media types for ISUP and QSIG Objects*	Not Supported	Supported
RFC 3261 *SIP: Session Initiation Protocol*	Supported	Supported
RFC 3262 *Reliability of Provisional Responses in the Session Initiation Protocol (SIP)*	Optional	Optional
RFC 3264 *An Offer/Answer Model with the Session Description Protocol (SDP)*	Supported	Supported
RFC 3311 *The Session Initiation Protocol UPDATE Method*	Supported	Supported
RFC 3312 *Integration of Resource Management and Session Initiation Protocol (SIP)*	Optional	Optional
RFC 3323 *A Privacy Mechanism for the Session Initiation Protocol (SIP)*	Supported	Supported
RFC 3325 *Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks*	Supported	Supported
RFC 3326 *The Reason Header Field for the Session Initiation Protocol (SIP)*	Supported	Supported

C.1.2 SIP/SIP-I media references

C.1.2.1 References

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

[IETF RFC 3267] 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*.

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

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

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

[IETF RFC 3556] IETF RFC 3556 (July 2003), *Session Description Protocol (SDP) Bandwidth Modifiers for RTP Control Protocol (RTCP) Bandwidth.*

C.2 The P-Asserted-Identity SIP header extension (normative)

This clause incorporates by reference the contents of [IETF RFC 3325]. This RFC was made Informational rather than Standards Track because IETF policy is to standardize open rather than closed networks. Its domain of applicability is defined in the opening section of the document. Interworking units covered by this Recommendation shall support the P-Asserted-Identity header field as defined in this annex, and shall additionally conform to the trust conditions applicable to the SIP network within which this header field is used.

Annex D

IWU – MGW Interaction

(This annex forms an integral part of this Recommendation.)

D.1 Overview

The IWU shall control the functions of the IM-MGW, which are used to provide the connection between media streams of an IP-based transport network and bearer channels from a circuit switched (CS) network.

The IWU shall interact with the IM-MGW across the Mn reference point. The IWU shall terminate the signalling across the Mn interface towards the IM-MGW and the IM-MGW shall terminate the signalling from the IWU.

The signalling interface across the Mn reference point shall be defined in accordance with [ITU-T H.248.1].

This Recommendation describes Mn signalling procedures and their interaction with BICC/ISUP and SIP signalling in the control plane, and with user plane procedures.

D.2 Mn signalling interactions

The following clauses describe the Mn interface procedures triggered by SIP and BICC signalling relayed in the IWU.

The SIP signalling occurring at the IWU is described in [ITU-T Q.3403].

All message sequence charts in this clause are examples.

D.2.1 Network model

Figure D.2.1-1 shows the network model, applicable to BICC and ISUP cases. The broken (green) line represents the call control signalling. The dotted (red) line represents the bearer control signalling (if applicable) and the user plane. The IWU uses one context with two terminations in the MGW. The termination T1 is used towards the core network entity and the bearer termination T2 is used for the bearer towards the succeeding CS network element.

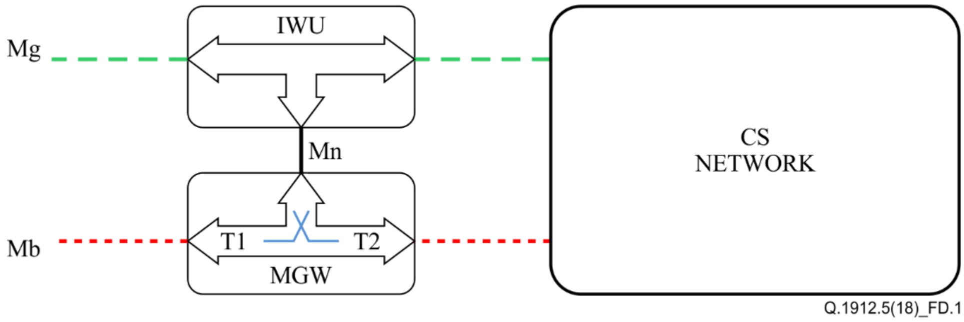


Figure D.2.1-1 – Network model

D.2.2 Reference points

Reference point IWU – core network (Mg reference point)

The Mg reference point allows the IWU to forward incoming session signalling (from the circuit switched network) to the core network for the purpose of interworking with circuit switched networks. The protocol used for the Mg reference point is SIP (as defined by [IETF RFC 3261], other relevant RFCs, and additional enhancements introduced to support this specification needs).

Reference point to IPv6 network services (Mb reference point)

Via the Mb reference point IPv6 network services are accessed. These IPv6 network services are used for user data transport. Note, that GPRS provides IPv6 network services to the UE, i.e., the GPRS Gi reference point and the IMS Mb reference point may be the same.

Reference point MGCF – IMS-MGW (Mn reference point)

The Mn reference point describes the interfaces between the MGCF and IMS-MGW in the IMS. It has the following properties:

– full compliance with the ITU-T H.248 standard functions for IMS – PSTN interworking;
– flexible connection handling which allows support of different call models and different media processing purposes not restricted to ITU-T H.323 usage;
– open architecture where extensions/Packages definition work on the interface may be carried out;
– dynamic sharing of IMS-MGW physical node resources. A physical IMS-MGW can be partitioned into logically separate virtual MGWs/domains consisting of a set of statically allocated terminations;
– dynamic sharing of transmission resources between the domains as the IMS-MGW controls bearers and manage resources according to the ITU-T H.248 protocols and functions for IMS.

D.2.3 Description of the IWU – MGW procedures

The procedures and interactions to control the MGW are described in clause 9 of [ITU-T Q.3629].

Annex E

Codec negotiation between a BICC CS network and the IM CN subsystem

(This annex forms an integral part of this Recommendation.)

E.1 Description of codec negotiation between a BICC circuit switched (CS) network and the IP multimedia (IM) core network (CN) subsystem

The optional procedures and interaction procedures for interworking of codec negotiation between a BICC CS network and the IM CN subsystem are described in Annex B of [ITU-T Q.3629].

Annex F

Multimedia interworking between the IP multimedia core network (CN) subsystem (IMS) and circuit switched (CS) networks

(This annex forms an integral part of this Recommendation.)

F.1 Description of multimedia interworking between the IP multimedia core network (CN) subsystem (IMS) and circuit switched (CS) networks

The normative procedures and interactions for the multimedia interworking between the IP multimedia core network (CN) subsystem (IMS) and circuit switched (CS) networks are described in Annex E of [ITU-T Q.3629].

Annex G

ITU-T T.38 interworking between the IP multimedia core network (CN) subsystem (IMS) and circuit switched (CS) networks

(This annex forms an integral part of this Recommendation.)

G.1 Description of the ITU-T T.38 interworking between the IP multimedia core network (CN) subsystem (IMS) and circuit switched (CS) networks

The normative procedures and interactions for the ITU-T T.38 interworking between the IP multimedia core network (CN) subsystem (IMS) and circuit switched (CS) networks are described in Annex K of [ITU-T Q.3629].

Annex H

PSTN XML Schema

(This annex forms an integral part of this Recommendation.)

H.1 Scope

This annex defines the PSTN XML schema to be used for providing the BearerCapability, Low Layer Compatibility, High Layer Compatibility and Progress Indicator embedded as body in SIP messages.

The support of this PSTN XML schema is a network option.

H.2 MIME type

The XML schema defined in the present annex is registered at IANA as "application/vnd.etsi.pstn+xml" MIME type.

If the XML scheme is embedded in SIP messages as body, the Content-Type header shall be set to "application/vnd.etsi.pstn+xml" and the Content-Disposition shall be set to "signal" with the "handling" parameter set to "optional".

H.3 XML Schema definition

The PSTN XML schema could be found in [ITU-T Q.3629] (see clause F.3 "XML Schema definition" of the endorsed ETSI TS 129 163).

H.4 IANA registration template

Within the present subclause, information required for an IANA registration at http://www.iana.org/cgi-bin/mediatypes.pl is provided.

1. Media Type Name

Application

2. Subtype name

"vnd.etsi.pstn+xml" (Vendor Tree)

3. Required parameters

none

4. Optional parameters

none

5. Encoding considerations

binary

6. Security considerations

The security considerations for XML in [IETF RFC 3023] apply.

Modifications of the MIME body by a man-in-the-middle can have severe consequences:

The information derived from ISUP that can be transported with this MIME body influence the call handling of the SIP session that is being set up.

However, this MIME body is used only attached to SIP messages, and modifications of other parts of the SIP signalling will lead to comparable consequences. Protection of the SIP signalling will also protect the present MIME body. The SIP signalling should be protected as described in [IETF RFC 3261].

7. Interoperability considerations

None

8. Published specification

[ITU-T Q.3629]

9. Applications which use this media type

This MIME type is used as a message body within SIP messages.

It is used for sending information derived from ISUP or DSS1 signalling with no SIP equivalent.

10. Additional information

Magic number(s): none (see [IETF RFC 3023])

File extension(s): ".xml" (see [IETF RFC 3023])

Macintosh File Type Code(s): "Text" (see [IETF RFC 3023])

Object Identifier(s) or OID(s): none

11. Intended usage

Limited Use

12. Other Information/General Comment

The content of this MIME type is UTF-8 encoded. The charset parameter is not used.

Appendix I

Interworking scenarios between SIP and BICC

(This appendix does not form an integral part of this Recommendation.)

I.1 Scope

This appendix defines typical interworking scenarios between SIP and BICC. ISDN Access flows are included for informational purposes only. The main body of the Recommendation takes precedence over this appendix.

I.2 Definitions

The vertical boxes represent two entities: a BICC SN and the IWU (SIP-BICC 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 signalling 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 action taken.

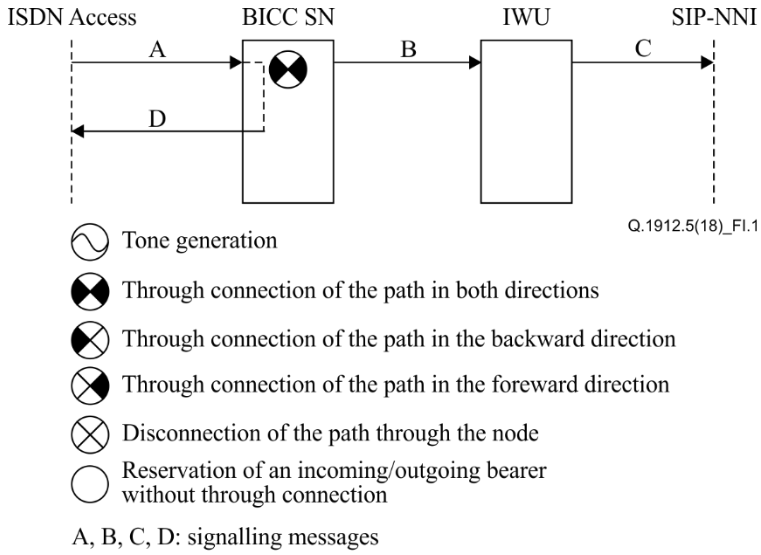


Figure I.1 – Example of a call flow or "arrow" diagram

I.3 Abbreviations

See clause 4.

I.4 Methodology

Call flow or "arrow' diagrams are provided to show the temporal relationships between signalling messages during execution of a call control procedure. The general format of an arrow diagram is shown in Figure I.1.

I.5 Interworking of SIP accesses to BICC

Clauses I.5.1 and I.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;

– release procedures;

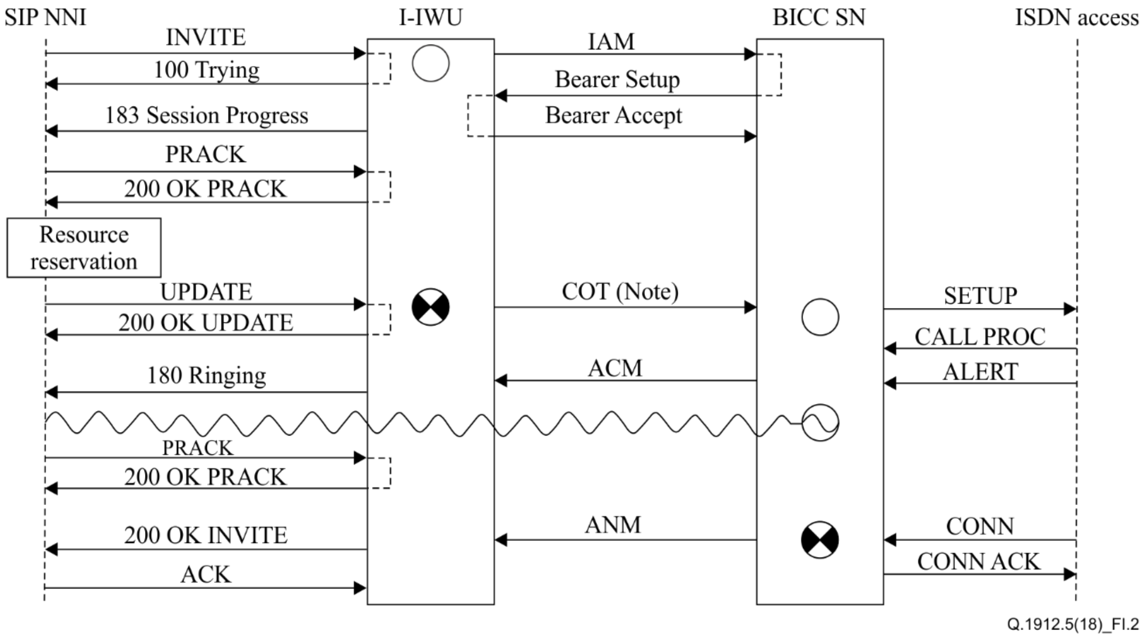
– simple message segmentation procedures.

I.5.1 Example scenarios for incoming call interworking from SIP to BICC at I-IWU

I.5.1.1 Successful call set-up procedures/call flow diagrams for basic call control

I.5.1.1.1 SIP preconditions used, backwards BICC bearer set-up, non-automatic answer

Figure I.2 shows the sequence of messages for successful call set-up for an incoming call from SIP to BICC. In this sequence, the SIP side indicates mandatory local resource reservation (such as sendrecv) in the INVITE. The IAM (with *"COT to be expected"* indication) is sent by the I-IWU once the initial INVITE is received, and a COT message is sent once the SIP side has reserved resources for the call (confirmed in the UPDATE). It is assumed that the ASN will be responsible for protecting against fraudulent use of the user plane.

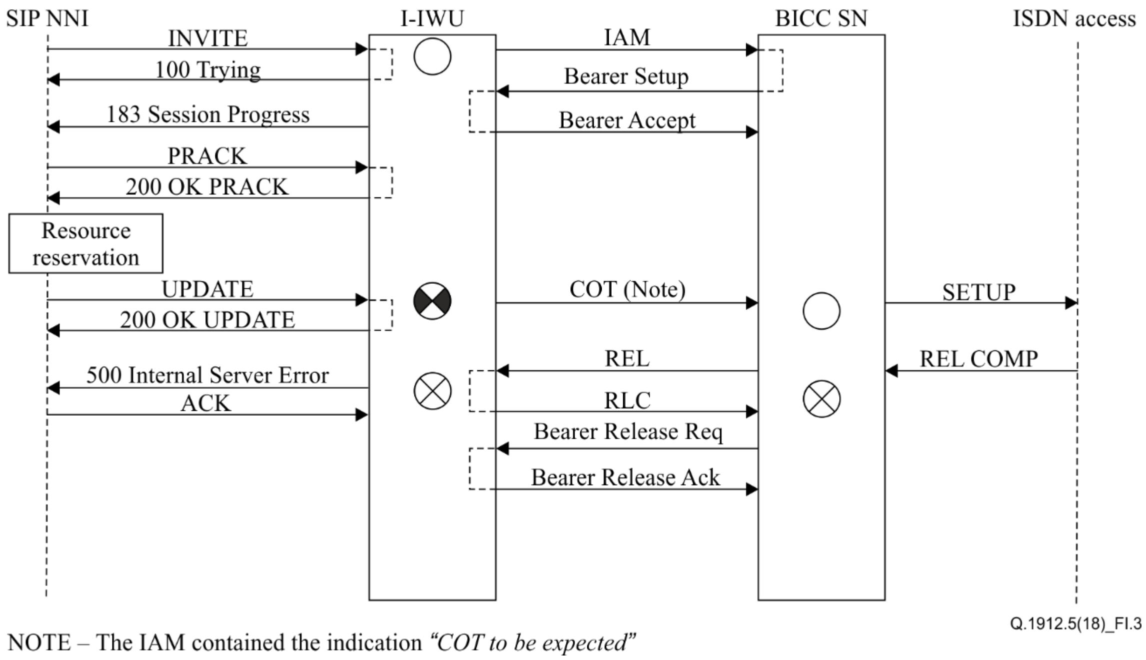


NOTE – The IAM contained the indication "*COT to be expected*".

Figure I.2 – Successful basic call set-up from SIP to BICC

I.5.1.2 Unsuccessful call set-up procedures/call flow diagrams for basic call control

Figure I.3 shows the sequence of messages for unsuccessful call set-up for an incoming call from SIP to BICC. In this sequence, the I-IWU sends the 500 Server Internal Error message upon receipt of the REL message (with Cause Value No. 34 (resource unavailable)) from the BICC side of the call.



NOTE – The IAM contained the indication *"COT to be expected"*

Figure I.3 – Unsuccessful basic call set-up from SIP to BICC

I.5.1.3 Release procedures/call flow diagrams for basic call control

I.5.1.3.1 Normal call release procedure, backward bearer set-up

Figure I.4 shows a normal call release procedure initiated from the SIP side of the call. This call flow assumes that no resource reservation teardown signalling is required on the SIP side.

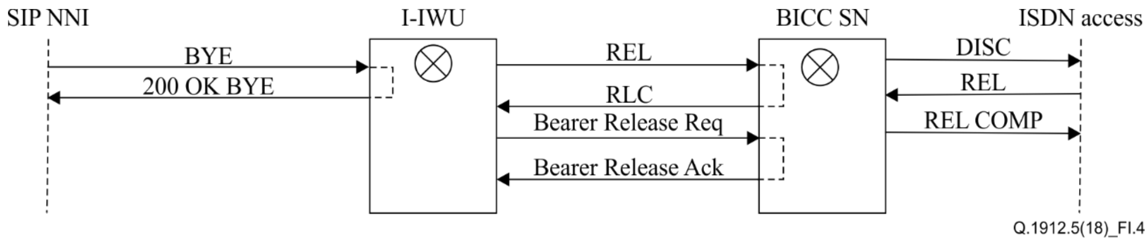
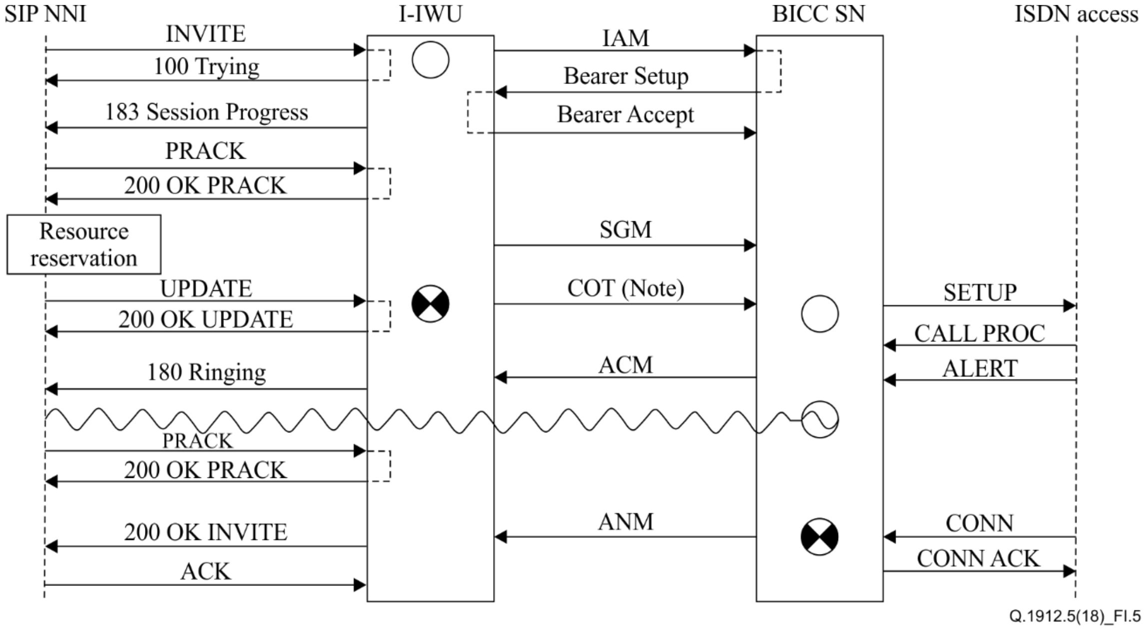


Figure I.4 – Normal call release from SIP to BICC

I.5.1.4 Simple segmentation procedures/call flow diagrams for basic call control

Figure I.5 shows a sequence of messages for successful call set-up for an incoming call from SIP to BICC using the segmentation procedures on the BICC side. In this example, the IWU sends the SGM independent of a message from the SIP side, and hence there is no interworking significance.



NOTE – The IAM contained the indication *"COT to be expected"*.

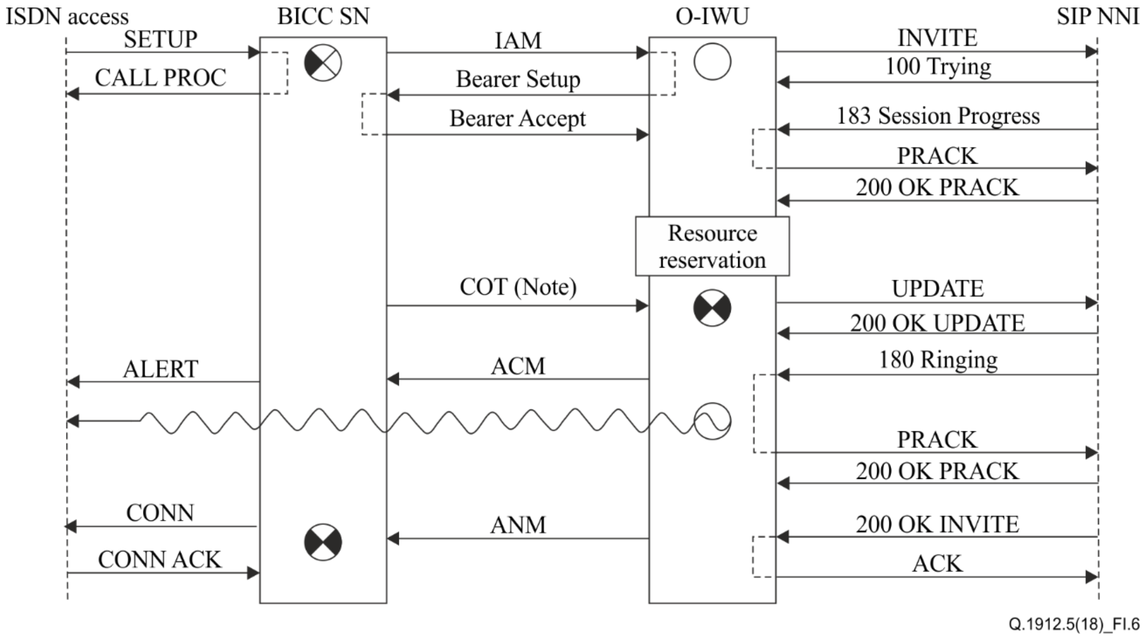
Figure I.5 – Basic call set-up using segmentation procedures from SIP to BICC

I.5.2 Example scenarios for outgoing call interworking from BICC to SIP at O-IWU

I.5.2.1 Successful call set-up procedures/call flow diagrams for basic call control

I.5.2.1.1 Backwards BICC bearer set-up, SIP preconditions used

Figure I.6 shows a sequence of messages for successful call set-up for an outgoing call from BICC to SIP. In this example, the O-IWU indicates mandatory local sendrecv preconditions in the INVITE. The O-IWU then sends the UPDATE message upon completion of bearer set-up, any local resource reservation and reception of a COT message (if the IAM indicated "*COT to be expected*"). The UPDATE message will confirm that local preconditions have been met. It is assumed that a SIP Proxy will be responsible for protecting against fraudulent use of the user plane.



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

Figure I.6 – Successful basic call set-up from BICC to SIP

I.5.2.2 Unsuccessful call set-up procedures/call flow diagrams for basic call control

Figure I.7 shows a sequence of messages for unsuccessful call set-up for an outgoing call from BICC to SIP. In this example, the O-IWU sends the REL message upon receipt of the 484 Address Incomplete message from the SIP side of the call.

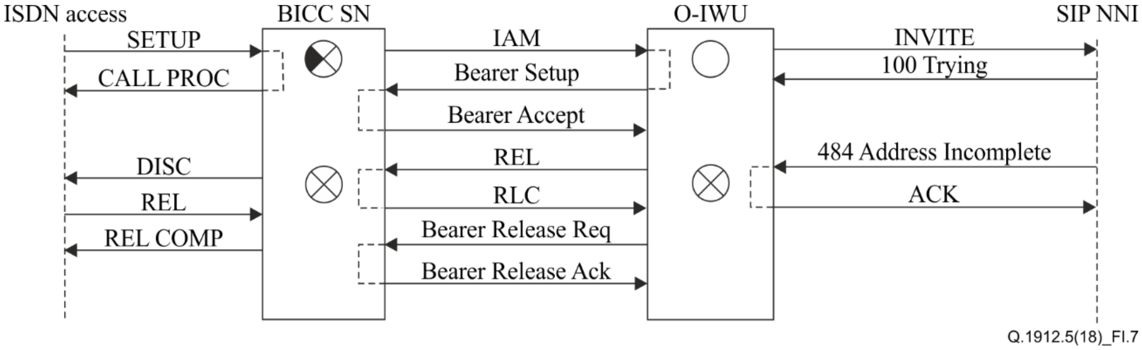


Figure I.7 – Unsuccessful basic call set-up from BICC to SIP

I.5.2.3 Release procedures/call flow diagrams for basic call control

I.5.2.3.1 Normal call release procedure, backwards bearer set-up

Figure I.8 shows a normal call release procedure initiated from the BICC side of the call. This call flow assumes that no resource reservation teardown signalling is required on the SIP side of the call.

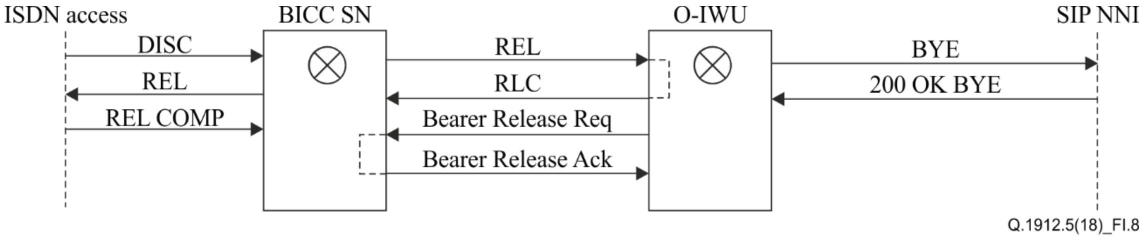


Figure I.8 – Normal call release from BICC to SIP

I.5.2.4 Simple segmentation procedures/call flow diagrams for basic call control

Figure I.9 shows a sequence of messages for successful call set-up for an outgoing call from BICC to SIP using the segmentation procedures. In this example, the O-IWU sends the INVITE message upon receipt of the SGM from the BICC side of the call.

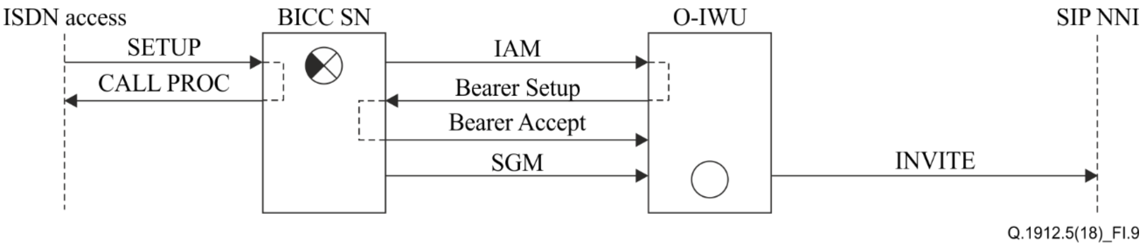


Figure I.9 – Basic call set-up using segmentation procedures from BICC to SIP

Appendix II

Interworking scenarios between SIP and ISUP

(This appendix does not form an integral part of this Recommendation.)

II.1 Scope

This appendix defines typical interworking scenarios between SIP and ISUP. ISDN Access flows are included for informational purposes only. The main body of the Recommendation takes precedence over this appendix.

II.2 Definitions

The vertical boxes represent two entities: an ISUP exchange and 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 signalling 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.

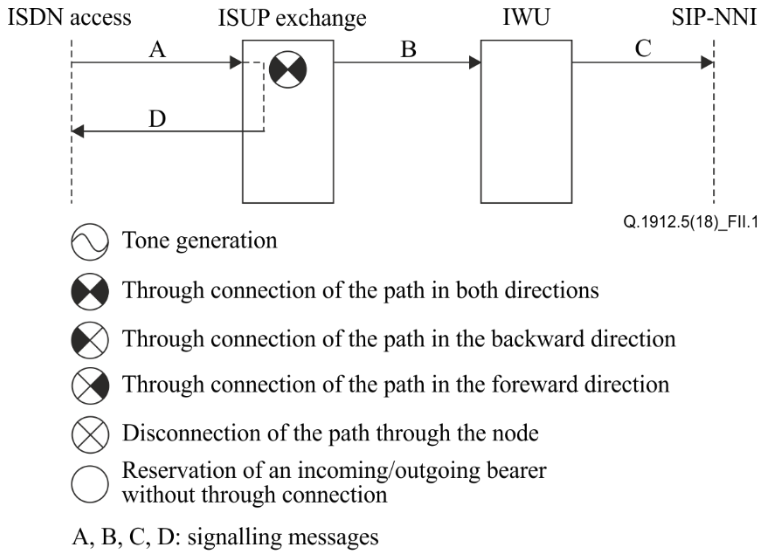


Figure II.1 – Example of a call flow or "arrow" diagram

II.3 Abbreviations

See clause 4.

II.4 Methodology

Call flow or "arrow" diagrams are provided to show the temporal relationships between signalling messages during execution of a call control procedure. The general format of an arrow diagram is shown in Figure II.1.

II.5 Interworking of SIP Access to ISUP

Clauses II.5.1 and II.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;

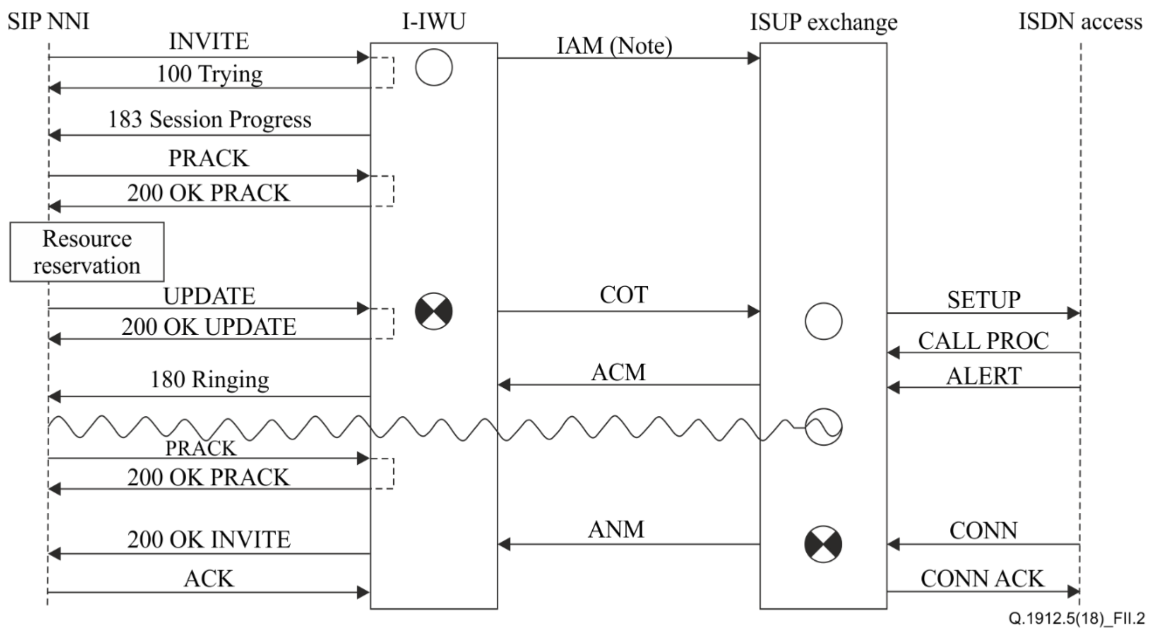
– release procedures.

II.5.1 Example scenarios for incoming call interworking from SIP to ISUP at I-IWU

II.5.1.1 Successful call set-up procedures and call flow diagrams for basic call control

II.5.1.1.1 SIP preconditions used

Figure II.2 shows the sequence of messages for successful call set-up for an incoming 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 I-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).

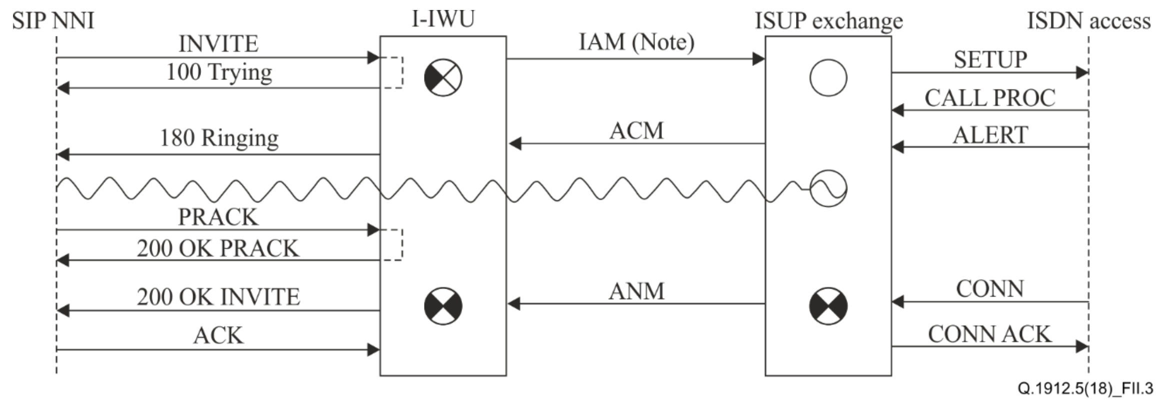


NOTE – The IAM contained the indication *"continuity check performed on previous circuit"* or *"continuity check required on this circuit"*.

Figure II.2 – Successful basic call set-up from SIP to ISUP (SIP preconditions and continuity check protocol used)

II.5.1.1.2 SIP preconditions not used

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

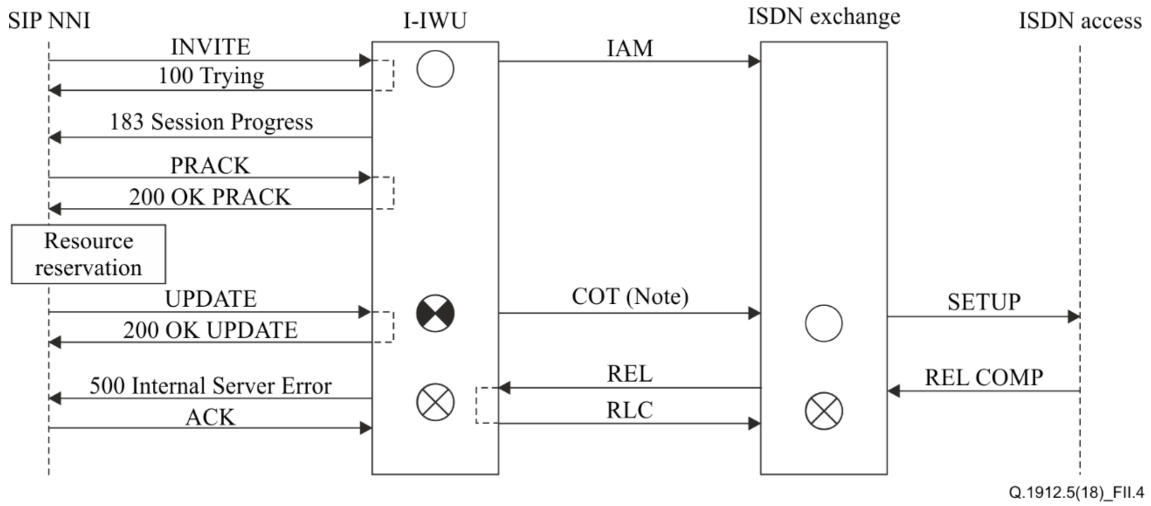


NOTE – The IAM contained the indication *"continuity check not required"*.

Figure II.3 – Successful basic call set-up from SIP to ISUP (SIP preconditions and continuity check protocol not used)

II.5.1.2 Unsuccessful call set-up procedures and call flow diagrams for basic call control

Figure II.4 shows the sequence of messages for unsuccessful call set-up for an incoming call from SIP to ISUP. In this sequence, the I-IWU sends the 500 Server Internal Error message upon receipt 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 II.4 – Unsuccessful basic call set-up from SIP to ISUP

II.5.1.3 Normal call release procedure

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

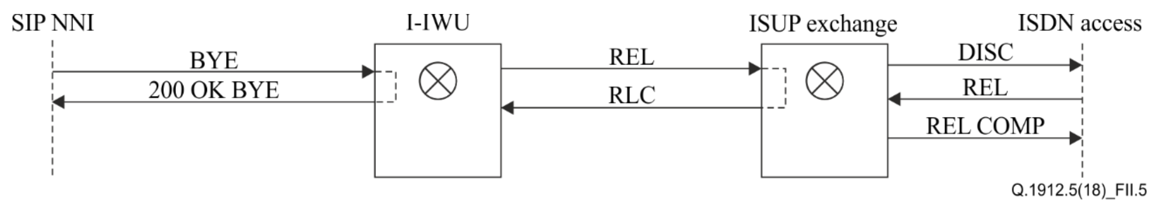


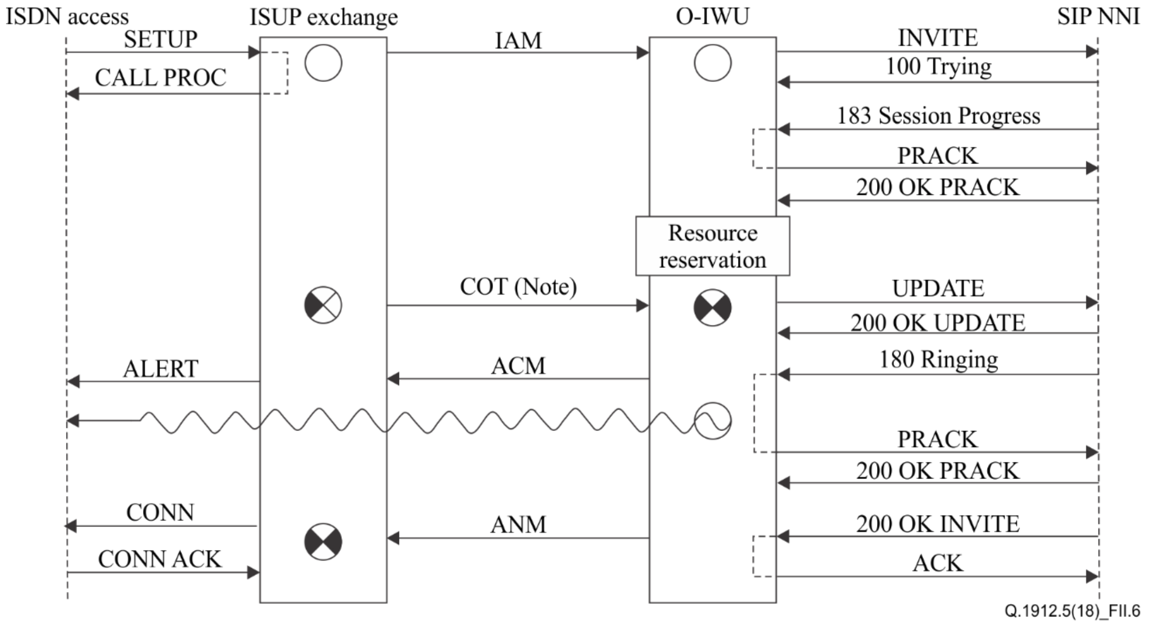
Figure II.5 – Normal call release from SIP to ISUP

II.5.2 Example scenarios for outgoing call interworking from ISUP to SIP at O-IWU

II.5.2.1 Successful call set-up procedures and call flow diagrams for basic call control

II.5.2.1.1 SIP preconditions used

Figure II.6 shows a sequence of messages for successful call set-up for an outgoing call from ISUP to SIP. In this example, the O-IWU indicates mandatory local sendrecv preconditions in the INVITE. The O-IWU then sends the UPDATE message upon receipt 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.

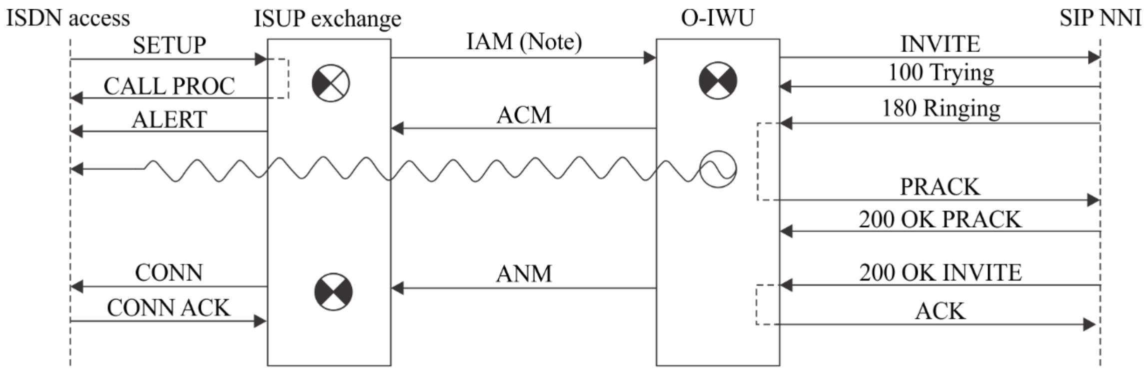


NOTE – This message is optional, depending on the indication in the IAM.

Figure II.6 – Successful basic call set-up from ISUP to SIP (SIP preconditions and continuity check protocol used)

II.5.2.1.2 SIP preconditions not used

Figure II.7 shows a sequence of messages for successful call set-up for an outgoing call from ISUP to SIP. In this example, the O-IWU sends the INVITE message upon receipt of an IAM (since the IAM indicated "*continuity check not required*").



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

Figure II.7 – Successful basic call set-up from ISUP to SIP (SIP preconditions and continuity check protocol not used)

II.5.2.2 Unsuccessful call set-up procedures and call flow diagrams for basic call control

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

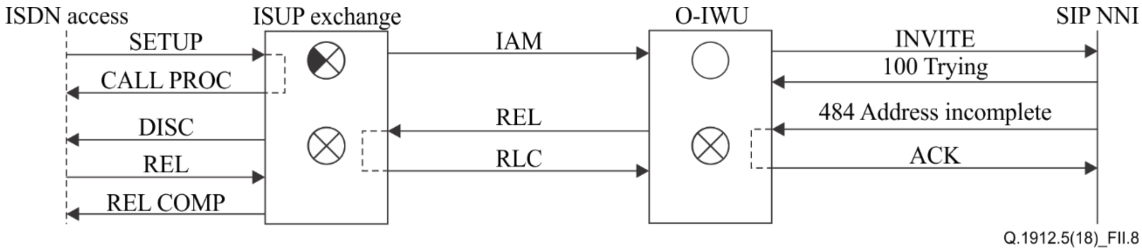


Figure II.8 – Unsuccessful basic call set-up from ISUP to SIP

II.5.2.3 Normal call release procedure

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

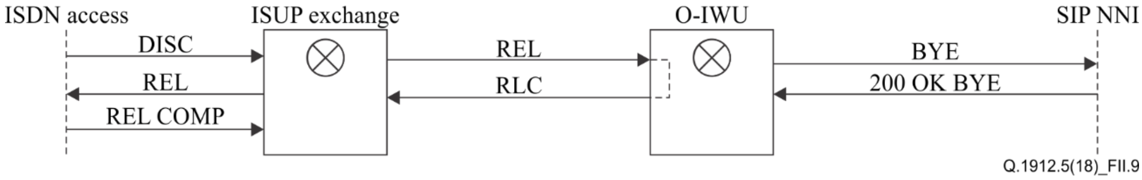


Figure II.9 – Normal call release from ISUP to SIP

Appendix III

Interworking scenarios between Profile C (SIP-I) and ISUP

(This appendix does not form an integral part of this Recommendation.)

III.1 General

III.1.1 Scope

This appendix defines some typical interworking scenarios between ISUP and SIP when Profile C (SIP-I) is in use. ISDN Access flows are included for informational purposes only. The operation of IWUs as a transit exchange is prearranged through configuration or analysis of received signalling information. The main body of the Recommendation takes precedence over this appendix.

III.1.2 Definitions

The vertical boxes represent originating and destination ISUP exchanges and outgoing and incoming IWUs (SIP-ISUP Interworking Units). Intermediate ISUP exchanges are not shown, since they do not change the basic call flows.

The vertical dashed lines represent the access interface, ISDN or non-ISDN depending on the example.

Solid horizontal arrows represent signalling 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 action taken.

III.1.3 Abbreviations

See clause 4.

III.1.4 Methodology

Call flow or "arrow" diagrams are provided to show the temporal relationships between signalling messages during execution of a call control procedure. The general format of an arrow diagram is shown in Figure III.1.

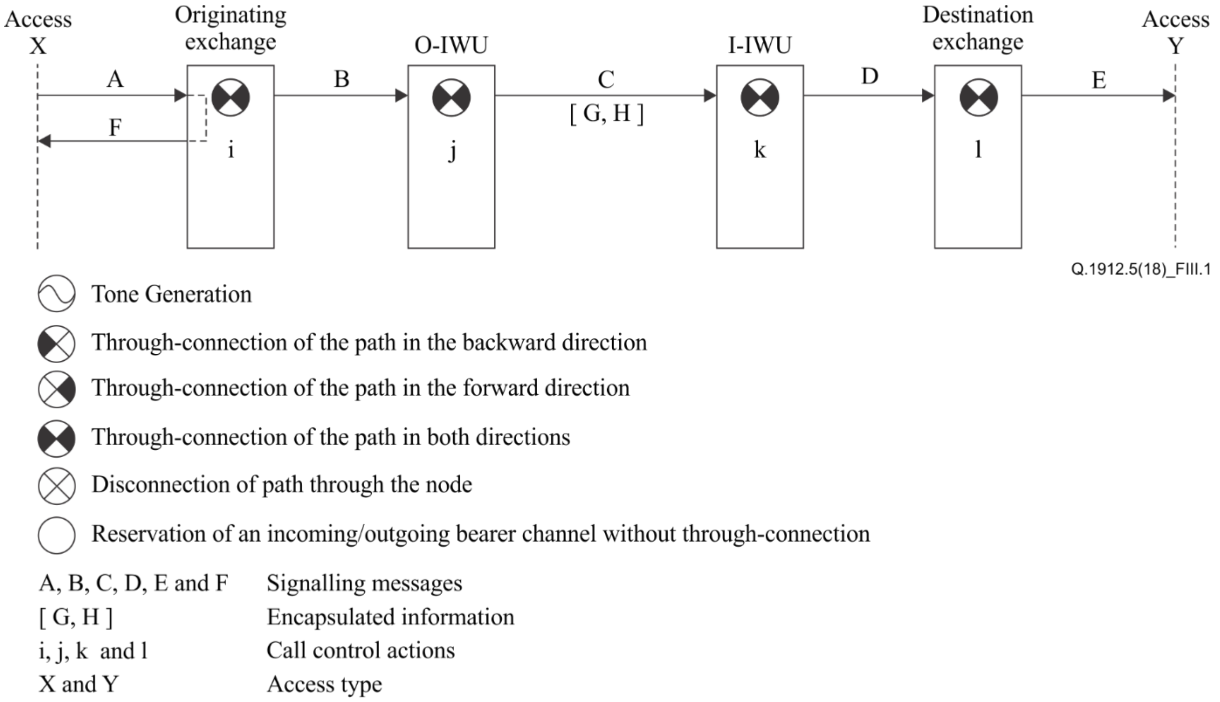


Figure III.1 – Example of a call flow or "arrow" diagram

III.2 Interworking of ISUP with SIP using Profile C (SIP-I)

Clauses III.2.1 to III.2.4 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;

– suspend/resume procedures.

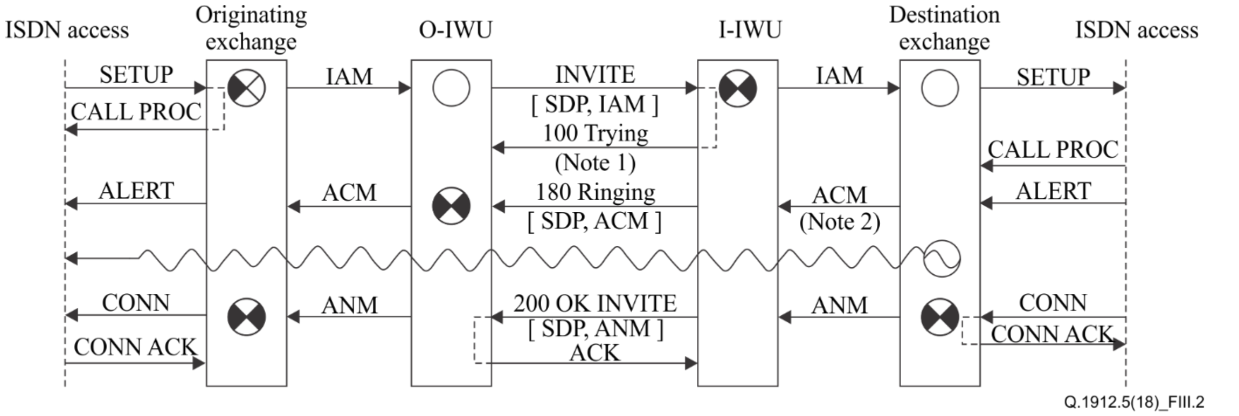
III.2.1 Successful call set-up procedures/call flow diagrams for basic call control

III.2.1.1 *En bloc*, subscriber free indication

See clause 2.1 of [ITU-T Q.764] and [IETF RFC 3261].

NOTE – Termed Late ACM.

Figure III.2 shows the sequence of messages for successful call set-up for an incoming ISUP call in the case of Profile C (SIP-I) operation. The O-IWU performs the through-connection of the bearer path in both directions after the receipt of SDP answer in the 180 Ringing response.



NOTE 1 – Any SIP entity along the signalling path to the I-IWU, or the I-IWU itself, may return a 100 Trying provisional response either by configuration or because it determines that a further response will take longer than 200 ms to generate. This is a purely SIP matter with no interworking significance, but is depicted for realism in this and subsequent figures.
NOTE 2 – ACM contained the following indicators:
Called Party Status = *"subscriber free"*, ISDN Access Indicator = *"ISDN access"*

Figure III.2 – *En bloc*, subscriber free indication

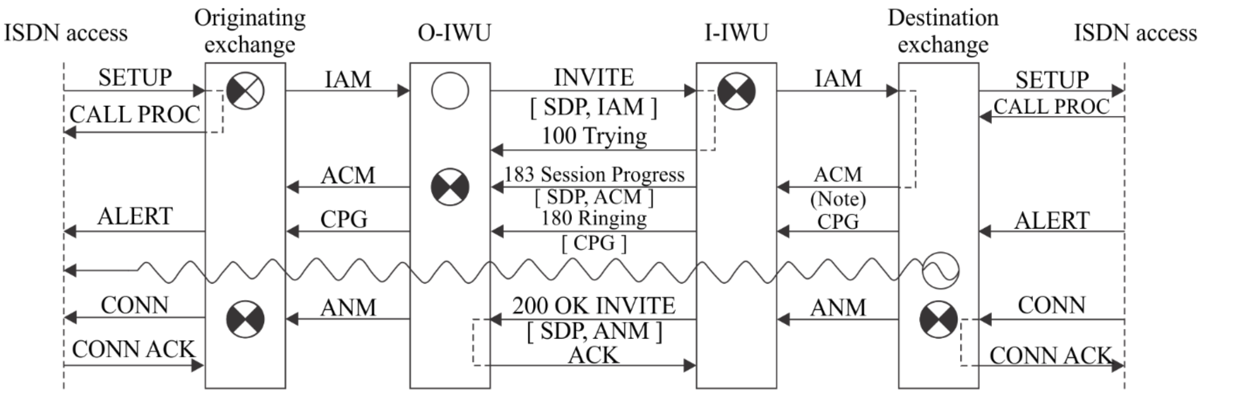
For detailed messages and parameter mapping, refer to:

– IAM – clauses 6.1.3 and 7.1.1 to 7.1.6.
– ACM – clauses 6.5 1) and 7.3.1.
– ANM – clauses 6.7 and 7.5.

III.2.1.2 *En bloc*, early ACM

See clause 2.1 of [ITU-T Q.764] and [IETF RFC 3261].

Figure III.3 shows the sequence of messages for successful call set-up for an incoming ISUP call in the case of Profile C (SIP-I) operation. At the I-IWU the ACM is mapped and encapsulated to 183 Session Progress provisional response preserving the ISUP signalling transparency. The O-IWU performs the through-connection of the bearer path in both directions after the receipt of SDP answer in the 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 indicators: Called Party Status = "*no indication*"; ISDN Access Indicator = "*ISDN access*"

Figure III.3 – *En bloc*, early ACM encapsulation

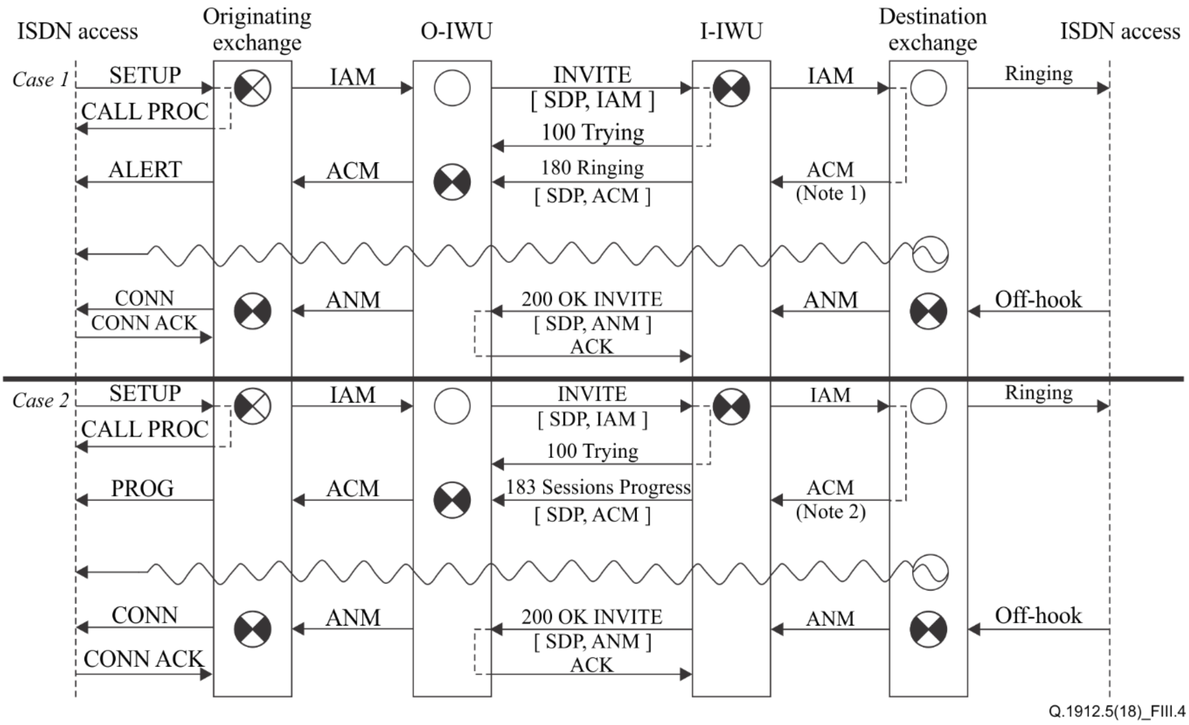
For detailed messages and parameter mapping, refer to:

– IAM – clauses 6.1.2 and 7.1.
– ACM – clauses 6.5 2) and 7.3.2.
– CPG message – clauses 6.6 and 7.3.1.
– ANM – clauses 6.7 and 7.5.

III.2.1.3 *En bloc*, early media scenarios

See clause 2.1 of [ITU-T Q.764] and [IETF RFC 3261].

Figure III.4 Cases 1 and 2 show sequences of messages for a call from an ISDN access to a non-ISDN access. The two cases differ based on the contents of the ACM generated at the destination exchange.



NOTE 1 – The ACM in case 1 is independently generated at the destination exchange with the following indicators: Called Party Status = *"subscriber free"*, ISDN Access Indicator = *"non-ISDN access"*.

NOTE 2 – The ACM in case 2 is independently generated at the destination exchange with the following indicators: Called Party Status = *"no indication"*, ISDN Access Indicator = *"non-ISDN access"*. In order to support user-generated in-band information (e.g., from a PBX, see 2.1.4.1b/ITU-T Q.764), the destination exchange may through-connect in the backward direction and include in the ACM the Optional Backward Call Indicators parameter indicating *"in-band information or an appropriate pattern is now available"*.

Figure III.4 – Early media call-flows

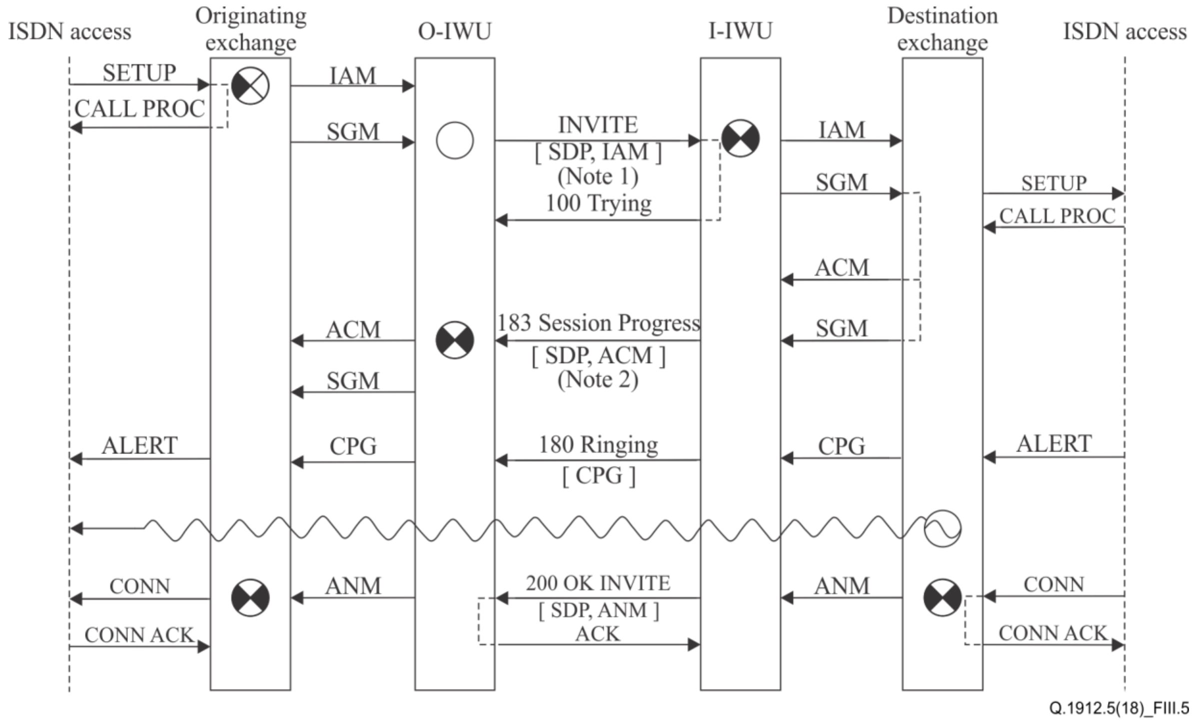
For detailed messages and parameter mapping, refer to:

– IAM – clauses 6.1.2 and 7.1.
– ACM – clauses 6.5 1)/6.5 2) and 7.3.1/7.3.2.
– CPG message – clauses 6.6 and 7.3.1.
– ANM – clauses 6.7 and 7.5.

III.2.1.4 *En bloc*, simple segmentation procedures

See clause 2.1.12 of [ITU-T Q.764] and [IETF RFC 3261].

Figure III.5 indicates the simple segmentation procedures in the forward and backward directions. Before the encapsulation, the IWU reassembles the incoming ISUP message with its segmented part (see clause 5.4.3.3). After de-encapsulation, the IWU applies ISUP segmentation procedures, if needed.



NOTE 1 – The complete re-assembled IAM message is encapsulated in the INVITE request.
NOTE 2 – The complete re-assembled ACM message is encapsulated in the 183 provisional response.

Figure III.5 – *En bloc*, simple segmentation in both directions

For detailed messages and parameter mapping, refer to:

– IAM – clauses 6.1.2 and 7.1.

– SGM – clause 5.4.3.3.

– ACM – clauses 6.5 2) and 7.3.2.

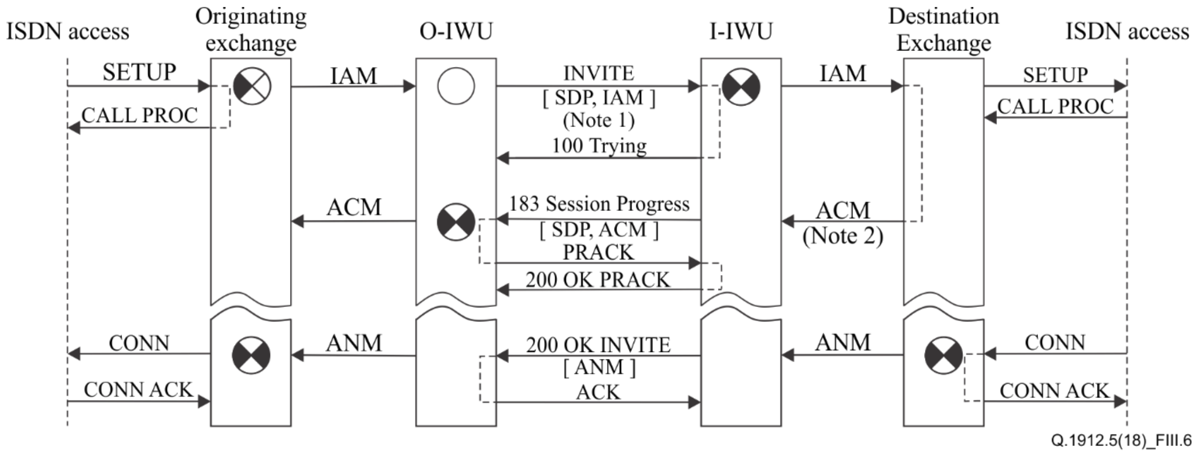
– CPG message – clauses 6.6 and 7.3.1.

– ANM – clauses 6.7 and 7.5.

III.2.1.5 *En bloc*, reliable provisional responses

See clauses 2.1 of [ITU-T Q.764] and 4 of [IETF RFC 3262].

Figure III.6 shows the sequence of messages for successful call set-up for an incoming ISUP call in the case of Profile C (SIP-I) operation. 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 is mapped and encapsulated in a 183 Session Progress response preserving the ISUP signalling transparency. The O-IWU confirms the receipt of provisional response with the PRACK request. Typically there will be an alerting phase, not shown here, with mapping of ISUP CPG message to 180 Ringing. 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 – ACM contained the following indicators:
Called Party Status = *"no indication"*, ISDN Access Indicator = *"ISDN access"*.

Figure III.6 – *En bloc*, use of reliable provisional responses

For detailed messages and parameter mapping, refer to:

– IAM – clauses 6.1.2 and 7.1.

– ACM – clauses 6.5 2) and 7.3.2.

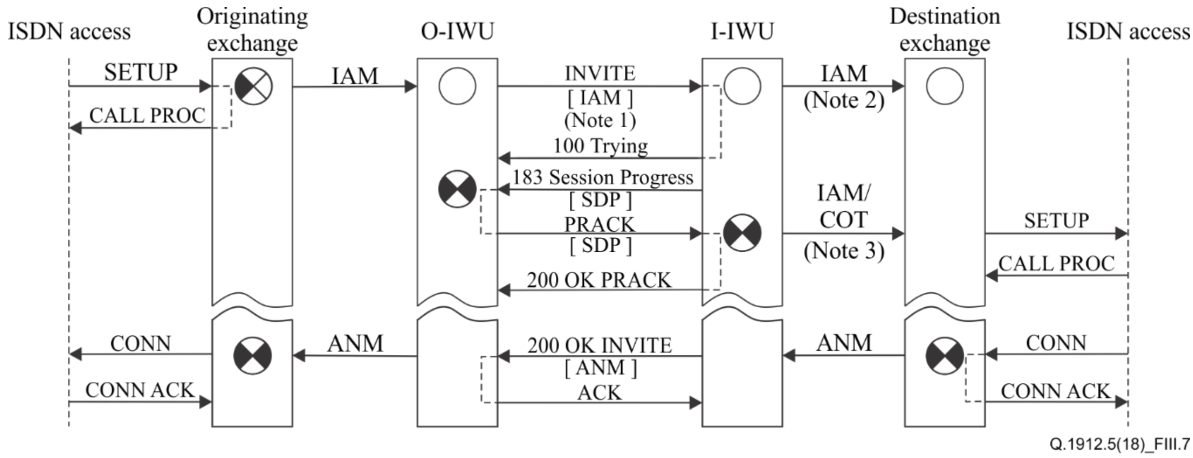
– ANM – clauses 6.7 and 7.5.

III.2.1.6 *En bloc*, backward SDP offer

See clause 2.1 of [ITU-T Q.764] and [IETF RFC 3261].

Figure III.7 shows the sequence of messages for successful call set-up for an incoming ISUP call in the case of Profile C (SIP-I) operation. 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. If the I-IWU supports the procedure, it can transfer an SDP offer via a 183 Session Progress response. The O-IWU responds with SDP answer and performs the through-connection of the bearer path in both directions after the receipt of SDP answer in the 183 Session Progress response.

Depending on configuration, I-IWU can directly send IAM with *"COT on previous circuit"* indication and continue the call set-up 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 clause 6.1.1 1). In any scenario, the I-IWU through-connects the bearer path on the receipt of SDP answer. The alerting phase is omitted from the figure for simplicity.



NOTE 1 – INVITE contains the Supported header field with the option tag 100rel.
NOTE 2 – In the case of immediate sending of IAM, it will contain *"COT on previous circuit"* indication.
NOTE 3 – The choice between deferred IAM and COT depends on the I-IWU configuration.

Figure III.7 – *En bloc*, backward session description initiation

For detailed messages and parameter mapping, refer to:

– IAM – clauses 6.1.1 1) and 7.1.

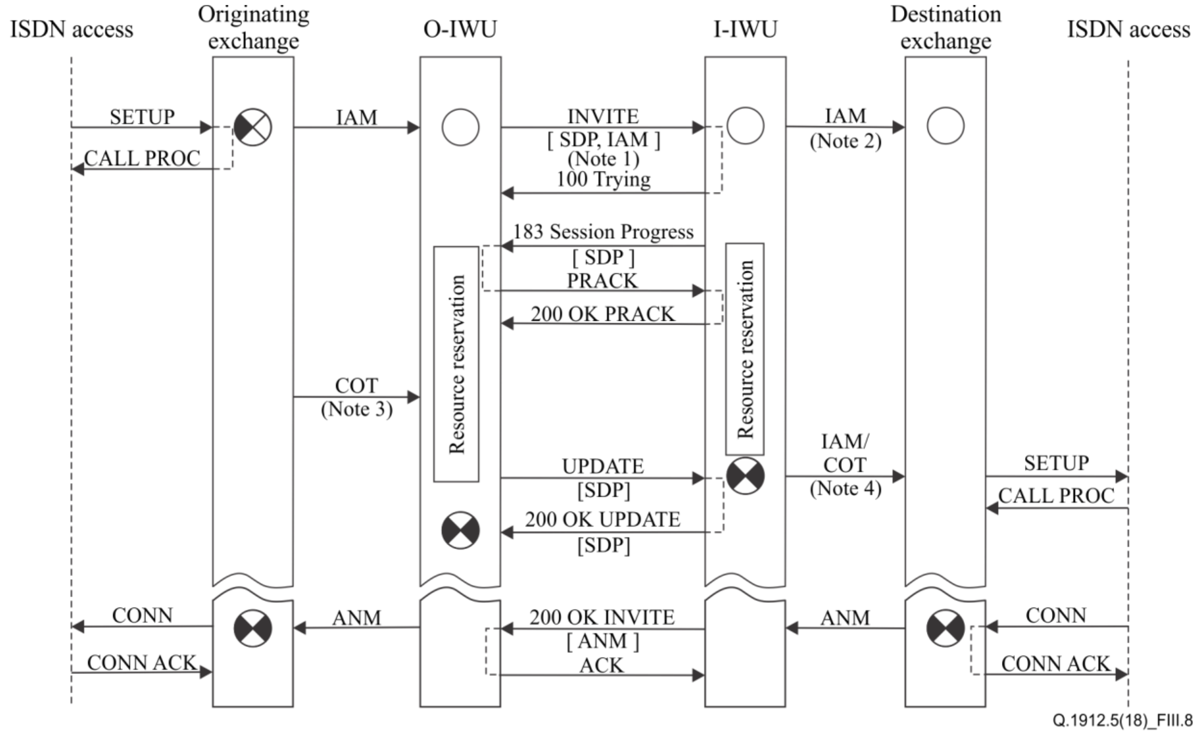
– ANM – clauses 6.7 and 7.5.

III.2.1.7 *En bloc*, end-to-end resource reservation

See clauses 2.1 of [ITU-T Q.764] and 13.1 of [IETF RFC 3312].

Figure III.8 shows the sequence of messages for successful call set-up for an incoming ISUP call in the case of Profile C (SIP-I) operation. 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-IWU requests confirmation from the O-IWU of end-to-end network resource reservation in the SDP of 183 Session Progress response and begins with its 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 indicates its status in the SDP of an 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 set-up by sending COT after meeting the preconditions. As an alternative, it can delay the sending of IAM until the meeting of preconditions. See clause 6.1.2 2).



NOTE 1 – INVITE contains mandatory end-to-end sendrecv preconditions and the Required header field with the option tag 100rel.
NOTE 2 – In the case of immediate sending of IAM, it will contain *"COT on previous circuit"* indication.
NOTE 3 – COT on the originating side is optional, depending on the indication in the IAM.
NOTE 4 – The choice between deferred IAM and COT depends on the I-IWU configuration, see 6.1.2.

Figure III.8 – *En bloc*, end-to-end preconditions for resource reservation

For detailed messages and parameter mapping, refer to:

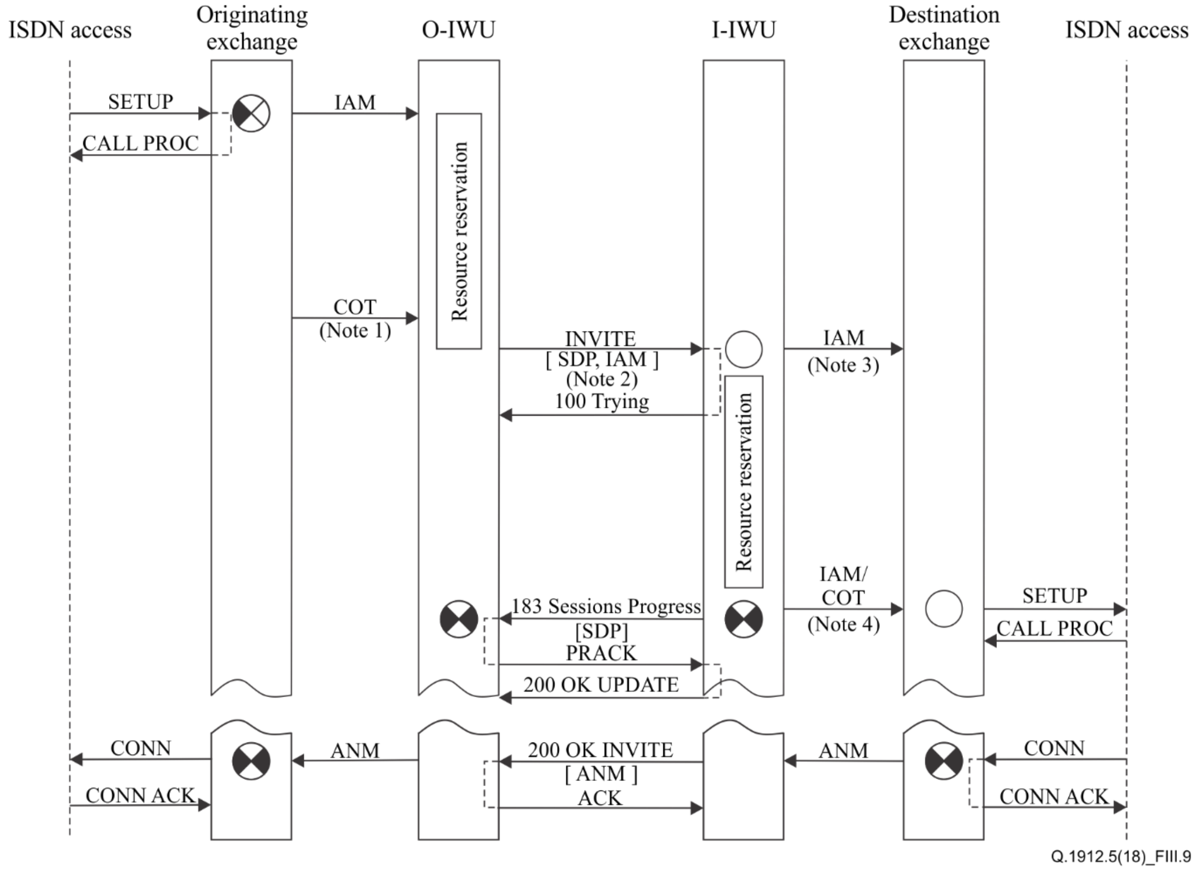
– IAM – clauses 6.1.2 (2) and 7.1 (B).
– COT message – clauses 6.3 and 7.1 (B).
– ANM – clauses 6.7 and 7.5.

III.2.1.8 *En bloc*, segmented resource reservation

See clause 2.1 of [ITU-T Q.764] and clause 13.2 of [IETF RFC 3312].

Figure III.9 shows the sequence of messages for successful call set-up for an incoming ISUP call in the case of Profile C (SIP-I) operation. 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*"), the O-IWU includes the request for the reservation of the local network resource at I-IWU and also the required use of reliable provisional responses in the SDP of the initial INVITE. After local network resource reservation the I-IWU notifies the O-IWU with SDP in 183 Session Progress 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 set-up by sending COT after meeting the preconditions. As an alternative, it can delay the sending of IAM until the meeting of preconditions.



NOTE 1 – COT on the originating side is optional, depending on the indication in the IAM.
NOTE 2 – INVITE contains mandatory segmented sendrecv preconditions and the Required header field with the option tag 100rel.
NOTE 3 – In the case of immediate sending of IAM, it will contain *"COT on previous circuit"* indication.
NOTE 4 – The choice between deferred IAM and COT depends on the I-IWU configuration, see 6.1.2.

Figure III.9 – *En bloc,* segmented preconditions for resource reservation

For detailed messages and parameter mapping, refer to:

– IAM – clauses 6.1.2 (2) and 7.1 (B).

– COT message – clauses 6.3 and 7.1 (B).

– ANM – clauses 6.7 and 7.5.

III.2.1.9 *En bloc,* automatic call answering

See clause 2.1 of [ITU-T Q.764] and [IETF RFC 3261].

Figure III.10 shows the sequence of messages for successful call set-up for an incoming ISUP call in the case of Profile C (SIP-I) operation. The I-IWU 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.

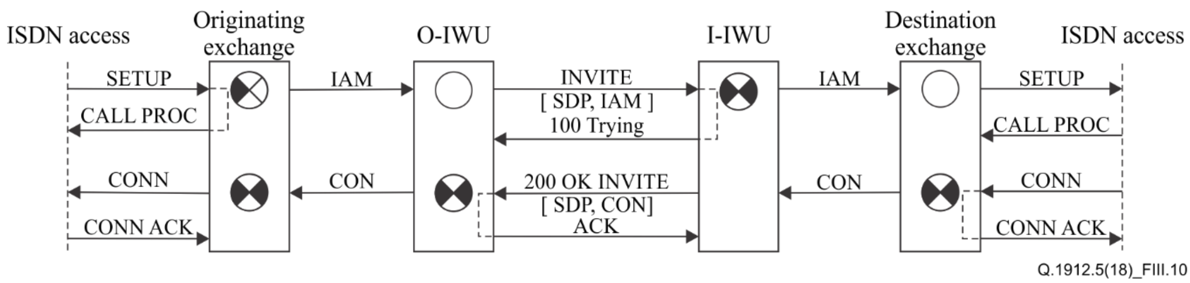


Figure III.10 – *En bloc,* automatic answering terminal

For detailed messages and parameter mapping, refer to:

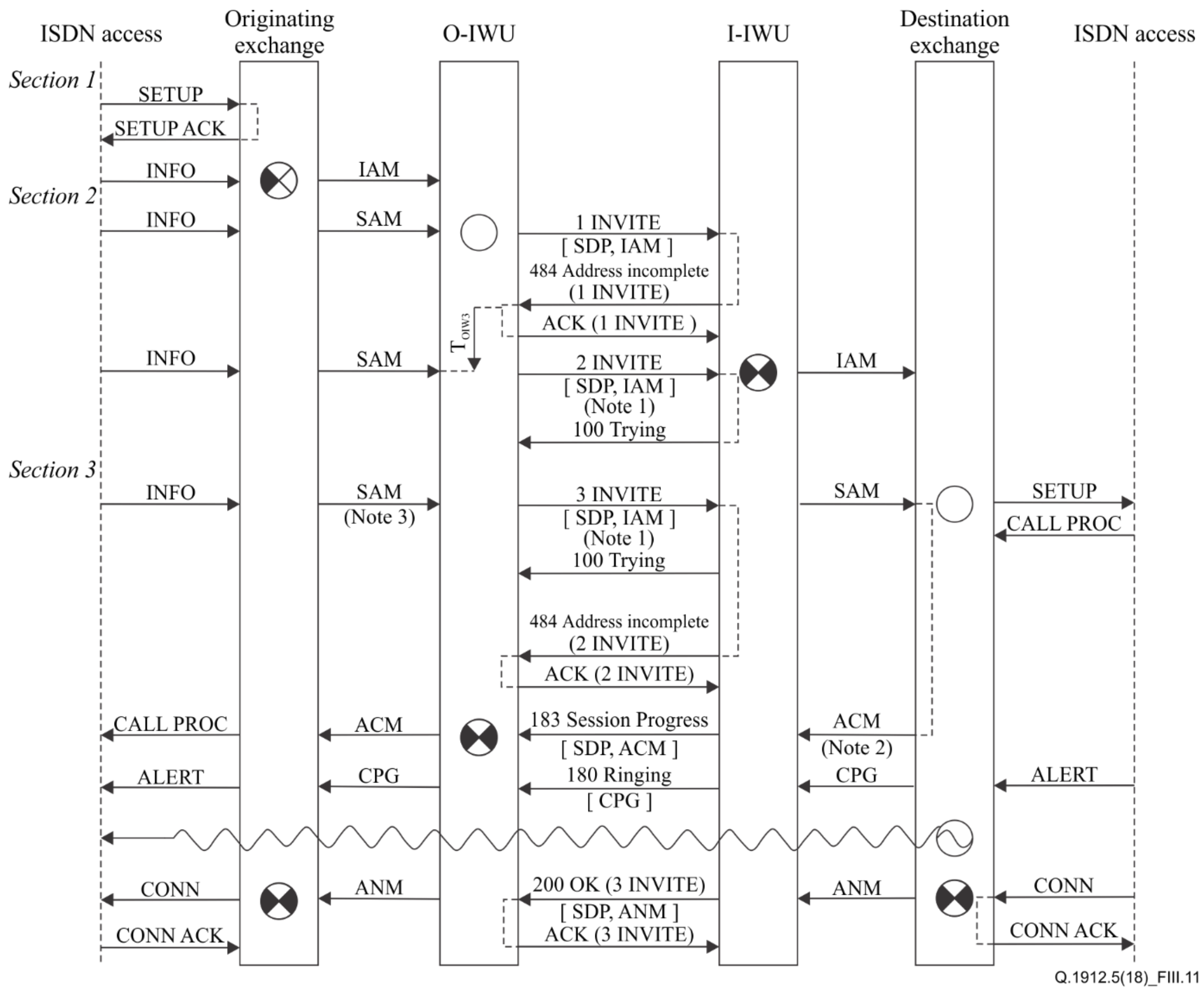
– IAM – clauses 6.1.1 1) and 7.1 A).

– CON message – clauses 6.4 and 7.5.

III.2.1.10 Overlap signalling

See clause 2.1 of [ITU-T Q.764] and [IETF RFC 3261].

Figure III.11 shows the sequence of messages when overlap sending is in use. The figure is divided into three sections where, in the first section, the O-IWU did not receive enough digits to progress the call. In the second section, O-IWU receives enough digits, but the I-IWU cannot progress the call and sends a 484 Address Incomplete final response. Since the O-IWU is configured to perform overlap sending, it does not release the call but starts the timer T_{OIW3}. Before timer T_{OIW3} expires a following SAM triggers the sending of subsequent INVITE 2 and clears the timer T_{OIW3}. In the third section, the next SAM triggers the sending of subsequent INVITE 3. On the reception of INVITE 3, the I-IWU sends the SAM to the destination exchange and terminates the INVITE 2 transaction with a 484 Address Incomplete final response. The O-IWU clears the transaction 2 INVITE, but does not start timer T_{OIW3} and does not release the call as the INVITE 3 transaction is still pending.



NOTE 1 – INVITE 2 and INVITE 3 have the same Call-ID and From tag as INVITE 1, but have Request-URIs updated to include all digits received to that point. For details see 7.2.
NOTE 2 – The ACM is independently generated at the destination exchange with the following indicators: Called Party Status = *"no indication"*, ISDN Access Indicator = *"ISDN access"*.
NOTE 3 – The number of SAMs shown is for illustration only. In practice there may be zero or more SAMs.

Figure III.11 – Overlap addressing

For detailed messages and parameter mapping, refer to:

– IAM – clauses 6.1.2 and 7.1.

– SAM – clauses 6.2.1 and 7.2.1.

– ACM – clauses 6.5 2) and 7.3.2.

– CPG message – clauses 6.6 and 7.3.1.

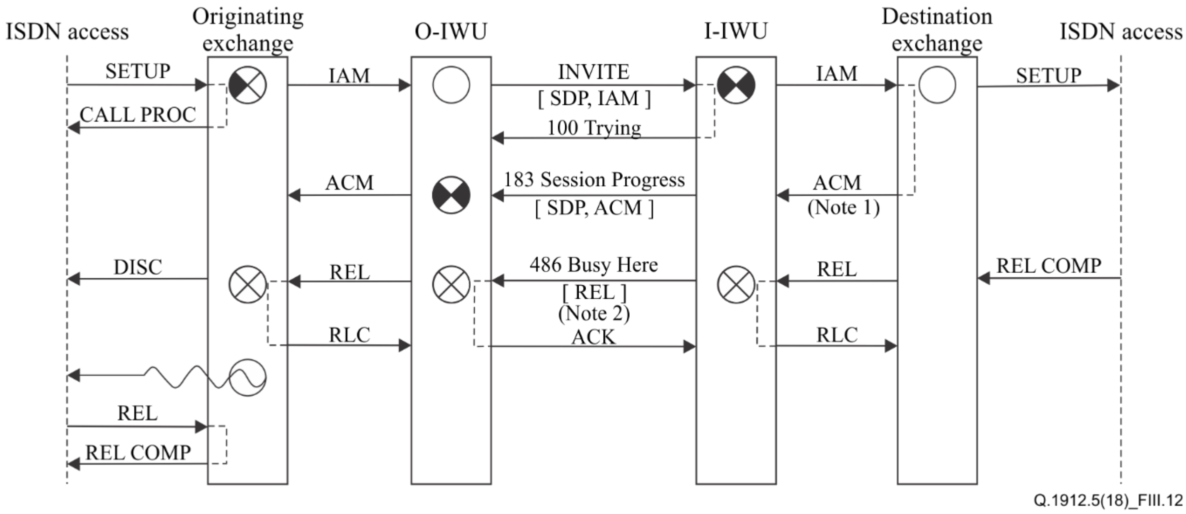
– ANM – clauses 6.7 and 7.5.

III.2.2 Unsuccessful call set-up procedures/call flow diagrams for basic call control

III.2.2.1 Backward release during call set-up

See clause 2.2 of [ITU-T Q.764] and [IETF RFC 3261].

Figure III.12 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 – If early ACM is used, the ACM is independently generated at the destination exchange with the following indicators: Called Party Status = *"no indication"*, ISDN Access Indicator = *"non-ISDN access"*.
NOTE 2 – See Tables 21 and 40 for mapping between release causes and SIP status codes.

Figure III.12 – Backward release during call set-up

For detailed messages and parameter mapping, refer to:

– IAM – clauses 6.1.2 and 7.1.

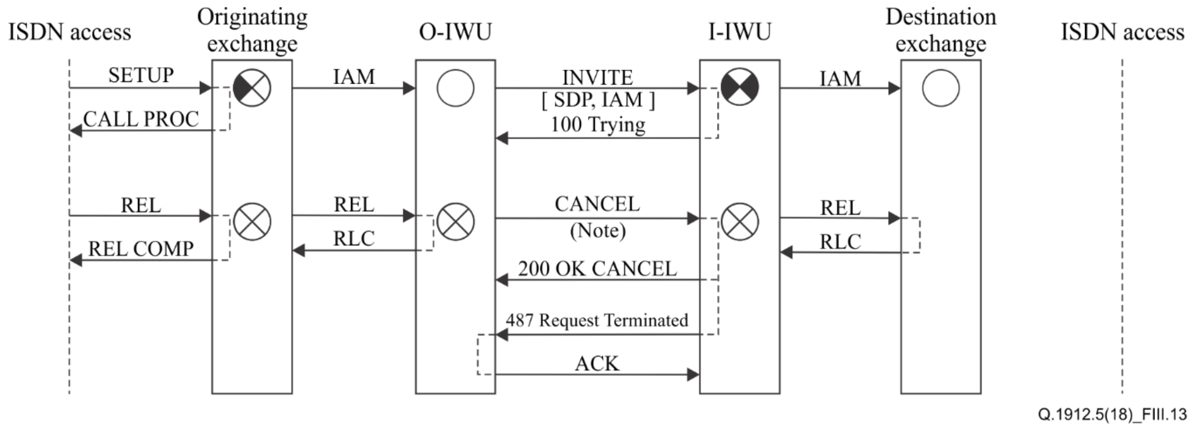
– ACM – clauses 6.5 2) and 7.3.2.

– REL message – clauses 6.11.2 (Table 6-50) and 7.7.6 (Table 39).

III.2.2.2 Forward release during call set-up, no early dialogue

See clause 2.2 of [ITU-T Q.764] and [IETF RFC 3261].

Figure III.13 shows a premature release situation where a Release message is received at the O-IWU prior to successful early dialogue set-up. In this situation, a CANCEL request is sent to the I-IWU and the normal release procedure is started.



NOTE – REL is not encapsulated in CANCEL because the latter is a hop-by-hop request. If the O-IWU supports the Reason header field the Cause Value is mapped to that field. See 6.11.1 and 7.7.1.

Figure III.13 – Forward release during call set-up, no early dialogue is established

For detailed messages and parameter mapping, refer to:

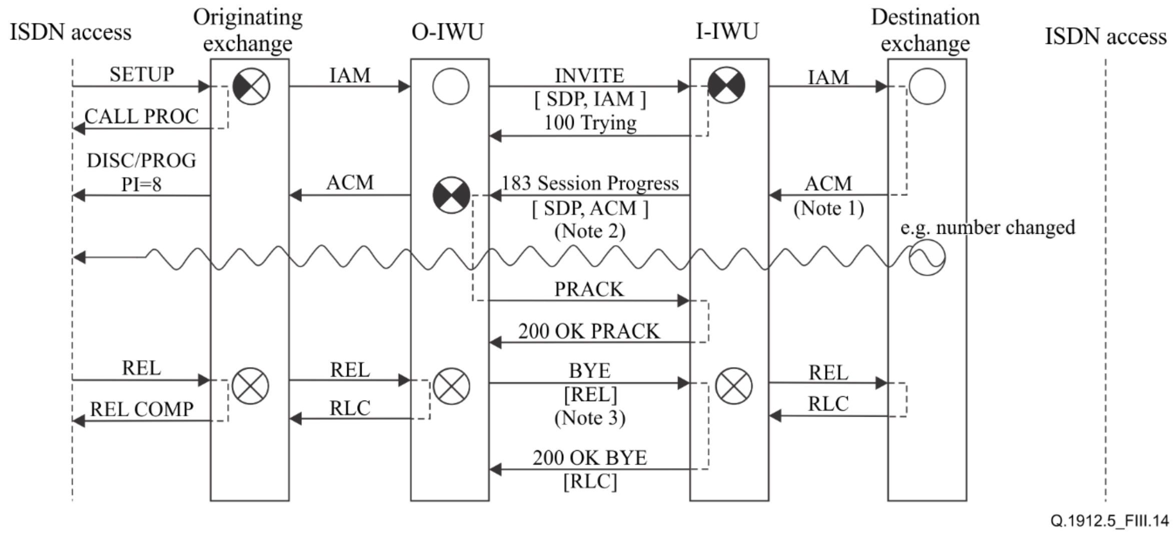
– IAM – clauses 6.1.2 and 7.1.

– REL message – clauses 6.11.1 and 7.7.1 1).

III.2.2.3 Forward release during call set-up, early dialogue is established

See clause 2.2 of [ITU-T Q.764] and [IETF RFC 3261].

Figure III.14 shows an unsuccessful call set-up where certain tones and announcements are generated in the destination exchange during call establishment. 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. The REL message is mapped and encapsulated in the BYE request as an early dialogue is already established through the reception of a To tag in the 183 Session Progress response.



NOTE 1 – The ACM is not mapped from a message from the destination user. It is independently generated at the destination exchange.
NOTE 2 – The 183 Session Progress response contains the To header field tag which creates an early dialogue.
NOTE 3 – Since an early dialogue has been established, the O-IWU can release the call with a BYE rather than a CANCEL. Since BYE is end-to-end, it can encapsulate the REL.

Figure III.14 – Forward release during call set-up, early dialogue is already established

For detailed messages and parameter mapping, refer to:

– IAM – clauses 6.1.2 and 7.1.

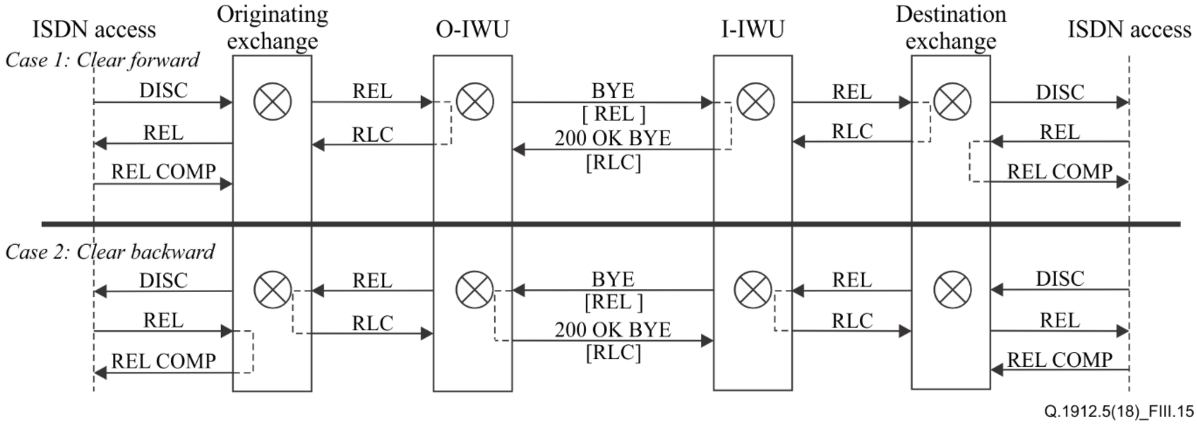
– ACM – clauses 6.5 2) and 7.3.2.
– REL message – clauses 6.11.1 and 7.7.1 2).

III.2.3 Release procedures/call flow diagrams for basic call control

III.2.3.1 Normal call release procedure without tone provision

See clause 2.3 of [ITU-T Q.764] and [IETF RFC 3261].

Figure III.15 shows the normal call release interworking procedures without tone provision. A REL message is mapped and encapsulated into BYE request to preserve the ISUP signalling transparency.



NOTE – This procedure is applicable where in-band tones or announcements are not provided, e.g., 64 kbit/s unrestricted bearer.

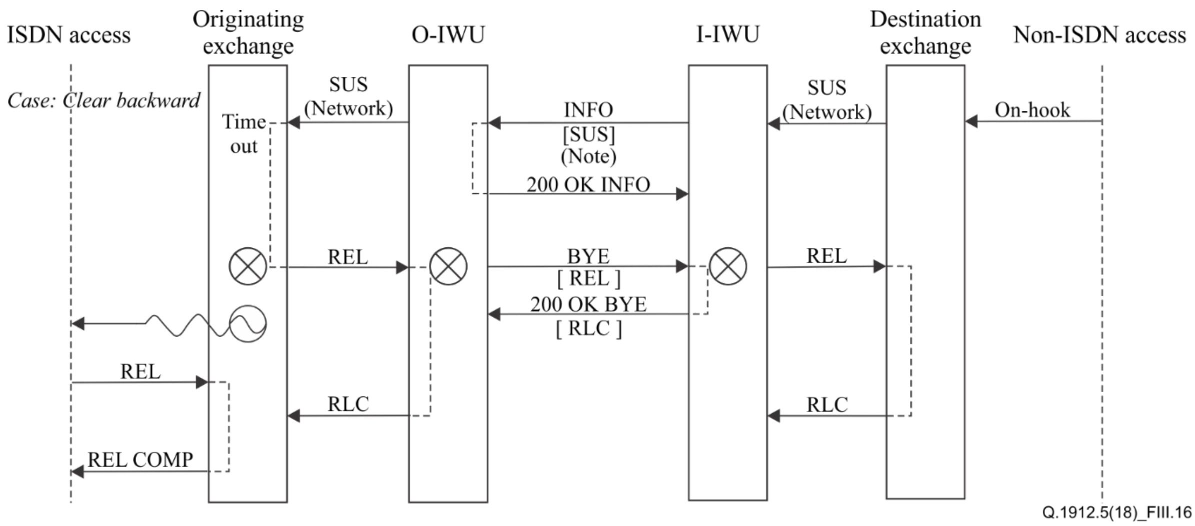
Figure III.15 – Normal call release procedure without tone provision

For detailed messages and parameter mapping, refer to:
– REL message – clauses 6.11.2 and 7.7.3.

III.2.3.2 Normal release with SUS message encapsulation

See clause 2.3 of [ITU-T Q.764] and [IETF RFC 3261].

Figure III.16 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 an INFO request.



NOTE – The transparent transport of SUS is possible only in the case of Profile C (SIP-I) operation.

Figure III.16 – Normal release with SUS message encapsulation

For detailed messages and parameter mapping, refer to:

– SUS message – clause 6.9 (no special interworking at O-IWU).

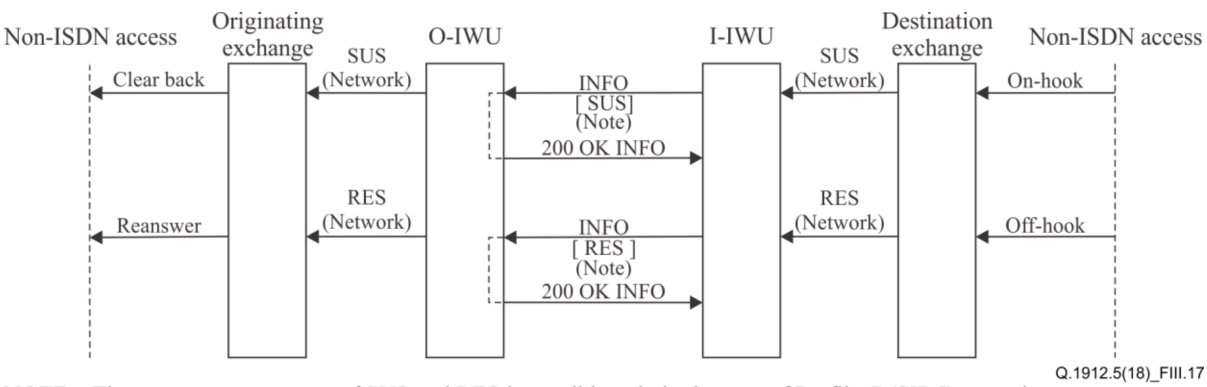
– REL message – clauses 6.11.1 and 7.7.1 2).

III.2.4 Suspend/resume procedures/call flow diagrams for basic call control

III.2.4.1 Suspend/resume non-ISDN access to non-ISDN access

See clause 2.4 of [ITU-T Q.764] and [IETF RFC 3261].

Figure III.17 illustrates suspend and resume procedures for non-ISDN access – non-ISDN access interworking in the case of Profile C (SIP-I) operation. At the I-IWU the SUS message is mapped and encapsulated into an INFO request. At the O-IWU, the RES message is also mapped and encapsulated into an INFO request.



NOTE – The transparent transport of SUS and RES is possible only in the case of Profile C (SIP-I) operation.

Figure III.17 – Suspend/resume non-ISDN access to non-ISDN access

For detailed messages and parameter mapping, refer to:

– SUS message – clause 6.9.

– RES message – clause 6.10.

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

Appendix IV

Echo canceller scenarios between SIP and ISUP

(This appendix does not form an integral part of this Recommendation.)

IV.1 Scope

This appendix defines typical echo canceller scenarios between SIP and ISUP.

The echo control procedure is used on a per call basis to convey information between exchange nodes about the demand and ability to insert echo control devices.

The procedure is invoked when a call is to be routed on a connection for which echo control is necessary. It could be initiated at the originating exchange or at an intermediate exchange.

IV.2 Definitions

The vertical boxes represent two entities: an ISUP exchange and IWU (SIP-ISUP Interworking Unit).

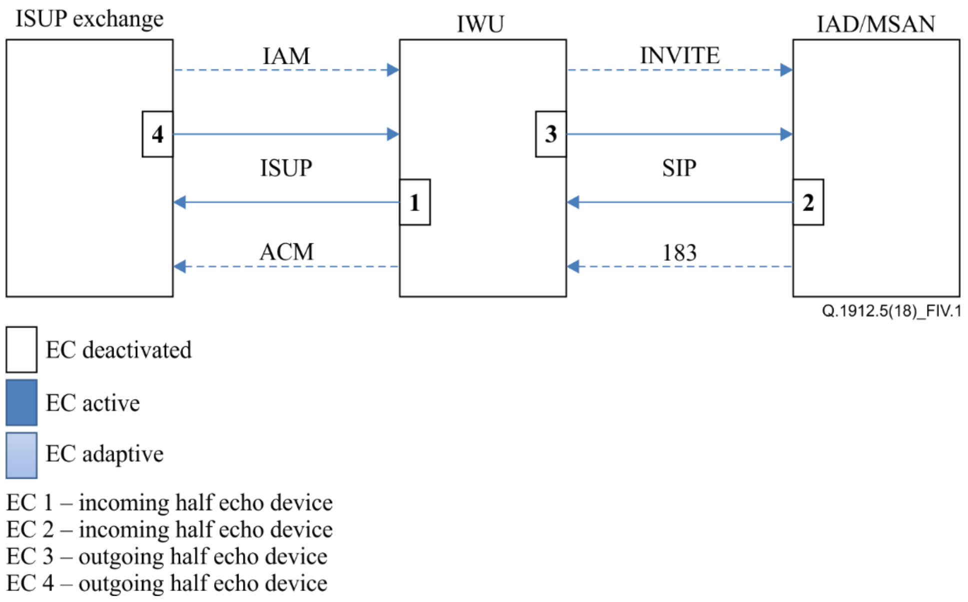


Figure IV.1 – ISUP exchange and IWU

IV.3 Example scenarios for call interworking between SIP and ISUP

IV.3.1 Example scenarios for call interworking from ISUP to SIP at I-IWU

IV.3.1.1 Call set-up procedures and call flow diagrams for call interworking from ISUP to SIP at I-IWU

Figure IV.2 shows the sequence of messages for successful call set-up for an incoming call from ISUP to SIP NNI. If the ISUP Trunk receives "OECD included", the IWU outgoing half echo device (3) will be deactivated and the incoming half echo device (1) will be deactivated.

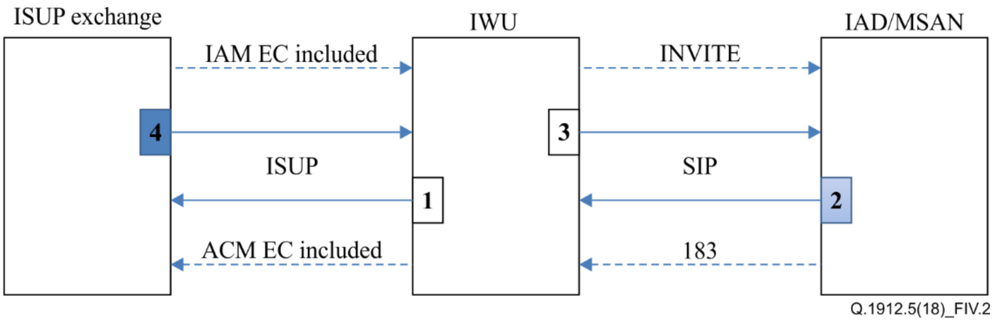


Figure IV.2 – Sequence of messages for successful call set-up for an incoming call from ISUP to SIP NNI, "OECD included"

Figure IV.3 shows the sequence of messages for successful call set-up for an incoming call from ISUP to SIP NNI. If the ISUP Trunk receives "OECD not included", the IWU outgoing half echo device (3) will be activated incoming half echo device (1) will be deactivated.

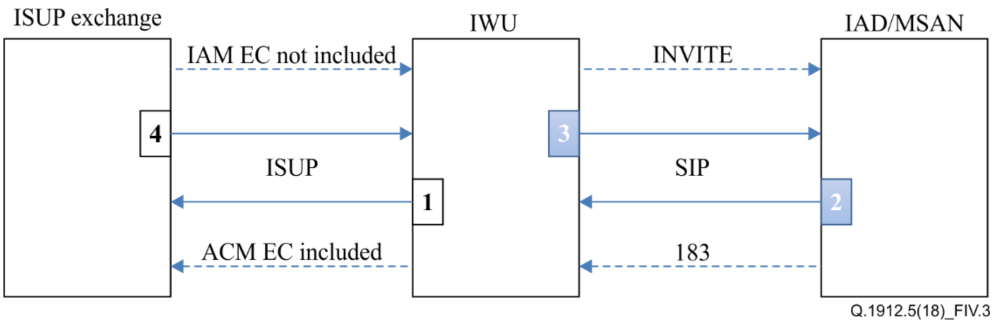


Figure IV.3 – Sequence of messages for successful call set-up for an incoming call from ISUP to SIP NNI, "OECD not included"

IV.3.1.2 Call set-up procedures and call flow diagrams for call interworking from ISUP to SIP-I at I-IWU

Figure IV.4 shows the sequence of messages for successful call set-up for an incoming call from ISUP to SIP-I. If the ISUP Trunk receives "OECD included", the IWU the outgoing half echo device (3) will be deactivated, the incoming half echo device (1) will be deactivated.

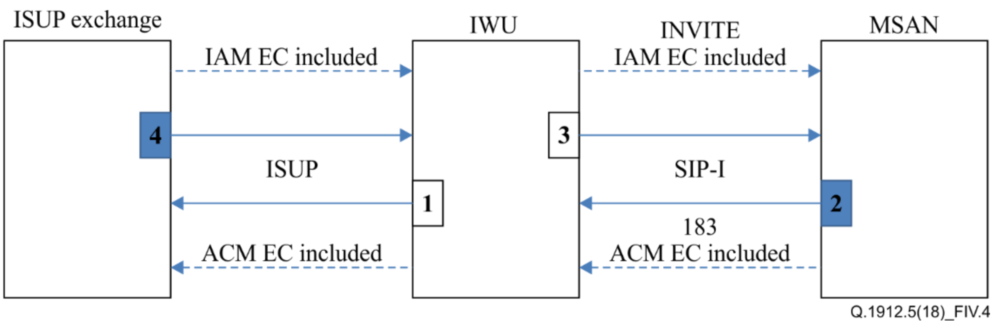


Figure IV.4 – Sequence of messages for successful call set-up for an incoming call from ISUP to SIP-I, "OECD included"

Figure IV.5 shows the sequence of messages for successful call set-up for an incoming call from ISUP to SIP-I. If the ISUP Trunk receives "OECD not included", the IWU the outgoing half echo device (3) will be activated, the incoming half echo device (1) will be deactivated.

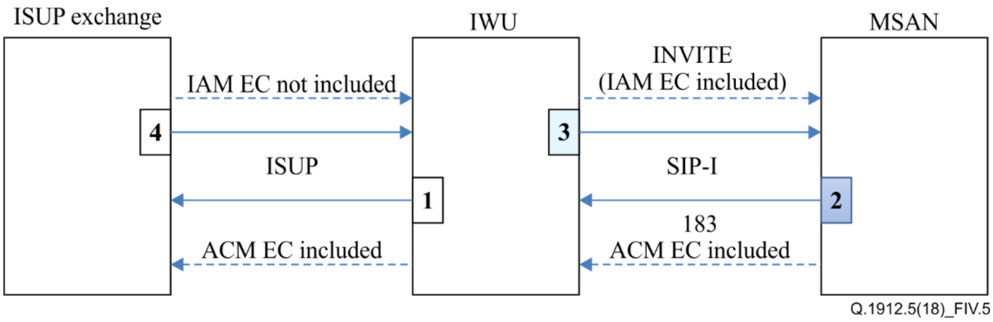


Figure IV.5 – Sequence of messages for successful call set-up for an incoming call from ISUP to SIP-I, "OECD not included"

IV.3.2 Example scenarios for call interworking from SIP to ISUP at I-IWU

IV.3.2.1 Call set-up procedures and call flow diagrams for call interworking from SIP to ISUP at I-IWU

Figure IV.6 shows the sequence of messages for successful call set-up for an incoming call from SIP to ISUP. If for the outgoing ISUP Trunk "IECD included", will be received the IWU the outgoing half echo device (3) will be deactivated, the incoming half echo device (1) will be deactivated.

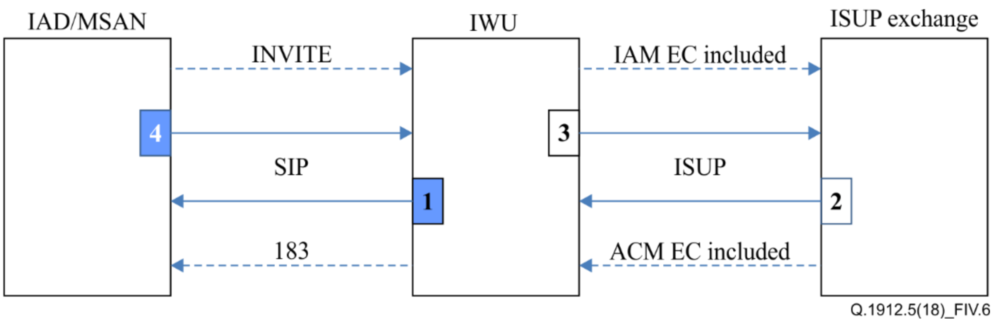


Figure IV.6 – Sequence of messages for successful call set-up for an incoming call from SIP to ISUP, "IECD included"

Figure IV.7 shows the sequence of messages for successful call set-up for an incoming call from SIP-II NNI to ISUP. If for the outgoing ISUP Trunk "IECD not included", will be received the IWU the outgoing half echo device (3) will be deactivated, the incoming half echo device (1) will be activated.

On TDM side of the IWU the setting of the "outgoing echo device included" must be trunk based controlled depending of the carrier.

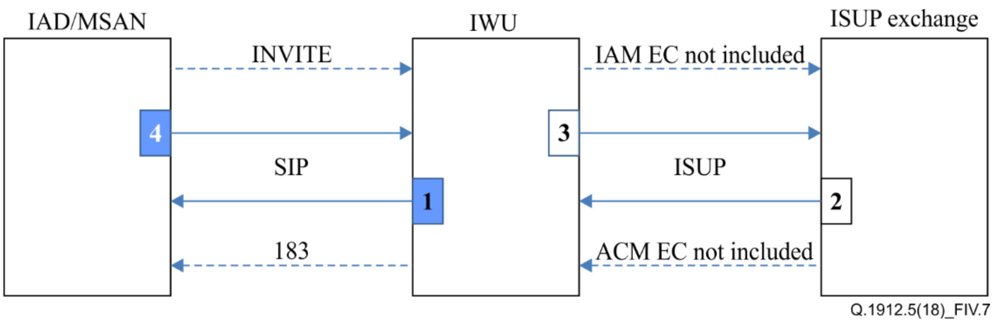


Figure IV.7 – Sequence of messages for successful call set-up for an incoming call from SIP‑II NNI to ISUP, "IECD not included"

IV.3.2.2 Call set-up procedures and call flow diagrams for call interworking from SIP-I to ISUP at I-IWU

Figure IV.8 shows the sequence of messages for successful call set-up for an incoming call from SIP-I to ISUP. If the ISUP Trunk receives "OECD included", the IWU the outgoing half echo device (3) will be deactivated, the incoming half echo device (1) will be deactivated.

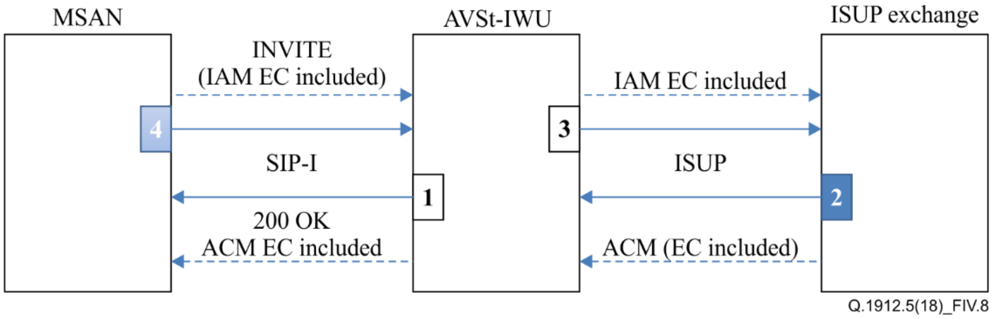


Figure IV.8 – Sequence of messages for successful call set-up for an incoming call from SIP-I to ISUP, "OECD included"

Figure IV.9 shows the sequence of messages for successful call set-up for an incoming call from SIP-I to ISUP. If the ISUP Trunk receives "OECD not included", the IWU the outgoing half echo device (3) will be activated, the incoming half echo device (1) will be deactivated.

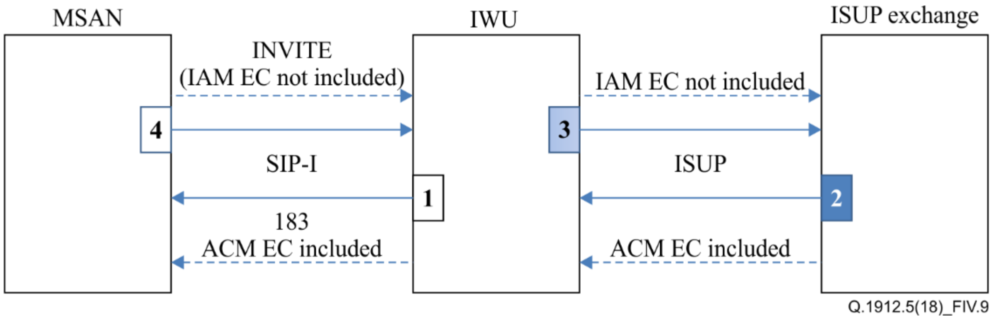


Figure IV.9 – Sequence of messages for successful call set-up for an incoming call from SIP-I to ISUP, "OECD not included"

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