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European Standard (Telecommunications series)

Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control (BICC) Protocol or ISDN User Part (ISUP)

[ITU-T Recommendation Q.1912.5, modified]

Reference

DEN/TISPAN-03008-NGN-R1

Keywords

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Foreword

This European Standard (Telecommunications series) has been produced by ETSI Technical Committee Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN), and is now submitted for the Vote phase of the ETSI standards Two-step Approval Procedure.

Proposed national transposition dates	
Date of latest announcement of this EN (doa):	3 months after ETSI publication
Date of latest publication of new National Standard or endorsement of this EN (dop/e):	6 months after doa
Date of withdrawal of any conflicting National Standard (dow):	6 months after doa

1 Scope

The present document provides the ETSI endorsement of the ITU-T Recommendation Q.1912.5 [1].

ITU-T Recommendation Q.1912.5 [1] 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.

The present document is applicable to ETSI PSTN/ISDN networks interworking with networks based on an IETF based SIP/SDP profile as defined in annex C of ITU-T Recommendation Q.1912.5 [1].

In the case where an IMS-based network interworks with the PSTN/ISDN, then the ETSI endorsement of 3GPP TS 29.163, in either ETSI TS 129 163 [29] and ETSI ES 283 027 as appropriate to the applicability of each document, takes precedence.

Formats, codes and procedures marked for national use or as network option are included for informative purposes for the international interface specification. If these items so marked are supported within a national network and operator's network, then it is proposed that they are supported in this manner.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication and/or edition number or version number) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

[1] ITU-T Recommendation Q.1912.5 (2004): "Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part".

[2] ETSI EN 302 213 (V1.1.2): "Services and Protocols for Advanced Networks (SPAN); Bearer Independent Call Control (BICC) Capability Set 2 (CS2); Protocol specification [ITU-T Recommendations Q.1902.1, Q.1902.2, Q.1902.3, Q.1902.4, Q.1902.5, Q.1902.6, Q.765.5 Amendment 1, Q.1912.1, Q.1912.2, Q.1912.3, Q.1912.4, Q.1922.2, Q.1950, Q.1970, Q.1990, Q.2150.0, Q.2150.1, Q.2150.2, Q.2150.3, modified]".

[3] ETSI EN 300 356-1 (V4.2.1): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 1: Basic services [ITU-T Recommendations Q.761 to Q.764 (1999) modified]".

[4] ETSI EN 300 356-3 (V4.2.1): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 3: Calling Line Identification Presentation (CLIP) supplementary service [ITU-T Recommendation Q.731, clause 3 (1993) modified]".

[5] ETSI EN 300 356-4 (V4.2.1): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 4: Calling Line Identification Restriction (CLIR) supplementary service [ITU-T Recommendation Q.731, clause 4 (1993) modified]".

[6] ETSI EN 300 356-5 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 5: Connected Line Identification Presentation (COLP) supplementary service [ITU-T Recommendation Q.731, clause 5 (1993) modified]".

[7] ETSI EN 300 356-6 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 6: Connected Line Identification Restriction (COLR) supplementary service [ITU-T Recommendation Q.731, clause 6 (1993) modified]".

[8] ETSI EN 300 356-7 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 7: Terminal Portability (TP) supplementary service [ITU-T Recommendation Q.733, clause 4 (1993) modified]".

[9] ETSI EN 300 356-8 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 8: User-to-User Signalling (UUS) supplementary service [ITU-T Recommendation Q.737, clause 1 (1997) modified]".

[10] ETSI EN 300 356-9 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 9: Closed User Group (CUG) supplementary service [ITU-T Recommendation Q.735, clause 1 (1993) modified]".

[11] ETSI EN 300 356-10 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 10: Subaddressing (SUB) supplementary service [ITU-T Recommendation Q.731, clause 8 (1992) modified]".

[12] ETSI EN 300 356-11 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 11: Malicious Call Identification (MCID) supplementary service [ITU-T Recommendation Q.731, clause 7 (1997) modified]".

[13] ETSI EN 300 356-12 (V4.2.1): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 12: Conference call, add-on (CONF) supplementary service [ITU-T Recommendation Q.734, clause 1 (1993) and implementors guide (1998) modified]".

[14] ETSI EN 300 356-14 (V4.2.1): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 14: Explicit Call Transfer (ECT) supplementary service [ITU-T Recommendation Q.732, clause 7 (1996) and implementors guide (1998) modified]".

[15] ETSI EN 300 356-15 (V4.2.1): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 15: Diversion supplementary service [ITU-T Recommendation Q.732, clauses 2 to 5 (1999) modified]".

[16] ETSI EN 300 356-16 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 16: Call Hold (HOLD) supplementary service [ITU-T Recommendation Q.733, clause 2 (1993) modified]".

[17] ETSI EN 300 356-17 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 17: Call Waiting (CW) supplementary service [ITU-T Recommendation Q.733, clause 1 (1992) modified]".

[18] ETSI EN 300 356-18 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 18: Completion of Calls to Busy Subscriber (CCBS) supplementary service [ITU-T Recommendation Q.733, clause 3 (1997) modified]".

[19] ETSI EN 300 356-19 (V4.2.1): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 19: Three-Party (3PTY) supplementary service [ITU-T Recommendation Q.734, clause 2 (1996) and implementors guide (1998) modified]".

[20] ETSI EN 300 356-20 (V4.3.1): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 20: Completion of Calls on No Reply (CCNR) supplementary service [ITU-T Recommendation Q.733, clause 5 (1999) modified]".

[21] ETSI EN 300 356-21: "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 21: Anonymous Call Rejection (ACR) supplementary service [ITU-T Recommendation Q.731, clause 4 (1993)]".

[22] ETSI EN 300 485 (V1.2.3): "Integrated Services Digital Network (ISDN); Definition and usage of cause and location in Digital Subscriber Signalling System No. one (DSS1) and Signalling System No.7 ISDN User Part (ISUP) [ITU-T Recommendation Q.850 (1998), modified]".

[23] IETF RFC 3261 (2002): "SIP: Session Initiation Protocol".

[24] IETF RFC 3264 (2002): "An Offer/Answer Model with Session Description Protocol (SDP)".

[25] IETF RFC 3323 (2002): "A Privacy Mechanism for the Session Initiation Protocol (SIP)".

[26] ITU-T Recommendation T.38: "Procedures for real-time Group 3 facsimile communication over IP networks".

[27] IETF RFC 4040 (2005): "RTP Payload Format for a 64 kbit/s Transparent Call".

[28] IETF RFC 3966 (2004): "The tel URI for Telephone Numbers".

[29] ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163)".

3 Definitions and abbreviations

For the purposes of the present document, the terms, definitions and abbreviations given in [1] apply.

Endorsement notice

The elements of ITU-T Recommendation Q.1912.5 [1] apply, with the following modifications:

NOTE: Underlining and/or strike-out are used to highlight detailed modifications where necessary.

Global modifications to ITU-T Recommendation Q.1912.5

Throughout the text of ITU-T Recommendation Q.1912.5

Replace references as shown below.

Reference in ITU-T Recommendation Q.1912.5 [1]	Modified reference
ITU-T Recommendation Q.731.3	ITU-T Recommendation Q.731.3 as modified by EN 300 356-3 [4]
ITU-T Recommendation Q.731.4	ITU-T Recommendation Q.731.4 as modified by EN 300 356-4 [5]
ITU-T Recommendation Q.731.5	ITU-T Recommendation Q.731.5 as modified by EN 300 356-5 [6]
ITU-T Recommendation Q.731.6	ITU-T Recommendation Q.731.6 as modified by EN 300 356-6 [7]
ITU-T Recommendation Q.731.7	ITU-T Recommendation Q.731.7 as modified by EN 300 356-11 [12]
ITU-T Recommendation Q.731.8	ITU-T Recommendation Q.731.8 as modified by EN 300 356-10 [11]
ITU-T Recommendation Q.732.2	ITU-T Recommendation Q.732.2 as modified by EN 300 356-15 [15]
ITU-T Recommendation Q.732.3	ITU-T Recommendation Q.732.3 as modified by EN 300 356-15 [15]
ITU-T Recommendation Q.732.4	ITU-T Recommendation Q.732.4 as modified by EN 300 356-15 [15]
ITU-T Recommendation Q.732.5	ITU-T Recommendation Q.732.5 as modified by EN 300 356-15 [15]
ITU-T Recommendation Q.732.7	ITU-T Recommendation Q.732.7 as modified by EN 300 356-14 [14]

Reference in ITU-T Recommendation Q.1912.5 [1]	Modified reference
ITU-T Recommendation Q.733.1	ITU-T Recommendation Q.733.1 as modified by EN 300 356-17 [17]
ITU-T Recommendation Q.733.2	ITU-T Recommendation Q.733.2 as modified by EN 300 356-16 [16]
ITU-T Recommendation Q.733.3	ITU-T Recommendation Q.733.3 as modified by EN 300 356-18 [18]
ITU-T Recommendation Q.733.4	ITU-T Recommendation Q.733.4 as modified by EN 300 356-7 [8]
ITU-T Recommendation Q.733.5	ITU-T Recommendation Q.733.5 as modified by EN 300 356-20 [20]
ITU-T Recommendation Q.734.1	ITU-T Recommendation Q.734.1 as modified by EN 300 356-12 [13]
ITU-T Recommendation Q.734.2	ITU-T Recommendation Q.734.2 as modified by EN 300 356-19 [19]
ITU-T Recommendation Q.735.1	ITU-T Recommendation Q.735.1 as modified by EN 300 356-9 [10]
ITU-T Recommendation Q.737.1	ITU-T Recommendation Q.737.1 as modified by EN 300 356-8 [9]
ITU-T Recommendation Q.761	ITU-T Recommendation Q.761 as modified by EN 300 356-1 [3]
ITU-T Recommendation Q.762	ITU-T Recommendation Q.762 as modified by EN 300 356-1 [3]
ITU-T Recommendation Q.763	ITU-T Recommendation Q.763 as modified by EN 300 356-1 [3]
ITU-T Recommendation Q.764	ITU-T Recommendation Q.764 as modified by EN 300 356-1 [3]
ITU-T Recommendation Q.850	ITU-T Recommendation Q.850 as modified by EN 300 485 [22]
ITU-T Recommendation Q.1902.1	ITU-T Recommendation Q.1902.1 as modified by EN 302 213 [2]
ITU-T Recommendation Q.1902.2	ITU-T Recommendation Q.1902.2 as modified by EN 302 213 [2]
ITU-T Recommendation Q.1902.3	ITU-T Recommendation Q.1902.3 as modified by EN 302 213 [2]
ITU-T Recommendation Q.1902.4	ITU-T Recommendation Q.1902.4 as modified by EN 302 213 [2]
ITU-T Recommendation Q.1912.5 [1]	ITU-T Recommendation Q.1912.5 [1] as modified by the present document
IETF RFC 2806	IETF RFC 3966 [28]. NOTE: RFC 2806 is obsolete. RFC 3966 replaces RFC 2806.

General

Throughout the present document "should" is replaced by "shall".

Clause 1

Modify 1st paragraph after Figure 2:

~~ITU-T Supplement 45 to Q-series Recommendations (TRQ.2815) specifies the set of common capabilities supported by the interworking between SIP and BICC/ISUP for three different profiles (A, B, and C) in forms of Tables. Tables 1 and 2 of Supplement 45 (TRQ.2815) specify interworking capabilities for Profile A, Tables 3 and 4 specify interworking capabilities for Profile B, and Tables 5 and 6 specify 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.

NOTE: The profiles A,B and C are described within Annex C.1

Clause 5.3.3

Modify 1st paragraph:

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.~~

Clause 6.1.3.1

Modify Table 3:

Table 3/Q.1912.5 - Coding of the Called Party Number

INVITE→	IAM→
Request-URI	Called Party Number
	Odd/even indicator: set as required
	Nature of address indicator: Analyse the information contained in received URI with user=phone, and if it is in the format:- **+CC NDC SN** where CC is the country code of the network in which the next hop terminates, then set Nature of Address indicator to 0 0 0 0 0 1 1 "National (significant) number" , remove "+CC" and use the remaining digits to fill the Address signals". **+CC NDC SN** where CC is not the country code of the network in which the next hop terminates, then set Nature of Address indicator to 0 0 0 0 1 0 0 "International number", remove "+" and use the remaining digits to fill the Address signals.
	Internal Network Number Indicator: 1 routing to internal network number not allowed
	Numbering plan Indicator: 001 ISDN (Telephony) numbering plan (Rec. E.164)
userinfo (sip: URI with user=phone)	Address Signals
NOTE: RFC 3966 [28] describes the tel format of a userportion.	

Clause 6.1.3.3

Modify 2nd paragraph:

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

Clause 6.1.3.4

Replace 3rd and 4th paragraphs:

For Profile A and B, the following mapping shall apply: ~~indicator values in Table 5 should~~ shall ~~be set by the I-IWU as default in the FCI parameter:~~

~~**Table 5/Q.1912.5 - Default values for Forward Call Indicators**~~

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

~~For Profile 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.~~

Forward call indicators

bits CB End-to-end method indicator

0 0 no end-to-end method available (only link-by-link method available)

bit D Interworking indicator

1 interworking encountered

As a network operator option, the value D = 0 "No interworking encountered" is used in case where the TMR = 64 kBit/s unrestricted is used.

NOTE: This avoids the sending of a Progress indicator with Progress information 0 0 0 0 0 0 1 1."Call is not end-to-end ISDN; further call progress information may be available in band ",, so the call will not be released for that reason at an ISDN terminal.

bit E End-to-end information indicator (national use)

0 no end-to-end information available

bit F ISDN user part/BICC indicator

0 ISDN user part/BICC not used all the way

As a network operator option, the value F = 1 "ISDN user part/BICC used all the way" is used in case where the TMR = 64 kBit/s unrestricted is used.

NOTE: This avoids the sending of a Progress indicator with Progress information 0 0 0 0 0 0 1 1. "Call is not end-to-end ISDN; further call progress information may be available in band ", so the call will not be released for that reason at an ISDN terminal.

bits HG ISDN user part/BICC preference indicator

0 1 ISDN user part/BICC not required all the way

bit I ISDN access indicator

0 originating access non-ISDN

As a network operator option, the value I = 1 "originating access ISDN" is used in case where the TMR = 64 kBit/s unrestricted is used.

NOTE: This avoids the sending of a Progress indicator with Progress information 0 0 0 0 0 1 1 3. "Originating access is non-ISDN", so the call will not be released for that reason at an ISDN terminal..

bits KJ SCCP method indicator

0 0 no indication bits CB End-to-end method indicator

0 0 no end-to-end method available (only link-by-link method available)

Clause 6.1.3.5

Modify title 6.1.3.5:

6.1.3.5 Transmission Medium Requirement (mandatory), User Service Information ~~(optional),~~ and Higher Layer Compatibility information element within Access Transport Parameter ~~(optional)~~

Replace text of clause 6.1.3.5 by the following text:

6.1.3.5.1 Profile A

In all instances the TMR parameter shall be set to the value "3.1 kHz audio", the USI and the Access Transport parameters shall not be sent.

Transcoding shall be applied if required.

6.1.3.5.2 Profile B

6.1.3.5.2.1 I-IWU not acting as an International Gateway

If transcoding is not supported at the I-IWU, and a SDP is received from the remote peer before the IAM is sent, then the TMR, USI and HLC shall be derived from SDP as described in clause 6.1.3.5.4.

6.1.3.5.2.2 I-IWU acting as an International Gateway

If a SDP is received from the remote peer before the IAM is sent, then the TMR and HLC shall be derived from SDP as described in clause 6.1.3.5.4 below.

NOTE: Only a-Law shall be supported.

If the incoming call is an ISDN originated call and a G.711 codec is used, 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 incoming call is not an ISDN originated call, then the USI parameter shall not be sent.

The offer-answer procedures for the G.711 codec are modified as follows:

- If both G.711 a-law and μ-law codecs are received in the SDP offer, then independent from the received order of preference the G.711 a-law codec shall be returned in the SDP answer as the preferred codec.
- If G.711 a-law codec is received in the SDP offer without u-law codec, then the normal offer answer procedures apply.
- If G.711 μ -law codec is received in the SDP offer without an a-law codec, then the u-law codec shall be rejected.

6.1.3.5.3 Profile C (SIP-I)

The TMR, USI and HLC, if present, shall be taken from the encapsulated ISUP to populate the associated ISUP parameters.

Add clause 6.1.3.5.4 Transcoding not available at the I-IWU (Profile B only)

The SDP Media Description Part received by the I-IWU should indicate only one media stream. If more than one media stream is indicated the following rules are valid:

- based on operator policy the call can be refused with a 415 Unsupported media type response sent back; or
- if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered ; the other streams shall be rejected in accordance with the procedures of RFC 3264 [24]; and
- if the SDP offer contains several audio type media streams, the IWU shall only consider one, and reject the other streams in accordance with RFC 3264 [24].

Modify Table 6:

Table 6/Q.1912.5 - Coding of TMR/USI/HLC from SDP: SIP to BICC/ISUP

m= line			b= line (Note 4)	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.	rtpmap:<dynamic-PT> <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~~	~~*"3.1kHz audio"*~~	~~*"3.1kHz audio"*~~	~~*"G.711 μ-law"*~~	~~(Note 3)~~
~~audio~~	~~RTP/AVP~~	~~Dynamic PT~~	~~N/A or up to 64 kbit/s~~	~~rtpmap:<dynamic-PT> PCMU/8000~~	~~*"3.1kHz audio"*~~	~~*"3.1kHz audio"*~~	~~*"G.711 μ-law"*~~	~~(Note 3)~~
audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A	*"3.1kHz audio"*	*"3.1kHz audio"*	*"G.711 A-law"*	(Note 3)
audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMA/8000	*"3.1kHz audio"*	*"3.1kHz audio"*	*"G.711 A-law"*	(Note 3)
audio	RTP/AVP	9	AS:64 kbit/s	rtpmap:9 G722/8000	*"64 kbit/s unrestricted"*	*"Unrestricted digital inf. w/tones/ann"*		
audio	RTP/AVP	Dynamic PT	AS:64 kbit/s	rtpmap:<dynamic-PT> CLEARMODE/8000 (Note 2)	*"64 kbit/s unrestricted"*	*"Unrestricted digital information"*		
image	udptl	t38	N/A or up to 64 kbit/s	Based on T.38 [26]	*"3.1 kHz audio"*	*"3.1kHz audio"*		*"Facsímile Group 2/3"*
image	tcptl	t38	N/A or up to 64 kbit/s	Based on T.38 [26]	*"3.1 kHz audio"*	*"3.1kHz audio"*		*"Facsímile 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 RFC 4040 [27].~~has not yet been standardized; and its usageFFS~~
NOTE 3 - HLC is normally absent in this case. It is possible for HLC to be present with the value "Telephony", although 6.3.1/Q.939 indicates that this would normally be accompanied by a value of "Speech" for the Information Transfer Capability element.
NOTE 4 - If the b=line indicates a bandwidth greater than 64kbit/s then the call may either use compression techniques or be rejected by returning a 415 response indicating only one media stream of 64kbit/s is supported.

Clause 6.1.3.6

Modify the 3rd paragraph and then add a new paragraph as follows:

…………..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.~~

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 not the same as the respective value contained in the encapsulated ISUP, then the encapsulated ISUP address shall take precedence.

NOTE: If the SIP address information has changed due to a SIP application elsewhere in the SIP domain and this is the address information that needs to be delivered to the called party, then it is assumed that the SIP application will have de-encapsulated the ISUP protocol and invoked an ISUP state machine to change the ISUP CLI information to be consistent with the SIP address information.

Modify 4th paragraph:

If ~~Should~~ any discrepancy occurs in privacy settings during the alignment process the strongest privacy setting shall prevail.

Modify Table 7:

Table 7/Q.1912.5 - 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 8) or omit the Address Signals. (Note 4)	If a Privacy header field was received, set APRI as indicated in Table 9, otherwise, network option to set APRI to "*presentation restricted by the network*" 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 8) or omit the Address Signals. (Note 4)	If a Privacy header field was received, set APRI as indicated in Table 9, otherwise, network option to set APRI to either "*presentation restricted by the network*" or "*presentation allowed*" (Note 4)	Network option to either omit the parameter (if CgPN has been omitted) or derive from the SIP From if it is originated by a trusted network (see Table 10) (Note 1)	See Table 10
Yes	No	Derive from SIP P-Asserted-Identity (See Table 9)	APRI = "*presentation restricted*" or "*presentation allowed*" depending on SIP Privacy header. (See Table 9)	Parameter ~~N~~not included	Not applicable

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
Yes	Yes	Derived from SIP P-Asserted-Identity (See Table 9)	APRI = "*presentation restricted*" or "*presentation allowed*" depending on SIP Privacy. (See Table 9)	Network Option to either omit the parameter or derive from the SIP From if it is originated by a trusted network (Note 1) (See Table 10)	APRI = "*presentation restricted*" or "*presentation allowed*" depending on SIP Privacy. (See Table 10)
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.~~ The Tel URI shall take precedence if the SIP URI is received without user = phone. 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. RFC 3261 [23] recommends that the display-name component contain "*Anonymous*". RFC 3323 [25] 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"*.					

Clause 6.1.3.6.1

Modify Table 8:

Table 8/Q.1912.5 - 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"*~~ Set according to the length of the address signals
Numbering Plan Indicator	*"ISDN/Telephony (E.164)"*
Address Presentation Restricted Indicator	*"Presentation allowed/restricted by the network*" (see Table 7)
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 shall~~ould~~ be included. If NOA is "*international number",* then the country code of the network-provided number shall~~ould~~ be included.

Modify Table 9, last row:

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~~. However the information included in the tel URI is the userinfo component of the SIP URI. The Tel URI shall take precedence if the SIP URI is received without user = phone.
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*".

Clause 6.1.3.6.2

Modify Table 10:

Table 10/Q.1912.5 - 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 (Recommendation E.164)*"
-	-	Address Presentation Restricted Indicator (APRI)	Use same setting as for calling party number <u>unless Calling party number APRI = "presentation restricted by network" then set GN APRI to "presentation allowed".</u>
-	-	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

Clause 6.1.3.7

Modify title 6.1.3.7:

6.1.3.7 User Service Information ~~(Optional)~~

Clause 6.1.3.9

Modify title 6.1.3.9:

6.1.3.9 Hop Counter ~~(Optional)~~

Clause 6.11.1

Modify clause 6.11.1 5th paragraph 1st sentence:

If the Reason header field with Q.850 Cause Value is included in the BYE or CANCEL, then the Cause Value ~~may~~ <u>shall</u> be mapped to the ISUP Cause Value field in the ISUP REL~~ depending on local policy~~.

NOTE: The above mapping to a Reason header in SIP responses does not form an accepted part of the SIP, and alternative solutions are being addressed in IETF, despite being endorsed in Q.1912.5. It is therefore not expected that a SIP terminal will support these procedures, and these procedures should only be used between two SIP/ISUP gateways. A future revision of this document will provide a solution conformant with SIP."

Modify clause 6.11.1 5th paragraph, 2nd sentence:

'The mapping of the ~~Cause Indicators~~<u>Reason Header</u> parameter to the ~~Reason header~~<u>Cause Indicators</u> is shown in Table 18.'

Clause 6.11.2

Modify clause 6.11.2, 1st paragraph:

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,~~ When the internal bearer path has been disconnected then an ISUP RLC is returned to the ISUP side.

NOTE: The above mapping to a Reason header in SIP responses does not form an accepted part of the SIP, and alternative solutions are being addressed in IETF, despite being endorsed in Q.1912.5. It is therefore not expected that a SIP terminal will support these procedures, and these procedures should only be used between two SIP/ISUP gateways. A future revision of this document will provide a solution conformant with SIP."

Modify clause 6.11.2, 4th paragraph, last sentence:

~~Depending on local policy a~~A Reason header field containing the received (Q.850) Cause Value of the REL ~~may~~ shall be added to the SIP final response or BYE sent as a result of this clause.

NOTE: The above mapping to a Reason header in SIP responses does not form an accepted part of the SIP, and alternative solutions are being addressed in IETF, despite being endorsed in Q.1912.5. It is therefore not expected that a SIP terminal will support these procedures, and these procedures should only be used between two SIP/ISUP gateways. A future revision of this document will provide a solution conformant with SIP."

Modify Table 20 last sentence:

NOTE 2: Due to the fact that the Cause Indications parameter does not include the definition text as defined in Table1/Q.850 this is based on provisioning in the ~~O~~ I-IWU.

Clause 6.11.3

Modify clause 6.11.3, 4th paragraph, 1st sentence:

~~Depending on local policy a~~ A Reason header field containing the (Q.850) Cause Value of the REL message sent by the I-IWU ~~may~~ shall be added to the SIP Message (BYE or final response) sent by the SIP side of the I-IWU.

NOTE: The above mapping to a Reason header in SIP responses does not form an accepted part of the SIP, and alternative solutions are being addressed in IETF, despite being endorsed in Q.1912.5. It is therefore not expected that a SIP terminal will support these procedures, and these procedures should only be used between two SIP/ISUP gateways. A future revision of this document will provide a solution conformant with SIP."

Clause 6.11.4

Modify clause 6.11.4 5th paragraph:

On receipt of a GRS or CGB message, one SIP message is sent for each call association. Therefore, multiple (Note 1) SIP messages may be sent on receipt of a single GRS or CGB message.

NOTE 1: i.e. In the case were Range subfield of the Range and Status Parameter contains a value equal to or than "1" multiple SIP messages will be sent on receipt of a single GRS or CGB message.

Clause 6.11.5

Modify clause 6.11.5 5th paragraph:

On receipt of a GRS message, one SIP message is sent for each call association. Therefore, multiple SIP messages (Note 1) may be sent on receipt of a single GRS message.

NOTE 1: I.e. In the case were the Range and Status Parameter value is bigger than *"1"* multiple SIP messages will be send on receipt of a single GRS message.

Clause 7.1.1

Modify clause 7.1.1 delete 5th paragraph:

- ~~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~~.

Clause 7.1.1.1

Modify clause 7.1.1.1 Table 26:

Table 26/Q.1912.5 - 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>	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>
~~*"speech"*~~	~~*"Speech"*~~	~~*"G.711 µ-law"*~~	~~Ignore~~	~~audio~~	~~RTP/AVP~~	~~0 (and possibly 8)~~ ~~Note 1~~	~~AS:64~~	~~rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)~~ ~~Note 1~~
~~*"speech"*~~	~~*"Speech"*~~	~~*"G.711 µ-law"*~~	~~Ignore~~	~~audio~~	~~RTP/AVP~~	~~Dynamic PT (and possibly a second Dynamic PT)~~ ~~Note 1~~	~~AS:64~~	~~rtpmap:<dynamic-PT> PCMU/8000~~ ~~(and possibly rtpmap:<dynamic-PT> PCMA/8000)~~ ~~Note 1~~
"speech"	*"Speech"*	*"G.711 A-law"*	Ignore	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
"speech"	*"Speech"*	*"G.711 A-law"*	Ignore	audio	RTP/AVP	Dynamic PT	AS:64	rtpmap:<dynamic-PT> PCMA/8000
"3.1 KHz audio"	USI Absent		Ignore	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
~~*"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~~	~~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 2	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
"3.1 KHz audio"	*"3.1 KHz audio"*		*"Facsimile Group 2/3"*	image	udptl	t38	AS:64	Based on T.38 [26].
"3.1 KHz audio"	*"3.1 KHz audio"*		*"Facsimile Group 2/3"*	image	tcptl	t38	AS:64	Based on T.38 [26].
"64 kbit/s unrestricted"	*"Unrestricted digital inf. W/tone/ann."*	N/A	Ignore	audio	RTP/AVP	9	AS:64	rtpmap:9 G722/8000
"64 kbit/s unrestricted"	*"Unrestricted digital information"*	N/A	Ignore	audio	RTP/AVP	Dynamic PT	AS:64	rtpmap:<dynamic-PT> CLEARMODE/8000 Note 1
~~*"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~~

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>	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>
~~*"1536 kbit/s unrestricted"*~~	~~*"Unrestricted digital information"*~~	~~N/A~~	~~Ignore~~	~~FFS~~	~~FFS~~	~~FFS~~	~~FFS~~	~~FFS~~
~~*"1920 kbit/s unrestricted"*~~	~~*"Unrestricted digital information"*~~	~~N/A~~	~~Ignore~~	~~FFS~~	~~FFS~~	~~FFS~~	~~FFS~~	~~FFS~~
~~*"N x 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 7.1.1.~~

NOTE 1~~2~~: ~~Since~~ The usage of the CLEARMODE IETF Draft~~has not yet been standardized, its use~~ is specified in RFC 4040 [27].~~for further study~~.

NOTE 2~~3~~: HLC normally absent in this case. It is possible for HLC to be present with the value "Telephony", although 6.3.1/Q.939 indicates that this would normally be accompanied by a value of "Speech" for the Information Transfer Capability element.

Clause 7.1.2

Modify clause 7.1.2 after 3rd paragraph, add table:

Table 27A/Q.1912.5 - Mapping BICC/ISUP Called Party Number to SIP Request-URI/To header field

IAM	**INVITE**
Called Party Number	**Request-URI/To header field**
Nature of address indicator: **National (significant) number**	**Insert "+CC" before the Address signals** NOTE: CC = Country Code of the network in which the O-IWU is located.
International number	**Insert "+" before the Address signals**

Clause 7.1.3

Add note to Table 27 4th column:

Table 27/Q.1912.5 - Mapping 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?				
	Has a Generic Number ("*additional calling party number*") with a complete E.164 number, with Screening Indicator = "UP~~VP~~NV", and with APRI = "presentation allowed" been received?			
		P-Asserted-Identity header field	From header field: display-name (optional) and addr-spec (Note 4)	Privacy header field
N	N	Header field not included	unavailable@hostportion	Header field not included
N (Note 3)	Y	Header field not included	display-name derived from Generic Number (ACgPN) ~~if possible~~ addr-spec derived from Generic Number (ACgPN) address signals (see Table 30) ~~or uses network provided value~~	Header field not included
Y (Note 1)	N	Derived from Calling Party Number parameter Address Signals (See Table 29)	if APRI = "*presentation allowed*", display-name ~~may~~ shall be derived from Calling Party Number (CgPN) if possible if APRI = "*presentation restricted or address not available*" the, display-name is ~~"Anonymous"~~ omitted if APRI = "*presentation allowed*", addr-spec is derived from Calling Party Number parameter Address Signals (see Table 30) or uses network provided value if APRI = "*presentation restricted or address not available the*", addr-spec is set to the "Anonymous URI" (Note 2) If CgPN APRI = *"presentation restricted by network"*, addr-spec is set to "unavailable@hostportion"	~~If Calling Party Number parameter APRI = "*presentation restricted*" then priv-value includes "*id*". For other APRI settings Privacy header is not included or if included, "id" is not included~~ (See Table 31)
Y	Y	Derived from Calling Party Number parameter Address Signals (See Table 29)	display-name ~~may~~ shall be derived from Generic Number (ACgPN) if possible ~~(Note 2)~~ addr-spec is derived from Generic Number (ACgPN) Address Signals (see Table 28)	~~If Calling Party Number parameter APRI = "*presentation restricted*" then priv-value includes "id" . For other APRI settings Privacy header is not included or if included, "id" is not included~~ (See Table 31)

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?				
	Has a Generic Number ("*additional calling party number*") with a complete E.164 number, with Screening Indicator = "UP~~VP~~NV", and with APRI = "presentation allowed" been received?			
		P-Asserted-Identity header field	From header field: display-name (optional) and addr-spec (Note 4)	Privacy header field
NOTE 1 - A Network Provided CLI in the CgPN parameter may occur on call from any type of access line ~~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 2 - The "From" header may contain an "Anonymous URI". An "Anonymous URI" includes information that does not point to the calling party. RFC 3261 [23] 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 3 - This combination of CgPN and ACgPN is either an error case but is shown here to ensure consistent mapping across different implementations, OR will occur when the CgPN APRI is "presentation restricted by network". NOTE 4: - A display-name can only be included if the O-IWU has access to a database that converts E.164 numbers to subscriber names useful for SIP				

Add clarification in column 4 "Value" of table 28.

Table 28/Q.1912.5 - 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	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 GenericNumber Address Signals used prefixed with a "+" to user portion of URI scheme used.
Address Signals	if NOA is "*national (significant) number"* then the format of the address signals is: NDC + SN If NOA is "*international number"* then the format of the address signals is: CC + NDC + SN	Display-name (optional)	display-name ~~may~~shall be mapped from Address Signals, if ~~possible and~~ network policy allows it.
		Addr-spec	"+" CC NDC SN mapped to user portion of URI scheme used

Add clarification in column 4 "Value" of table 29.

Table 29/Q.1912.5 - 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	Add "+" CC (of the country where the IWU is located) to CgPN Address Signals then map to user portion of URI scheme used.
	"international number"		Map complete CgPN Address Signals used prefixed with a "+" to user portion of URI scheme used.
Address Signals	If NOA is "*national (significant) number"* then the format of the Address Signals is: NDC + SN If NOA is "*international number"* then the format of the address signals is: CC + NDC + SN	display-name (optional)	display-name ~~may~~ shall be mapped from Address Signals, if ~~possible and~~ network policy allows it
		addr-spec	"+" CC NDC SN mapped to the appropriate global number portion of URI scheme used

Add clarification in column 4 "Value" of table 30.

Table 30/Q.1912.5 - 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.
	"international number"		Map complete CgPN Address Signals used prefixed with a "+" to user portion of URI scheme used.
Address Signals	If NOA is "*national (significant) number"* then the format of the Address Signals is: NDC + SN If NOA is "*international number"* then the format of the Address Signals is: CC + NDC + SN	display-name (optional)	Display-name ~~may~~ shall 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

Clause 7.1.4

Modify title 7.1.4:

7.1.4 Hop Counter ~~(Optional)~~

Modify clause 7.1.4. Delete 1st paragraph add section for Profile A, B and C:

~~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.~~

7.1.4.1 Profile A, B and C

~~For Profiles A and B t~~ 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 32, where the factor is constructed according to the following principles:

a) Max-Forwards for a given message ~~should~~shall never increase, and ~~should~~shall 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~~ shall be large enough to accommodate the maximum number of hops that might be expected of a validly routed call.

Table 32/Q.1912.5 - 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~~ shall 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.

7.1.4.2 Additional Procedure for Profile C

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. If the Hop Counter parameter is not available, then the O-IWU shall take no further action.

Clause 7.3.1.1

Replace clause 7.3.1.1, 4th and 5th paragraph and Table 34:

For Profile A and B the following mapping shall apply: ~~, the default settings are shown in Table 34~~:

~~**Table 34/Q.1912.5 - Default Backward Call Indicators settings for Profile A**~~

~~Parameter~~	~~Bits~~	~~Codes~~	~~Meaning~~
~~Interworking Indicator~~	~~I~~	~~1~~	~~*"interworking encountered"*~~
~~ISDN User part/BICC Indicator~~	~~K~~	~~0~~	~~*"ISDN user part/BICC not used all the way"*~~
~~ISDN Access Indicator~~	~~M~~	~~0~~	~~*"terminating access non-ISDN"*~~

~~For Profile B, the O-IWU shall set the appropriate values of other indicators in the Backward Call Indicators parameter (other than Called Party's Status Indicator) based on analysis of various information such as signalling, internal states and/or local policies.~~

Backward call indicators:

bits AB Charge indicator Contributors

1 0 *charge*

bits DC Called party's status indicator

0 1 *subscriber free* if the 180 Ringing has been received.

0 0 *no indication* otherwise

bits FE Called party's category indicator

0 0 no indication

bits HG End-to-end method indicator

00 no end-to-end method available

bit I Interworking indicator
interworking encountered

As a network operator option, the value I = 0 "no interworking encountered" is used in case of TMR = 64 kBit/s unrestricted

NOTE: This avoids the sending of a Progress indicator with Progress information 0 0 0 0 0 0 1 1."Call is not end-to-end ISDN; further call progress information may be available in-band ", so the call will not be released for that reason at an ISDN terminal.

.

bit J End-to-end information indicator

0 no end-to-end information available

bit K ISDN user part/BICC indicator

0 ISDN user part not used all the way

As a network operator option, the value K = 1 "ISDN user part/BICC used all the way" is used in case of TMR = 64 kBit/s unrestricted

NOTE: This avoids the sending of a Progress indicator with Progress information 0 0 0 0 0 0 1 1."Call is not end-to-end ISDN; further call progress information may be available in-band ", so the call will not be released for that reason at an ISDN terminal.

.bit L Holding indicator (national use)

0 holding not requested

bit M ISDN access indicator

0 terminating access non-ISDN

As a network operator option, the value M = 1 "terminating access ISDN" is used in case of TMR = 64 kBit/s unrestricted.

NOTE: This avoids the sending of a Progress indicator with Progress information 0 0 0 0 0 1 0 No 2. " Destination access is non-ISDN", so the call will not be released for that reason at an ISDN terminal.

Clause 7.7.1

Modify clause 7.7.1, last paragraph, 1st sentence:

~~Depending on local policy, a~~A Reason header field containing the received (Q.850) Cause Value of the REL message ~~may~~ shall be added to the CANCEL or BYE request.

Clause 7.7.2

Modify clause 7.7.2, 5th paragraph, 1st sentence:

If a Reason header field with Q.850 Cause Value is included in the BYE, then the Cause Value ~~may~~ shall be mapped to the ISUP Cause Value field in the ISUP REL~~ depending on local policy~~.

Clause 7.7.3

Modify clause 7.7.3, 3rd paragraph:

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

Clause 7.7.4

Modify clause 7.7.4, 2nd paragraph:

On receipt of a GRS or CGB message, one SIP message is sent for each call association. Therefore, multiple (Note 1) SIP messages may be sent on receipt of a single GRS or CGB message.

NOTE 1: I.e. In the case were the Range and Status Parameter value is bigger than *"1"* multiple SIP messages will be sent on receipt of a single GRS or CGB message.

Modify clause 7.7.4, 4th paragraph:

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

Clause 7.7.5

Modify clause 7.7.5, 2nd paragraph:

On receipt of a GRS message, one SIP message is sent for each call association. Therefore, multiple (Note 1) SIP messages may be sent on receipt of a single GRS message.

NOTE 1: I.e. In the case were the Range and Status Parameter value is bigger than *"1"* multiple SIP messages will be sent on receipt of a single GRS message.

Modify clause 7.7.5, 4th paragraph, 1st sentence:

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

Annex B.1

Modify Annex B.1 2nd paragraph:

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~~SIP to ISUP calls), or the priv-value of the 'calling' Privacy header field (in the case of ~~SIP to ISUP~~ISUP to SIP calls), is set to the appropriate 'restriction/privacy' value.

Modify Annex B.1 last paragraph:

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 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 use of ISUP encapsulation. In the contrary case the Calling Party Sub-address is deleted from the ATP.~~The information contained in the encapsulated ISUP Calling Party Number parameter and the encapsulated ISUP Generic Number parameter shall be de-encapsulated and used.

Annex B.10

Interworking of Call Hold (HOLD) supplementary service to SIP networks

Modify annex B.10, paragraph starting with: If the party wants to retrieve the call, then the stream to be retrieved will be marked as:

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~~recvonly media stream, or the attribute may be omitted, since sendrecv is the default
- "a=recvonly", if the stream was previously an inactive media stream

Annex B.17

Interworking of Multi-Level Precedence and Preemption (MLPP) supplementary service to SIP networks

Not supported

Annex B.18

Interworking of Global Virtual Network Service (GVNS) supplementary service to SIP networks

Profile A&B

GVNS is not supported as an ETSI service therefore no Interworking is required.

Profile C

GVNS is not supported as an ETSI service, but the ITU-T parameters can still be used in conjunction with Core INAP CS2. Therefore a traversal of ITU-T Parameters is allowed.

Annex B.19

Interworking of International Telecommunication Charge Card (ITCC) supplementary service to SIP networks

Not supported

Annex B.20

Interworking of Reverse Charging (REV) supplementary service to SIP networks

Not supported

Add Annex B.22

Annex B.22

This section describes the interworking of the ETSI ACR service as described ETSI EN 300 356-21 [21].

Profiles A and B:

ISUP-SIP protocol interworking at the I-IWU

Coding of the mapping of REL to 433 (Anonymity Disallowed)

If ISUP Cause Value field in the ISUP REL includes Cause Value 24 *"call rejected due to ACR supplementary service"* the I-IWU maps this to a 433 (Anonymity Disallowed) response.

SIP-ISUP protocol interworking at the O-IWU

N/A

Profile C (SIP-I): no additional interworking beyond use of ISUP encapsulation required.

Annex ZA (informative): Bibliography

ETSI EN 300 356-2 (V4.2.1): "Integrated Services Digital Network (ISDN); Signalling System No. 7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 2: ISDN supplementary service [ITU-T Recommendation Q.730 (1999) modified]".

IETF RFC 2806 (2000): "URLs for Telephone Calls".

NOTE: RFC 2806 is now replaced by RFC 3966.

IETF RFC 2976 (2000): "The SIP INFO Method".

IETF RFC 3262 (2002): "Reliability of Provisional Responses in Session Initiation Protocol (SIP)".

IETF RFC 3311 (2002): "The Session Initiation Protocol (SIP) UPDATE Method".

IETF RFC 3312 (2002): "Integration of Resource Management and Session Initiation Protocol (SIP)".

IETF RFC 2046 (1996): "Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types".

IETF RFC 3204 (2001): "MIME media types for ISUP and QSIG Objects".

IETF RFC 3325 (2002): "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks".

IETF RFC 3326 (2002): "The Reason Header Field for the Session Initiation Protocol (SIP)".

IETF RFC 2833 (2000): "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals".

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 (2002): "Real-time Transport Protocol (RTP) Payload for Comfort Noise (CN)".

ETSI ES 283 027: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Interworking SIP-ISUP for TISPAN-IMS".

History

Document history				
V1.1.1	June 2005	Public Enquiry	PE 20051014:	2005-06-15 to 2005-10-14
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