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*European Standard (Telecommunications series)*

**Telecommunications and Internet converged Services and  
Protocols for Advanced Networking (TISPAN);  
Interworking for SIP/SIP-T (BICC, ISUP)**

[ITU-T Recommendation Q.1912.5, modified]

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Reference

DEN/TISPAN-03008-NGN-R1

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Keywords

SIP, BICC, ISUP, interworking

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# Contents

|  |           |
|--|-----------|
| Intellectual Property Rights .....                   | 4         |
| Foreword.....  | 4         |
| 1 Scope .....  | 5         |
| 2 References .....                                   | 5         |
| 3 Definitions and abbreviations.....                 | 7         |
| 4 Endorsement notice .....                           | 7         |
| Modifications to ITU-T Recommendation Q.1912.5 ..... | 8         |
| <b>Annex A (informative): Bibliography.....</b>      | <b>24</b> |
| History .....  | 25        |

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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 Public Enquiry phase of the ETSI standards Two-step Approval Procedure.

| <b>Proposed national transposition dates</b>   |                                 |
|--|---------------------------------|
| 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              |

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# 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 cases where an IMS-based network interworks with a PSTN/ISDN network the appropriate ETSI specification 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.

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# 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] IETF RFC 3550 (2003): "RTP: A Transport Protocol for Real-Time Applications".
- [27] IETF RFC 3551 (2003): "RTP Profile for Audio and Video Conferences with Minimal Control".
- [28] ITU-T Recommendation T.38: "Procedures for real-time Group 3 facsimile communication over IP networks".
- [29] IETF RFC 4040 (2005): "RTP Payload Format for a 64 kbit/s Transparent Call".

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## 3 Definitions and abbreviations

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

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## 4 Endorsement notice

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

Replace references as shown below.

| Reference in<br>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] |
| 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]     |

| Reference in<br>ITU-T Recommendation Q.1912.5 [1] | Modified reference  |
|---|---|
| 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 as modified by the present document |

## Modifications to ITU-T Recommendation Q.1912.5

### General

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

### Clause 1

Modify 1<sup>st</sup> 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.~~

### Clause 5.3.3

Modify 1<sup>st</sup> 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: <b>set as required</b>  |
|  | Nature of address indicator:<br>Analyse the information contained in received URI with user=phone, and if it is in the format:-<br><b>+CC NDC SN</b> where CC is the country code of the network in which the next hop terminates, then set Nature of Address indicator to "National (significant) number", remove "+CC" and use the remaining digits to fill the Address signals".<br><b>+CC NDC SN</b> where CC is not the country code of the network in which the next hop terminates, then set Nature of Address indicator to "International number", remove "+" and use the remaining digits to fill the Address signals. |
|  | Internal Network Number Indicator:<br>1 routing to internal network number not allowed  |
|  | Numbering plan Indicator:<br>001 ISDN (Telephony) numbering plan (Rec. E.164)   |
| userinfo<br>(sip: URI with user=phone) | Address Signals   |

**Clause 6.1.3.3**

Modify 2<sup>nd</sup> paragraph:

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

**Clause 6.1.3.4**

Modify 3<sup>rd</sup> and 4<sup>th</sup> paragraphs:

For Profile A and B, the 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  |
|-----------|-----------|--|
| <b>D</b>  | <b>1</b>  | "Interworking encountered".                    |
| <b>F</b>  | <b>0</b>  | "ISDN user part/BICC not used all the way".    |
| <b>HG</b> | <b>01</b> | "ISDN user part/BICC not required all the way" |
| <b>I</b>  | <b>0</b>  | "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.~~

**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.

#### 6.1.3.5.2.1 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 below.

#### 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 G711 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 u-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 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 u-law codec is received in the SDP offer without a-law codec, then 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><br><br>NOTE: <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s. | rtpmap:<dynamic-PT><br><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 $\mu$ -law"                          | (Note 3)                                  |
| audio   | RTP/AVP     | Dynamic PT | N/A or up to 64 kbit/s  | rtpmap:<dynamic-PT><br>PCMU/8000  | "3.1KHz audio"           | "3.1KHz audio"                          | "G.711 $\mu$ -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><br>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><br>CLEARMODE/8000<br>(Note 2)                         | "64 kbit/s unrestricted" | "Unrestricted digital information"      |   |   |
| image   | udptl       | t38        | N/A or up to 64 kbit/s  | Based on T.38 [28]  | "3.1 KHz audio"          | "3.1KHz audio"                          |   | "Facsimile Group 2/3"                     |
| image   | tcptl       | t38        | N/A or up to 64 kbit/s  | Based on T.38 [28]  | "3.1 KHz audio"          | "3.1KHz audio"                          |   | "Facsimile Group 2/3"                     |

NOTE 1 - In this table the codec G.711 is used only as an example. Other codec is possible.

NOTE 2 - CLEARMODE is specified in RFC 4040 [29], has not yet been standardized; and its usage FFS

NOTE 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.

NOTE 4 - If the b=line indicates a bandwidth greater than 64kbit/s then the call may use compression techniques or the reject the call with a 415 response being sent back indicating one media stream of 64kbit/s only supported.

### Clause 6.1.3.6

Modify the 3<sup>rd</sup> 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 an application in the SIP domain and this is the address information that needs to be delivered to the called party. Then this application will have to de-encapsulate the ISUP protocol and invoke an ISUP state machine to change the ISDN CLI information.

Modify 4<sup>th</sup> paragraph:

~~If Should~~ any discrepancy occurs in privacy settings during the alignment process the strongest privacy 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 " <i>presentation restricted by the network</i> " or " <i>presentation allowed</i> " (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 " <i>presentation restricted by the network</i> " or " <i>presentation allowed</i> " (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 = " <i>presentation restricted</i> " or " <i>presentation allowed</i> " depending on SIP Privacy header. (See Table 9)   | Parameter <del>N</del> 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<br>Address Signals  | Calling Party Number parameter<br>APRI   | Generic Number ("Additional calling party number")<br>Address Signals   | Generic Number parameter APRI   |
| Yes   | Yes | Derived from SIP P-Asserted-Identity (See Table 9) | APRI = " <i>presentation restricted</i> " or " <i>presentation allowed</i> " depending on SIP Privacy. (See Table 9) | Network Option to either omit the parameter or derive from the SIP From <u>if it is originated by a trusted network</u> (Note 1) (See Table 10) | APRI = " <i>presentation restricted</i> " or " <i>presentation allowed</i> " 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. <del>The handling of this case is for further study.</del> 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 " <i>Anonymous</i> ". RFC 3323 [25] recommends that the Anonymous URI itself have the value " <i>anonymous@anonymous.invalid</i> ". |     |  |  |   |   |
| NOTE 4 - A national option exists to set the APRI to " <i>Address not available</i> ".  |     |  |  |   |   |

### 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                       | " <i>network provided</i> "  |
| Number Incomplete Indicator               | <del>"complete"</del> Set according to the length of the address signals   |
| Numbering Plan Indicator                  | " <i>ISDN/Telephony (E.164)</i> "  |
| Address Presentation Restricted Indicator | " <i>Presentation allowed/restricted by the network</i> " (see Table 7)  |
| Nature of Address Indicator               | If next BICC/ISUP node is located in the same country set to " <i>national (significant) number</i> " else set to " <i>international number</i> ".   |
| Address Signals                           | If NOA is " <i>national (significant) number</i> " no country code shall be included. If NOA is " <i>international number</i> ", then the country code of the network-provided number shall 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. <del>The handling of this case is for further study.</del> 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 " <i>none</i> ", the other " <i>id</i> ". In this case, APRI shall be set to " <i>presentation restricted</i> ".   |

**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.<br>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 5<sup>th</sup> paragraph 1<sup>st</sup> sentence:

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

**Clause 6.11.2**

Modify clause 6.11.2, 1<sup>st</sup> 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 re-selection, When the internal bearer path has been disconnected ISUP Circuit is released then~~ an ISUP RLC is returned to the ISUP side.

Modify clause 6.11.2, 4<sup>th</sup> 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.

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  $\Theta$  I-IWU.

### Clause 6.11.3

Modify clause 6.11.3, 4<sup>th</sup> paragraph, last sentence:

~~Depending on local policy a~~  $\Delta$  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.

### Clause 6.11.4

Modify clause 6.11.4 5<sup>th</sup> 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 send on receipt of a single GRS or CGB message.

### Clause 6.11.5

Modify clause 6.11.5 4<sup>th</sup> 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 5<sup>th</sup> paragraph:

- ~~If the call is coming from a  $\mu$ -law PSTN network, the O-IWU shall send an SDP Offer with both  $\mu$ -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>:<br><bandwidth-value> | rtpmap:<dynamic-PT><br><encoding name>/<clock rate>/[encoding parameters]                        |
| "speech"                     | "Speech"                                | "G.711 $\mu$ -law"                          | Ignore                                    | audio   | RTP/AVP     | 0 (and possibly 8)<br>Note 1                               | AS:64                            | rtpmap:0 PCMU/8000<br>(and possibly rtpmap:8 PCMA/8000)<br>Note 1                                |
| "speech"                     | "Speech"                                | "G.711 $\mu$ -law"                          | Ignore                                    | audio   | RTP/AVP     | Dynamic PT<br>(and possibly a second Dynamic PT)<br>Note 1 | AS:64                            | rtpmap:<dynamic-PT><br>PCMU/8000<br>(and possibly<br>rtpmap:<dynamic-PT><br>PCMA/8000)<br>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><br>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 $\mu$ -law"                          | Note 3                                    | audio   | RTP/AVP     | 0 (and possibly 8)<br>Note 1                               | AS:64                            | rtpmap:0 PCMU/8000<br>(and possibly rtpmap:8 PCMA/8000)<br>Note 1                                |
| "3.1 KHz audio"              | "3.1 KHz audio"                         | "G.711 A-law"                               | Note 3                                    | 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 [28].  |
| "3.1 KHz audio"              | "3.1 KHz audio"                         |   | "Facsimile Group 2/3"                     | image   | tcptl       | t38  | AS:64                            | Based on T.38 [28].  |
| "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><br>CLEARMODE/8000<br>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  |



| 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>:<br><bandwidth-value> | rtpmap:<dynamic-PT><br><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",<br>N from 3 to 29   | "Unrestricted digital information" | N/A   | Ignore                                    | FFS     | FFS         | FFS        | FFS                              | FFS   |
| <p>Note 1 – Both PCMA and PCMU required under the conditions stated in 7.1.1.</p> <p>Note 2 - Since The usage of the CLEARMODE IETF Draft has not yet been standardized, its use is specified in RFC 4040 [29] for further study.</p> <p>Note 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.</p> |                                    |   |   |         |             |            |                                  |   |

### Clause 7.1.2

Modify clause 7.1.2 after 3<sup>rd</sup> paragraph, add table:

**Table 27A/EN XXX XXX v1.1.1 - Mapping BICC/ISUP Called Party Number to SIP Request-URI/To header field**

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

### Clause 7.1.3

Add note to Table 27 4<sup>th</sup> 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 (" <i>additional calling party number</i> ") with a complete E.164 number, with Screening Indicator = "UPVNPV", and with APRI = "presentation allowed" been received?              |   |  |   |   |
|   |   | P-Asserted-Identity header field   | From header field: display-name (optional) and addr-spec (Note 5)   | Privacy header field  |
| N   | N | Header field not included  | unavailable@hostportion   | Header field not included   |
| N (Note 4)  | Y | Header field not included  | display-name derived from Generic Number (ACgPN) if possible<br>addr-spec derived from Generic Number (ACgPN) address signals (see Table 30)<br>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 = " <i>presentation allowed</i> ", display-name may shall be derived from Calling Party Number (CgPN) if possible<br>if APRI = " <i>presentation restricted or address not available</i> " the, display-name is " <i>Anonymous</i> "-omitted<br>if APRI = " <i>presentation allowed</i> ", addr-spec is derived from Calling Party Number parameter Address Signals (see Table 30) or uses network provided value<br>if APRI = " <i>presentation restricted or address not available the</i> ", addr-spec is set to the " <i>Anonymous URI</i> " (Note 3)<br>If CgPN APRI = " <i>presentation restricted by network</i> ", addr-spec is set to " <i>unavailable@hostportion</i> " | if Calling Party Number parameter APRI = " <i>presentation restricted</i> " then priv value includes " <i>id</i> ". For other APRI settings Privacy header is not included or if included, " <i>id</i> " 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)<br>addr-spec is derived from Generic Number (ACgPN) Address Signals (see Table 28)   | if Calling Party Number parameter APRI = " <i>presentation restricted</i> " then priv value includes " <i>id</i> ". For other APRI settings Privacy header is not included or if included, " <i>id</i> " 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 (" <i>additional calling party number</i> ") with a complete E.164 number, with Screening Indicator = "UPVPNV", and with APRI = "presentation allowed" been received?   |   |                      |  |
| P-Asserted-Identity header field   | From header field: display-name (optional) and addr-spec (Note 5) | Privacy header field |  |
| 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. |   |                      |  |
| <del>NOTE 2 - Whether it is possible to derive the display-name from the Generic Number Parameter is FFS.</del>  |   |                      |  |
| NOTE 3 - 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".  |   |                      |  |
| <del>NOTE 4 - This combination of CgPN and ACgPN is an error case but is shown here to ensure consistent mapping across different implementations.</del>   |   |                      |  |
| Note 4 - 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 5: 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   |   |                      |  |

**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 | " <i>additional calling party number</i> "   | From header field       | display-name (optional) and addr-spec   |
| Nature of Address Indicator               | " <i>national (significant) number</i> "   | 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. |
|   | " <i>international number</i> "  |                         | Map complete GenericNumber Address Signals used prefixed with a "+" to user portion of URI scheme used.                             |
| Address Signals                           | if NOA is " <i>national (significant) number</i> " then the format of the address signals is:<br>NDC + SN<br>If NOA is " <i>international number</i> " then the format of the address signals is:<br>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   |

**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 <u>user portion of URI scheme used</u> . |
|                             | "international number"   |                                  | Map complete CgPN Address Signals used <u>prefixed with a "+"</u> to <u>user portion of URI scheme used</u> .                     |
| Address Signals             | If NOA is "national (significant) number" then the format of the Address Signals is:<br>NDC + SN<br>If NOA is "international number" then the format of the address signals is:<br>CC + NDC + SN | display-name (optional)          | display-name <del>may</del> <u>shall</u> be mapped from Address Signals, if <del>possible and</del> network policy allows it      |
|                             |  | addr-spec                        | "+" CC NDC SN mapped to the appropriate global number portion of URI scheme used  |

**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 <u>used</u> <u>prefixed with a "+"</u> to user portion of URI scheme used.                  |
| Address Signals             | If NOA is "national (significant) number" then the format of the Address Signals is:<br>NDC + SN<br>If NOA is "international number" then the format of the Address Signals is:<br>CC + NDC + SN | display-name (optional) | Display-name <del>may</del> <u>shall</u> be mapped from Address Signals, if <del>possible and</del> 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 1<sup>st</sup> 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~~ 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

Modify clause 7.3.1.1, 4<sup>th</sup> and 5<sup>th</sup> paragraph and Table 34:

For Profile A and B, 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.~~

#### Clause 7.7.1

Modify clause 7.7.1, last paragraph, 1<sup>st</sup> 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, 5<sup>th</sup> paragraph, 1<sup>st</sup> 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, 3<sup>rd</sup> 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, 2<sup>nd</sup> 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, 4<sup>th</sup> 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, 2<sup>nd</sup> 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, 4<sup>th</sup> paragraph, 1<sup>st</sup> 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

Modify annex B1 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 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 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 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**

Interworking of Anonymous Call Rejection (ACR) supplementary service to SIP networks

Profile A&B: Profiles A and B: The IWU shall act in accordance with the procedures described within EN 300 356-21 [21], under the sub clause "Interactions with other networks".

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

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## Annex A (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)".



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## History

| <b>Document history</b> |           |                |                                       |
|-------------------------|-----------|----------------|---------------------------------------|
| V1.1.1                  | June 2005 | Public Enquiry | PE 20051014: 2005-06-15 to 2005-10-14 |
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