ND1017:2006/07

TSG/SPEC/017

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

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TSG SPECIFICATION NUMBER 017
Interworking between Session Initiation Protocol (SIP) and UK ISDN User Part (UK ISUP)

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0.0.2 Normative Information

All enquiries about distribution, reproduction, changes and clarifications should be addressed in the first instance to the Technical Secretary of the NICC at the address on the title page.

0.0.3 Contents

0.0.4 History

Revision	Date of Issue	Updated By	Description
Issue 1.0	1st July 2006	P Wilks Editor	Approved by TSG

0.0.5 Issue Control

Section	Issue	Date
0	**Issue 1.0**	**1st July 2006**
1	**Issue 1.0**	**1st July 2006**
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3	**Issue 1.0**	**1st July 2006**
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6	**Issue 1.0**	**1st July 2006**
7	**Issue 1.0**	**1st July 2006**

NOTE: Sections that are revised in this issue are shown in this table as **bold**.

0.0.6 References

[1] ITU-T Recommendation Q.1912.5 (03/2004): "Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control Protocol or ISDN User Part"

[2] ETSI EN 302 213 V1.1.2 (2004-01) "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 (2001-07) "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 (2001-07) "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 (2001-07) "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 (2001-07) "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 (2001-07) "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 (2001-07) "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 (2001-07) "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 (2001-07) “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 (2001-07) “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 (2001-07) “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 (2001-07) “Title: 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 (2001-07) “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 (2001-07) “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 (2001-07) “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 (2001-07) “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 (2001-07) “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 (2001-07) “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 (2001-07) “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 the Session Description Protocol (SDP)

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

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

[27] IETF RFC 4040 (2005): “RTP Payload Format for a 64 kbits/s Transparent Call”

[28] IETF RFC 3966 (2004): “The tel URI for telephone numbers”.

[29] NICC ND1007:2004/12 (TSG/SPEC/007 Issue 3.2), UK ISUP

[30] NICC ND1016:2004/09 (PNO-ISC/SPEC/016 Issue 1), Requirements on Communications Providers in Relation to Customer Line Identification Display Services and Other Related Services

[31] ETSI EN 383 001 v1.1.1 2006-06 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)

[32] ND 1610 Multi-Service Interconnect of UK Next Generation Networks

[33] ND 1611 Multi-Service NGN Interconnect Common Transport

[34] ND 1612 Generic IP Connectivity for PSTN/ISDN Services between UK Next Generation Networks

[35] RFC 3455 (2003): "Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3rd-Generation Partnership Project (3GPP)"

[36] ITU-T Recommendation E.164 (02/2005) The international public telecommunication numbering plan.

[37] ITU-T Recommendation G.711 (11/88) Pulse Code Modulation (PCM) of Voice Frequencies.

[38] ITU-T Recommendation Q.939 (03/93) Digital Subscriber Signalling System No.1 (DSS1) – Typical DSS1 Service Indicator Codings for ISDN Telecommunications Services.

[39] RFC 2327 SDP: Session Description Protocol

Note: Documents with a "NICC ND" reference may be obtained from:
http://www.nicc.org.uk/nicc-public/Public/interconnectstandards/isc.htm

0.0.7 Glossary of terms

0.0.7.1 Abbreviations

3PTY	Three party
3GPP	3rd Generation Partnership Project
ACgPN	Additional Calling Party Number
ACM	Address Complete Message
ACR	Anonymous Call Rejection
ANM	Answer Message
APRI	Address Presentation Restricted Indicator
ASCII	American Standard Code for Information Interchange
AVP	Audio/Video profile
BCD	Binary Coded Decimal
BI	Beyond Interworking
BICC	Bearer Independent Call Control
CBI	CLI blocking indicator
CC	Country Code
CCBS	Completion of calls to busy subscriber
CCNR	Completion of calls on no reply
CD	Call Deflection
CFB	Call Forwarding Busy
CFNR	Call Forwarding No Repy
CFNRc	Call Forwarding on subscriber Not Reachable
CFU	Call Forwarding Unconditional
CGB	Circuit Group Blocking
CgPN	Calling party number (parameter)
CLI	Calling line identity
CLIP	Calling line identification presentation
CLIR	Calling line identification restriction

COL	Connected line identity
COLP	Connected line identification presentation
COLR	Connected line identification restriction
CONF	Conference call, add-on
CPC	Calling party's category
CS2	Capability Set 2
CUG	Closed User Group
CW	Call Waiting
DDI	Direct Dialling In
DSS1	Digital Subscriber Signalling System No.1
DTMF	Dual Tone Multi-Frequency
ECT	Explicit Call Transfer
ETSI	European Telecommunications Standards Institute
FCI	Forward Call Indicators
GN	Generic Number
GRS	Circuit Group Reset message
GVNS	Global virtual network services
HLC	High Layer Compatibility
HOLD	Call hold
IAM	Initial address message
IETF	Internet Engineering Task Force
I-IWU	Incoming (to BICC/ISUP) InterWorking Unit
INAP	Intelligent Network Application Protocol
INF	Information message
INR	Information request message
IP	Internet Protocol
ISDN	Integrated services digital network
ISUP	ISDN user part
ITCC	International Telecommunication Charge Card
ITU-T	International Telecommunication Union - Telecommunication standardisation sector
IUP	Interconnect User Part
IWU	Interworking Unit
LDLI	Last Diverting Line Identity
MCID	Malicious call identification
MIME	Multipurpose Internet Mail Extension
MLPP	Multilevel Precedence and Pre-emption
ND	NICC Document
NDC	National Destination Code
NFCI	National Forward Call Indicators
NGN	Next Generation Network
NICC	Network Interoperability Consultative Committee
NN	Network number
NOA	Nature of Address

NP	Network provided
Ofcom	Office of Communications (The Regulator for the UK Communications Industries)
O-IWU	Outgoing (from BICC/ISUP) InterWorking Unit
PECN	Public Electronic Communications Network
PN	Presentation number
PNP	Presentation number preference (indicator)
PNO-ISC	Public network operators' – Interconnect Standards Committee
PSTN	Public Switched Telephone Network
QoR	Query On Release
REL	Release
RES	Resume
Rev	Reverse charge
RFC	Request For Comments
RLC	Release Complete
RSC	Reset Circuit message
RTP	Real-time Transport Protocol
SAM	Subsequent Address Message
SCCP	Signalling Connection Control Part
SDP	Session Description Protocol
SI	Screening Indicator
SIP	Session Initiation Protocol
SIP-I	Session Initiation Protocol with encapsulated ISUP
SN	Subscriber Number
SPAN	Services and Protocols for Advanced Networking
SS7	Signalling System No.7
SUB	Sub-addressing
SUS	Suspend
TISPAN	Telecommunications and Internet converged Services and Protocols for Advanced Networking
TMR	Transmission Medium Requirement
TN	Terminal Number
TN	Transit Network
TP	Terminal Portability
TSG	Technical Steering Group
UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
UK	United Kingdom of Great Britain and Northern Ireland
UPNV	User provided, not verified
UPVP	User provided, verified and passed
URI	Uniform Resource Identifier
USI	User Service Information
USN	UK Specific Number
UUS	User-to-user signalling

0.0.7.2 Definitions

The following definitions are contained in PNO-ISC/SPEC/016 [30]:

calling line identity (CLI)
calling line identity presentation (CLIP) service
calling line identity restriction (CLIR) service
CLI available
CLI restricted/withheld
CLI unavailable
COL available
COL restricted/withheld
COL unavailable
connected line identity (COL)
network number (NN)
network provided (NP) number
presentation number (PN)
user provided, not verified (UPNV) number
user provided, verified and passed (UPVP) number

codespace
A parameter or discrete part of a parameter to which a range of values may be assigned.

codepoint
A value assigned to a parameter or discrete part of a parameter.

controlling point/network
The point/network that is responsible for the charging function associated with the call, and is thus ultimately responsible for releasing the connection.

incoming network
The network to which a call is passed from a point of interconnection between two networks.

NOTE: The incoming network may be the terminating network or a transit network.

not required
Where a service/feature is qualified in this specification as “not required” it is not necessary for either the associated underlying functionality or signalling protocols to be supported by the implementation concerned. The compatibility rules shall apply to the messages, parameters and codepoints needed to support the feature/service. These messages, parameters, and codepoints may be treated as recognised or unrecognised according to the capabilities of the recipient exchange. Implementations shall not rely on “not required” features being disabled (or enabled) at a peer entity.

Note: Interconnected or communicating implementations that provide support of the service/feature/message/parameter identified will not be considered as non-conformant to the specification.

originating network
The network to which the customer who originates a call is directly connected.

outgoing network
The network from which a call is passed to a point of interconnection between two networks.

NOTE: The outgoing network may be the originating network or a transit network.

required
Where a service/feature/message/parameter is qualified as “required” it shall be fully supported by the implementation concerned.
Note: The term may be applied independently to an interface protocol and/or the underlying functionality.

reserved
Codespace and codepoints defined in this specification as “reserved” are available for use only with the agreement of the TSG

shall not be sent/used
Where a message/parameter/codepoint is qualified by the statement “shall not be sent/used”, it is a mandatory

requirement that it shall not be sent across a point of interconnect, and that normal operation of the system shall not depend on its reception.

spare
In ITU-T recommendations, codespace and codepoints indicated as “spare” or “spare for international use” are available for future ITU-T assignment. Codespace and codepoints indicated as “spare for national use” are not available for ITU-T assignment. In TSG specifications no UK-specific codespace or codepoint shall be shown as “spare”. All unallocated UK-specific codespace and codepoints shall be defined as “reserved”. Therefore, the only application of the term “spare” in the UK specifications is with reference to ITU-T recommendations. The UK meaning of “spare” as applied to ITU-T codespace and codepoints is the same as the ITU-T meaning.

terminating network
The network to which the customer who receives a call is directly connected.

UK-specific codespace
Codespace that is specified by the TSG for use in the UK only.

UK-specific codepoint
A codepoint that has been assigned a meaning by the TSG for use in the UK only.

0.0.8 Scope

The scope of this document is described in section 0.1.

0.1 Introduction to TSG/SPEC/017

This UK Interworking specification between UK SIP and UK ISUP has been produced by the Technical Steering Group (TSG) of the Network Interoperability Consultative Committee (NICC). This specification is intended for use in the architectural environment described in the Purple release of TSG documents ND 1610 [32], 1611 [33] and 1612 [34]. This document specifies the SIP to/from UK ISUP interworking required within public electronic communications networks (PECNs) in the UK to support PSTN/ISDN services between customers. The PSTN/ISDN services supported shall be those described in TSG/SPEC/007 [29].

The TSG wishes that it shall be clearly understood that this version of the specification has been produced to support SIP-I for phase 1 of the NGN Project Plan. References to Profiles A and B are included in this issue of the Specification for information only. It is anticipated that future issues of the Specification will develop the definition of Profiles A and B for use in the UK.

Further it shall be understood that SIP-I is required to support the full set of PSTN/ISDN services. Hence a route that supports SIP-I shall be selected if the call contains any messages, parameters or code points which can only be supported by the encapsulation of UK ISUP protocol.

"In the UK" is defined as the UK network up to and including the National component of an International Switching Centre. The actions at an International Gateway are beyond the scope of this document. Text describing any such actions is included for information only.

This issue of the specification contains the functionality of the ITU-T Rec. Q.1912.5 [1] Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control or ISDN User Part modified to include changes specified by ETSI EN 383 001 [31], and to include additional functionality which is required for the UK. This document does not specify the functionality required to support the interworking of BICC to SIP. The references to BICC in ITU-T Q.1912.5 [1] and ETSI EN 383 001 [31] shall be ignored.

This specification is written as exceptions to ITU-T and ETSI documents. This is done by listing each paragraph of the ITU-T recommendation to be modified, identifying the exceptions in ETSI and the additional UK requirements.

Items for which no table row has been included are required with the exception of items in ITU-T recommendations or ETSI specifications marked as "national use" or "network option", which are not required.

Comments have been added against specific paragraphs and qualified as either:

UK: = additional UK specific national requirements, and errors or clarifications to the reference documents.

E: = ETSI exceptions to the ITU-T recommendations.

0.2 Index of TSG/SPEC/017

Section 1 Q.1912.5 Main Text
Exceptions to ITU-T recommendation Q.1912.5 [1] as modified by EN 383 001 [31].

Section 2 Q.1912.5 Annex A BICC specific interworking for basic call
Exceptions to ITU-T recommendation Q.1912.5 [1] Annex A as modified by EN 383 001 [31].
Note: UK not required

Section 3 Q.1912.5 Annex B Interworking for ISDN supplementary services
Exceptions to ITU-T recommendation Q.1912.5 [1] Annex B as modified by EN 383 001 [31].

Section 4 Q.1912.5 Annex C
Exceptions to ITU-T recommendation Q.1912.5 [1] Annex C as modified by EN 383 001 [31].

Section 5 Q.1912.5 Appendix I Interworking scenarios between SIP and BICC
Exceptions to ITU-T recommendation Q.1912.5 [1] Appendix 1 as modified by EN 383 001 [31].
Note: UK not required

Section 6 Q.1912.5 Appendix II Interworking scenarios between SIP and ISUP
Exceptions to ITU-T recommendation Q.1912.5 [1] Appendix 2 as modified by EN 383 001 [31].

Section 7 Q.1912.5 Appendix III Interworking scenarios between Profile C (SIP-I) and ISUP
Exceptions to ITU-T recommendation Q.1912.5 [1] Appendix 3 as modified by EN 383 001 [31].

END OF TSG/SPEC/017§0

1 Q.1912.5 Main Text

1.1 Exceptions

Global modifications to ITU-T Recommendation Q.1912.5

E: Throughout the text of ITU-T Recommendation Q.1912.5
Replace references as shown below.

Reference in ITU-T Recommendation Q.1912.5	Modified reference
ITU-T Recommendation Q.731.3	ITU-T Recommendation Q.731.3 as modified by EN 300 356-3
ITU-T Recommendation Q.731.4	ITU-T Recommendation Q.731.4 as modified by EN 300 356-4
ITU-T Recommendation Q.731.5	ITU-T Recommendation Q.731.5 as modified by EN 300 356-5
ITU-T Recommendation Q.731.6	ITU-T Recommendation Q.731.6 as modified by EN 300 356-6
ITU-T Recommendation Q.731.7	ITU-T Recommendation Q.731.7 as modified by EN 300 356-11
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ITU-T Recommendation Q.732.2	ITU-T Recommendation Q.732.2 as modified by EN 300 356-15
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ITU-T Recommendation Q.733.1	ITU-T Recommendation Q.733.1 as modified by EN 300 356-17
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ITU-T Recommendation Q.733.4	ITU-T Recommendation Q.733.4 as modified by EN 300 356-7
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ITU-T Recommendation Q.734.1	ITU-T Recommendation Q.734.1 as modified by EN 300 356-12
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ITU-T Recommendation Q.735.1	ITU-T Recommendation Q.735.1 as modified by EN 300 356-9
ITU-T Recommendation Q.737.1	ITU-T Recommendation Q.737.1 as modified by EN 300 356-8
ITU-T Recommendation Q.761	ITU-T Recommendation Q.761 as modified by EN 300 356-1
ITU-T Recommendation Q.762	ITU-T Recommendation Q.762 as modified by EN 300 356-1
ITU-T Recommendation Q.763	ITU-T Recommendation Q.763 as modified by EN 300 356-1
ITU-T Recommendation Q.764	ITU-T Recommendation Q.764 as modified by EN 300 356-1
ITU-T Recommendation Q.850	ITU-T Recommendation Q.850 as modified by EN 300 485
ITU-T Recommendation Q.1902.1	ITU-T Recommendation Q.1902.1 as modified by EN 302 213
ITU-T Recommendation Q.1902.2	ITU-T Recommendation Q.1902.2 as modified by EN 302 213
ITU-T Recommendation Q.1902.3	ITU-T Recommendation Q.1902.3 as modified by EN 302 213
ITU-T Recommendation Q.1902.4	ITU-T Recommendation Q.1902.4 as modified by EN 302 213
ITU-T Recommendation Q.1912.5]	ITU-T Recommendation Q.1912.5 as modified by the present document
IETF RFC 2806	IETF RFC 3966 NOTE: RFC 2806 is obsolete. RFC 3966 replaces RFC 2806

Q.1912.5 Paragraph	**Title**	**Comment**
General		**E**: Throughout the present document "should" is replaced by "shall". **UK**: Throughout this document references to ITU-T Q.761-4 shall be replaced with TSG/SPEC/007 [29]
1	**Scope**	**E**: 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
5.3.3	**Interworking of ISUP overlap signalling**	**E**: 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.~~
5.4.1.2	**Header fields for ISUP MIME bodies**	**UK:** Replace the ITU-T text with the following "The Content-Type header field associated with the ISUP MIME body shall be supplied as follows: Content-Type: application/ISUP; version = X-UKISUP and if a base value is included it shall be itu-t92+. The Content-Disposition header field associated with the ISUP MIME body shall be set as follows: Content-Disposition: signal; handling = required.
5.4.3	**Exclusions and special considerations**	**UK:** Replace with the text and table shown below: Table 1 lists all of the messages supported in UK ISUP. The table indicates whether or not they are capable of being encapsulated. The table also provides a reference to either the Q.1912.5 sub-section (§) or other place that describes how the message is handled. This table applies not only to messages received on the BICC/ISUP side and interworked but also to messages generated internally.

Table 1

Message type	O-IWU	I-IWU	Encapsulation Possible	Comments
Address complete	§ 7.3	§ 6.5	Y	
Answer	§ 7.5	§ 6.7	Y	
Application transport	§ 5.4.3.2	§ 5.4.3.2	Y	
Blocking	§ 5.4.3.1	§ 5.4.3.1	N	
Blocking acknowledgement	§ 5.4.3.1	§ 5.4.3.1	N	
Call progress	§ 7.3.1	§ 6.6	Y	See also Annex B.10
Circuit group blocking	§ 7.7.4	§ 6.11.4	N	
Circuit group blocking acknowledgement	§ 5.4.3.1	§ 5.4.3.1	N	
Circuit group reset	§7.7.4	§ 6.11.4	N	
Circuit group reset acknowledgement	§ 5.4.3.1	§ 5.4.3.1	N	
Circuit group unblocking	§ 5.4.3.1	§ 5.4.3.1	N	
Circuit group unblocking acknowledgement	§ 5.4.3.1	§ 5.4.3.1	N	
Confusion	§ 5.4.3.1 or 2	§ 5.4.3.1 or 2	Y/N	Note 2
Connect	§ 7.5	§ 6.4	Y	
Facility	§ 5.4.3.2	§ 5.4.3.2	Y	
Facility accepted	§ 5.4.3.2	§ 5.4.3.2	Y	
Facility reject	§ 5.4.3.1 or 2	§ 5.4.3.1 or 2	Y/N	Note 2
Facility request	§ 5.4.3.2	§ 5.4.3.2	Y	
Identification request	§ 5.4.3.2	§ 5.4.3.2	Y	
Identification response	§ 5.4.3.2	§ 5.4.3.2	Y	
Information	§ 7.w.w	§ 6.z.z	Y	
Information request	§ 7.w.w	§ 6.z.z	Y	
Initial address	§ 7.1	§ 6.1	Y	
Loop prevention	Annex B.8	Annex B.8	Y	
Pre-release information	§ 5.4.3.2	§ 5.4.3.2	Y	
Release	§ 7.7	§ 6.11	Y	
Release complete	§ 5.4.3.4.	§ 5.4.3.4	Y	
Reset circuit	§ 7.7.4	§ 6.11.4	N	Note 1
Resume	§ 7.x.x	§ 6.10	Y	See also Annex B.13
Segmentation	§ 5.4.3.3	§ 5.4.3.3	N	Normal ISUP procedures apply
Subsequent address	§ 7.2	§ 6.2	N	Digits carried in subsequent INVITE but SAM not carried
Suspend	§ 7.x.x	§ 6.9	Y	See also Annex B.13
Unblocking	§ 5.4.3.1	§ 5.4.3.1	N	

Message type	O-IWU	I-IWU	Encapsulation Possible	Comments
Unblocking acknowledgement	§ 5.4.3.1	§ 5.4.3.1	N	
User Part available	§ 5.4.3.1	§ 5.4.3.1	N	
User Part test	§ 5.4.3.1	§ 5.4.3.1	N	
User-to-user information	§ 5.4.3.2	§ 5.4.3.2	Y	See also Annex B.21
NOTE 1 – Where the ISUP procedures would send reset circuit (RSC) to an ISUP exchange, the IWU shall send an encapsulated REL with release cause 31 (Normal, unspecified). NOTE 2 – These messages are either locally terminated or sent transparently depending on whether they are destined for the IWU or for another exchange.				

Q.1912.5 Paragraph	**Title**	**Comment**
6.	**Incoming call interworking from SIP to BICC/ISUP at I-IWU**	**UK:** Add the following text at the end of the section Within the UK a route shall be marked to indicate which profiles are supported. The following text describes the actions that a network may take on receiving an INVITE. a) If a call is received on a route marked as supporting SIP-I and the INVITE does contain an encapsulated ISUP IAM then the call shall be accepted and processed based on the SIP-I profile. b) If a call is received on a route marked as supporting SIP-I and the INVITE does **not** contain an encapsulated ISUP IAM then it shall be a network operator's option to either: i) reject the call or ii) to have agreed with the outgoing network on how the call shall be handled. (This type of call is not a PSTN/ISDN call and falls outside of the scope of TSG/SPEC/017 Phase 1). c) If a call is received on a route marked as supporting SIP without ISUP encapsulation and the INVITE does contain an encapsulated ISUP IAM then how the call is handled is dependent on the setting of the Content-Disposition header field. If the Content-Disposition is set to "Content-Disposition: signal;handling = required" then the call shall be rejected. If the Content-Disposition is set to "Content-Disposition: signal;handling = optional" then it shall be a network operator's option to either: i) reject the call or ii) to have agreed with the outgoing network on how the call shall be handled. i.e. the call shall be handled as though no encapsulated ISUP was present. (This type of call is not a PSTN/ISDN call and falls outside of the scope of TSG/SPEC/017 Phase 1). d) If a call is received on a route marked as supporting SIP without ISUP encapsulation and the INVITE does not contain an encapsulated ISUP IAM then the call shall be accepted and processed according to the agreement on how such calls shall be handled. (This type of call is not a PSTN/ISDN call and falls outside of the scope of TSG/SPEC/017 Phase 1).
6.1	**Sending of Initial Address Message (IAM)**	**UK:** Delete the existing text in paragraphs 6.1.1 & 6.1.2 and replace with the following: **6.1.1 INVITE received without an SDP offer** If the INVITE does not contain a SDP offer then the call shall be rejected by returning a "488 Not Acceptable Here" response. **6.1.2 INVITE received with an SDP offer** If the INVITE does contain a SDP offer the I-IWU shall immediately send out the IAM. Section 6.1.3 gives specific details related to the population of specific parameters of the IAM. Table 2 gives a summary of parameters within the IAM that are interworked from the INVITE along with a reference to the subclauses of 6.1.3 in which the specific interworking is described.

<table>
<tr><th>Q.1912.5 Paragraph</th><th>Title</th><th>Comment</th></tr>
<tr><td>6.1.3.1</td><td>Called Party Number</td><td>
E: Modify Table 3:
Table 3/Q.1912.5 – Coding of the Called Party Number
<table>
<tr><th>INVITE→</th><th>IAM→</th></tr>
<tr><td>Request-URI</td><td>Called Party Number</td></tr>
<tr><td rowspan="4"></td><td>Odd/even indicator: set as required</td></tr>
<tr><td>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 0000011 “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 0000100 “International number”, remove “+” and use the remaining digits to fill the Address signals.</td></tr>
<tr><td>Internal Network Number Indicator:
1 routing to internal network number not allowed</td></tr>
<tr><td>Numbering plan Indicator:
001 ISDN (Telephony) numbering plan (Rec. E.164)</td></tr>
<tr><td>Userinfo
(sip: URI with user=phone)</td><td>Address Signals</td></tr>
<tr><td colspan="2">Note: RFC3966 [28] describes the tel format of a userportion.</td></tr>
</table>

UK: UK Specific Address
To the above ETSI amendment add under Nature of address indicator immediately after the text
“Analyse the information contained in received URI with user=phone, and if it is in the format:-“
USA (UK Specific Address), then set the Nature of Address indicator to “111 1110 (126) UK Specific Address” and use all of the digits to fill the ISUP Called Party Number Address signals.
The Request-URI shall be identified as containing a USA if its userinfo component contains a series of digits commencing with a digit which is not ‘0’ and is followed by a context with a descriptor containing the global-number-digits”+44”.
If the userinfo component contains a series of digits commencing with “0” and is followed by a context with a descriptor containing the global-number-digits “+44”, the I-IWU shall fail the call by returning a 404 (not found).
</td></tr>
</table>

<table>
<tr><th>Q.1912.5 Paragraph</th><th>Title</th><th>Comment</th></tr>
<tr><td>6.1.3.2</td><td>Calling Party's Category (mandatory)</td><td>
UK: The following change is an interim method of setting priority for emergency calls. Delete the first paragraph and table, and replace with:
~~For Profiles A and B, the following codes should be set by the I-IWU as default in the Calling Party's Category parameter.~~
<table>
<tr><th>~~Bits/Codes~~</th><th>~~Meaning~~</th></tr>
<tr><td>~~0000 1010~~</td><td>~~"Ordinary calling subscriber"~~</td></tr>
</table>
For Profiles A and B, the following codes should be set by the I-IWU as default in the Calling Party's Category parameter as determined by analysis of the digits contained in the Request URI.
If analysis determines that the Nature of Address indicator contained in the Called Party Number parameter shall be set to “111 1110, (126) UK Specific Number” and that the Address signals shall be such that the call requires priority (e.g. “999”, “112”, “998” or “18000”) then the CPC shall be set to “0000 1011 Calling subscriber with priority”, otherwise set to “0000 1010 Ordinary calling subscriber”.
The definitive list of address signals requiring priority is outside the scope of this document.
</td></tr>
<tr><td>6.1.3.3</td><td>Nature of Connection Indicators (mandatory)</td><td>
E: Modify 2nd paragraph:
~~Other fields in the Nature of Connection Indicators should follow the current BICC/ISUP Recommendation.~~
</td></tr>
<tr><td>6.1.3.4</td><td>Forward Call Indicators</td><td>
E: 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~~
<table>
<tr><th>~~Bits~~</th><th>~~Codes~~</th><th>~~Meaning~~</th></tr>
<tr><td>~~D~~</td><td>~~1~~</td><td>~~“Interworking encountered”.~~</td></tr>
<tr><td>~~F~~</td><td>~~0~~</td><td>~~“ISDN user part/BICC not used all the way”.~~</td></tr>
<tr><td>~~HG~~</td><td>~~01~~</td><td>~~“ISDN user part/BICC not required all the way”~~</td></tr>
<tr><td>~~I~~</td><td>~~0~~</td><td>~~“Originating access non-ISDN”~~</td></tr>
</table>
~~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.~~“
</td></tr>
</table>

Q.1912.5 Paragraph	**Title**	**Comment**
6.1.3.4		Forward Call Indicators bit A National/International call indicator 0 Call to be treated as a national call 1 Call to be treated as a international call bits CB End to end method indicator 00 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 will allow the DSS1 protocol at the S/T interface to avoid sending 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 will allow the DSS1 protocol at the S/T interface to avoid sending 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 01 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 will allow the DSS1 protocol at the S/T interface to avoid sending 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 00 no indication **UK:** Delete the ETSI text and replace with UK text as shown: **ETSI text** ~~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 will allow the DSS1 protocol at the S/T interface to avoid sending 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.~~ **UK text** Bit D Interworking indicator 0 no interworking encountered (SS7 used all of the way)

Q.1912.5 Paragraph	**Title**	**Comment**
6.1.3.5	**Transmission Medium Requirement (mandatory), User Service Information (optional), and Higher Layer Compatibility information element within Access Transport Parameter (optional)**	**E:** Modify title 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 µ-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 µ-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. **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 RFC3264 [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 RFC3264 [24] **UK:** Add paragraph after 6.1.3.5.4 "Table 6 provides the default mapping relations based on the above procedure."

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

UK: Replace Table 6/Q.1912.5 – Coding of TMR/USI/HLC from SDP: SIP to BICC/ISUP with that shown below.

SIP Input						**ISUP Output**			
m = line			**b = line (Note 3)**	**a = line**		**TMR parameter**	**USI parameter**		**HLC parameter**
<media>	<transport>	<fmt-list>	<modifier>: <bandwidth-value> Note: <bandwidth-value> for <modifier> of AS is evaluated to be B kbit/s.	Rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/<encoding parameters>]	ptime: <packet time>	TMR codes	Information Transport Capability	User Information Layer 1 Protocol Indicator	Higher Layer Characteristics Identification
Audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A	ptime:10	*"3.1kHz audio"*	*"3.1kHz audio"*	*"G.711 A law"*	(Note 2)
Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMA/8000	ptime:10	*"3.1kHz audio"*	*"3.1kHz audio"*	*"G.711 A law"*	(Note 2)
Audio	RTP/AVP	Dynamic PT	AS: 64 kbit/s	rtpmap:<dynamic-PT> CLEARMODE/8000 (Note 1)	ptime:10	*"64 kbit/s unrestricted "*	*"Unrestricted digital information"*		

Note 1 – CLEARMODE is specified in RFC 4040 [27]

Note 2 – 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 3 – The b = line is optional in RFC 2327 [39] and therefore might not be present.

Q.1912.5 Paragraph	**Title**	**Comment**
6.1.3.6	**BICC/ISUP Calling Line Identification (CLI) parameters**	**UK:** Modify the last sentence of the first paragraph: Finally, Table 10 provides details for mapping to Generic Number & Presentation Number parameters when this is possible. **E:** 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. **UK:** Replace the 3rd paragraph and the first additional paragraph of the above ETSI amendment with the following: If the encapsulated IAM contains any CLI parameters (i.e. Calling Party Number, Generic Number or Presentation Number) then these parameters shall be copied unchanged into the ISUP IAM sent by the IWU. If all of these CLI parameters are absent from the encapsulated IAM then they shall be produced (if possible) by mapping from SIP headers as described in Tables 7A, 7B, 8, 9, & 10. **E:** Modify 4th Paragraph If ~~Should~~ any discrepancy occurs in privacy settings during the alignment process the strongest privacy setting shall prevail. **E:** Modify Table 7 **UK:** Replace the modified ETSI Table 7 with Table 7A and 7B below: **E:** Modify Table 8 **UK:** Replace the modified ETSI Table 8 with Table 8 below: **E:** 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*". **UK:** Table 9 replace the modified ETSI Note 1 with the following: “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~~. The sip: URI shall take precedence, unless the sip: URI is received without “user=phone”, in which case the tel: URI shall take precedence.”

Q.1912.5 Paragraph	**Title**	**Comment**
		E: Modify Table 10. **UK:** Replace title of section 6.1.3.6.2 with "Generic Number and Presentation Number". Replace the modified ETSI Table 10 with Table 10 below:

Table 7A/Q.1912.5 – Derivation of the ISUP Calling Party Number parameter and CLI Blocking Indicator (CBI)

Has a SIP P-Asserted-Identity been received, containing a URI (Note 1) with an identity in the format "+" CC + NDC + SN?			
	Calling Party Number parameter		**NFCI parameter**
	Address Signals	**APRI**	**CBI**
No	Include a network provided E.164 number from a number range allocated to that network operator (see Table 8).	Set APRI to "*presentation restricted by network*".	Set CBI to "*network number may not be disclosed to the called user*" (0).
Yes	Derive from SIP P-Asserted-Identity (See Table 9)	APRI = *"presentation restricted*" or "*presentation allowed*" depending on SIP Privacy header. (See Table 9)	Set CBI to "*Network Number may (subject to interaction with CLIR) be disclosed to the called user*" (1)
NOTE 1 – It is possible that the P-Asserted-Identity header field includes both a tel: URI and a sip: URI. The sip: URI shall take precedence, unless the sip: URI is received without "user=phone", in which case the tel: URI shall take precedence.			

Table 7B/Q.1912.5 – Derivation of the ISUP Generic Number (Additional Calling Party Number) parameter and Presentation Number parameter

Has a SIP From header been received containing a URI with an identity in the format "+" CC + NDC + SN ?		
	Generic Number ("*Additional calling party number*") & Presentation Number parameters	
	Address Signals	**APRI**
No	Parameters not included	Not applicable
Yes	Omit the parameters or derive from the SIP From header if it is known that this header was originated by a trusted network. (see Table 10) (Note 1)	See Table 10
NOTE 1 – This mapping effectively gives the equivalent of Special Arrangement to all SIP UAC with access to the I-IWU. Therefore, unless it is known that the SIP From header was originated by a trusted network, it is not permitted within the UK to use the From header to derive these parameters.		

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"* (only complete E.164 numbers shall be inserted)
Numbering Plan Indicator	*"ISDN/Telephony (E.164)"*
Address Presentation Restricted Indicator	*"Presentation ~~allowed/~~restricted by 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 ~~should~~ be included. If NOA is "*international number",* then the country code of the network-provided number shall ~~should~~ be included.

Table 10/Q.1912.5 – Mapping of SIP From header field to BICC/ISUP Generic Number (*"additional calling party number"*) & Presentation Number parameters ~~(network option)~~ (when permitted – see Table 7B Note 1)

Source SIP header field and component	Source component value	Generic Number / Presentation Number parameter field	Derived value of parameter field
–	–	Number Qualifier Indicator (GN only)	"*additional calling party number*"
-	-	Presentation Number Preference Indicator (PN only)	*"PN preferred for mapping to legacy (IUP) ISDN services"*
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 unless Calling party number APRI = "presentation restricted by network" then set the GN & PN APRI's to "presentation allowed".
–	–	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

Q.1912.5 Paragraph	**Title**	**Comment**
6.1.3.7	**User Service Information (Optional)**	**E:** Modify title 6.1.3.7 **"User Service Information ~~(Optional)~~ "**
6.1.3.9	**Hop Counter (Optional)**	**E:** Modify title 6.1.3.9 **"Hop Counter ~~(Optional)~~ "** **UK:** Add the following text to the end of this subsection The factor used in the calculation shall equal '1'. If after processing the Max Forward value utilising the algorithm with a factor of 1 the resultant Hop Counter value is greater than or equal to 31 then a value of 30 shall be inserted in the Hop Counter value field.
6.7	**Receipt of Answer Message (ANM)**	**UK: Modify text as shown** The mapping of ANM is shown in Table 15. On receipt of BICC/ISUP ANM, the I-IWU shall indicate to the SIP protocol to send a 200 OK INVITE to the UAC. ~~If no offer was received in the initial INVITE, and reliable provisional responses were not supported, the 200 OK INVITE shall include an SDP offer consistent with the TMR/USI used on the BICC/ISUP side.~~
6.9	**Receipt of Suspend message (SUS) network initiated**	**UK:** Modify paragraph 1 as shown If the I-IWU is the controlling exchange for the Suspend/Resume procedure, the actions taken on the BICC/ISUP side upon receipt of the Suspend message (SUS) are described in ~~2.4.1.c/Q.764 and 10.2.1.c/Q.1902.4~~TSG/SPEC/007 Section 4.2.2.1.
6.10	**Receipt of Resume message (RES) network initiated**	**UK:** Modify paragraph 1 as shown If the I-IWU is the controlling exchange for the Suspend/Resume procedure, the actions taken on the BICC/ISUP side upon receipt of the Resume message (RES) are described in ~~2.4.2.c/Q.764 and 10.2.2.c/Q.1902.4~~TSG/SPEC/007 Section 4.2.2.1.
6.11.1	**Receipt of BYE/CANCEL**	**E:** 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~~ shall be mapped to the ISUP Cause Value field in the ISUP REL ~~depending on local policy~~." Modify clause 6.11.1 5th paragraph, 2nd sentence. "The mapping of the ~~Cause Indicators~~Reason Header ~~parameter~~ to the ~~Reason header~~Cause Indicators parameter is shown in Table 18." **UK:** Modify clause 6.11.1 5th paragraph 4th sentence: In both cases, the Location Field shall be set to "~~*network beyond interworking point*~~ *user*".

Q.1912.5 Paragraph	**Title**	**Comment**
6.11.2	**Receipt of REL**	**E:** 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 re-selection,~~ When the internal bearer path has been disconnected then an ISUP RLC is returned to the ISUP side." **UK:** Delete ETSI modification **E:** Modify clause 6.11.2, 4th paragraph, 1st 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." **UK:** Delete ETSI note under clause 6.11.2, 4th paragraph, 1st sentence **E:** Modify Table 20 last sentence: "Note 2 - Due to the fact that the Cause Indicators parameter does not include the definition text as defined in Table1/Q.850 this is based on provisioning in the ~~O~~ I-IWU" **UK:** Replace Table 21 with the following:

Table 21/Q.1912.5 – Receipt of Release message (REL)

ISUP REL Cause Value	SIP Response Code
1 Unallocated (unassigned) number	404 Not Found
2 No route to specified transit network (national use)	404 Not Found
3 No route to destination	404 Not Found
4 Send special information tone	604 Does Not Exist Anywhere
5 Misdialled trunk prefix (national use)	404 Not Found
8 Preemption	480 Temporarily Unavailable
9 Preemption - circuit reserved for reuse	480 Temporarily Unavailable
14 QoR: ported number	410 Gone
16 Normal call clearing	480 Temporarily Unavailable
17 User busy	600 Busy Everywhere
18 No user responding	408 Request Timeout
19 No answer from user (user alerted)	480 Temporarily Unavailable
20 Subscriber absent	480 Temporarily Unavailable
21 Call rejected	603 Decline
22 Number changed	410 Gone
23 Redirection to new destination	302 Moved Temporarily
24 Call rejected due to ACR supplementary service	433 Anonymity Disallowed
25 Exchange Routing Error	483 Too Many Hops
27 Destination out of order	480 Temporarily Unavailable
28 Invalid number format (address incomplete)	484 Address Incomplete
29 Facility rejected	403 Forbidden
31 Normal unspecified	480 Temporarily Unavailable
34 No circuit/channel available.	486 Busy Here/ 600 Busy Everywhere. (Determined by Location value. “User “ = 600, others = 486)
38 Network out of order	500 Server Internal Error
41 Temporary failure	500 Server Internal Error
42 Switching equipment congestion	503 Service Unavailable
43 Access information discarded	500 Server Internal Error
44 Requested circuit/channel not available	500 Server Internal Error
46 Precedence call blocked	500 Server Internal Error
47 Resource unavailable, unspecified	500 Server Internal Error
50 Requested Facility Not Subscribed	403 Forbidden
53 Outgoing calls barred within CUG	403 Forbidden
55 Incoming calls barred within CUG	403 Forbidden
57 Bearer capability not authorized	488 Not Acceptable Here
58 Bearer capability not presently available	403 Forbidden
62 Inconsistency in designated outgoing access information and subscriber class	403 Forbidden
63 Service or option not available, unspecified	403 Forbidden

ISUP REL Cause Value	SIP Response Code
65 Bearer capability not implemented	501 Not Implemented
69 Requested facility not implemented	501 Not Implemented
70 Only restricted digital information bearer capability is available	488 Not Acceptable Here
79 Service or option not implemented, unspecified	501 Not Implemented
87 User not member of CUG	403 Forbidden
88 Incompatible destination	488 Not Acceptable Here
90 Non-existent CUG	404 Not Found
91 Invalid transit network selection	404 Not Found
95 Invalid message, unspecified	502 Bad Gateway
97 Message type non-existent or not implemented	502 Bad Gateway
99 Information element / parameter non-existent or not implemented	502 Bad Gateway
102 Recovery on timer expiry	504 Server Time-out
103 Parameter non-existent or not implemented, passed on	502 Bad Gateway
110 Message with unrecognized parameter, discarded	502 Bad Gateway
111 Protocol error, unspecified	502 Bad Gateway
127 Interworking, unspecified	502 Bad Gateway

Q.1912.5 Paragraph	Title	Comment
6.11.3	**Autonomous release at I-IWU**	**E:** Modify clause 6.11.3, 4th paragraph, last 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." **UK:** Delete ETSI note under clause 6.11.3, 4th paragraph, 2nd sentence
6.11.4	**Receipt of RSC, GRS or CGB (ISUP)**	**E:** 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 where the Range subfield of the Range and Status Parameter contains a value equal to or greater than "1" multiple SIP messages will be sent on receipt of a single GRS or CGB message. **UK**: Delete ETSI modification
6.11.5	**Receipt of RSC or GRS (BICC)**	**E:** 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 where the Range subfield of the Range and Status Parameter contains a value equal to or greater than "1" multiple SIP messages will be sent on receipt of a single GRS message. **UK**: Delete ETSI modification

Q.1912.5 Paragraph	**Title**	**Comment**
7.	**Outgoing call interworking from BICC/ISUP to SIP at O-IWU**	**UK:** Add The following text at the end of the section Within in the UK a route shall be marked to indicate which profiles are supported. The following text describes the actions which a network may take: a) If a call is to be originated on a route which is marked as supporting SIP-I then the INVITE shall contain an encapsulated IAM. If SIP messages which can contain encapsulated ISUP are received in the backward direction which do contain encapsulated ISUP information then they shall be accepted and processed based on the SIP-I profile. If SIP messages which are capable of containing encapsulated ISUP are received in the backwards direction but do not contain encapsulated ISUP information then it shall be a network operator’s option to determine how the call shall be handled. The network operator shall have the following options available: i) accept and process the messages, this option is only valid if the setting of the Content-Disposition header field in the SIP INVITE was NOT set to “Content-Disposition: signal;handling = required” (This type of call is not a PSTN/ISDN call and falls outside of the scope of TSG/SPEC/017 Phase 1); ii) fail the call due to a protocol error; iii) originate another call attempt with an INVITE which does not contain encapsulated information. i.e. the call proceeds as in (b) below with the outgoing network assuming responsibility for the interworking. (This type of call is not a PSTN/ISDN call and falls outside of the scope of TSG/SPEC/017 Phase 1). b) If a call is to be originated on a route which is marked as not supporting SIP-I then the INVITE shall not contain an encapsulated IAM. This type of call shall be handled according to the agreement that exists between the networks. (This type of call is not a PSTN/ISDN call and falls outside of the scope of TSG/SPEC/017 Phase 1).
7.1	**Scenario A**	**UK:** At end of 2nd paragraph add “and if there is sufficient bandwidth available over the bearer path to support the service being requested. If there is insufficient bandwidth available to support the service being requested then the call shall be failed by Release with Cause Value 34. The SIP INVITE shall contain a SDP offer.
	Scenario B	**UK:** Preconditions shall not be used at a point of interconnect.
7.1.1.	**Coding of SDP media description lines from TMR/USI**	**E:** 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.~~“

7.1.1.1 **Transcoding not available at the O-IWU**

E: Modify clause 7.1.1.1 Table 26

UK: Replace the ETSI table with the table below

ISUP Input	**SIP Output**					
TMR parameter	**m= line**			**b= line (Note 2)**	**a= line**	
TMR codes	**<media>**	**<transport>**	**<fmt-list>**	**<modifier>:<bandwidth-value>**	**rtpmap:<dynamic-PT> <encoding name> /<clock rate>[/<encoding parameters>]**	**ptime: <packet time>**
“speech”	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000	ptime:10
“3.1 KHz audio”	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000	ptime:10
“64 kbit/s unrestricted”	audio	RTP/AVP	Dynamic PT	AS:64	rtpmap:<dynamic-PT> CLEARMODE/8000 (Note 1)	ptime:10

Note 1 – CLEARMODE is specified in RFC 4040 [27].

Note 2 – The b=line is optional in RFC 2327 [39] and may be omitted.

<table>
<tr><th>Q.1912.5 Paragraph</th><th>Title</th><th>Comment</th></tr>
<tr><td>7.1.2</td><td>Request-URI and To header field</td><td>
E: 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
<table>
<tr><th>IAM</th><th>INVITE</th></tr>
<tr><td>Called Party Number
Nature of address indicator:
National (significant) number

International number</td><td>Request-URI/To header field

Insert "+CC" before the Address signals
Note: CC = Country Code of the network in which the O-IWU is located.
Insert "+" before the Address signals</td></tr>
</table>
UK: Modify above table.
<table>
<tr><th>IAM</th><th>INVITE</th></tr>
<tr><td>Called Party Number parameter
b) Nature of address indicator:
000 0011 (3) National (significant) number

000 0100 (4) International number

111 1110 (126) UK Specific Address</td><td>Request-URI/To header field

Insert "+CC" before the Address signals and copy to userinfo component of both URI's
Note: CC = Country Code of the network in which the O-IWU is located.
Insert "+" before the Address signals and copy to userinfo component of both URI's
Copy all address signals followed by a context with a descriptor containing the global-number-digits " +44" to the userinfo component of both URI's.</td></tr>
</table>
</td></tr>
<tr><td>7.1.3</td><td>P-Asserted-Identity, From and Privacy header fields</td><td>
UK: Replace the first paragraph as follows:
Table 27B shows the mapping of the ISUP CgPN parameter to SIP P-Asserted-Identity and Privacy header fields. Table 27C shows the mapping of ISUP PN, GN or CgPN parameters to SIP From header field. Table 28 shows the mapping of the ISUP CgPN, GN and PN parameter fields to SIP header fields. Table 31 provides details for mapping from the APRI subfield<s>s</s> of the Calling Party Number parameter into the Privacy header field.

E: Modify Tables 27, 28, 29 and 30.

UK: Replace the modified ETSI Tables 27 & 28 with the following UK Tables 27B, 27C and 28. Delete Tables 29 and 30, and modify Table 31.
</td></tr>
</table>

Table 27B/Q.1912.5 – Mapping BICC/ISUP Calling Party Number parameter to SIP P-Asserted-Identity and Privacy header fields

Has a NFCI parameter been received with CBI = 0?	Has a Calling Party Number parameter been received with complete E.164 number, with SI = UPVP or NP (See Note 1) and with APRI = "presentation allowed" or "presentation restricted"?	P-Asserted-Identity header field	Privacy header field
N	N	Header field not included	Header field not included
	Y (Note 1)	Map the Calling Party Number parameter to the P-Asserted-Identity header as described in Table 28	See Table 31
Y	Y or N	Header field not included	Header field not included
NOTE 1 – A Network Provided CLI in the CgPN parameter may occur on a call from any type of access line. This type of CLI is 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.			

Table 27C/Q.1912.5 – Mapping BICC/ISUP Presentation Number, Generic Number or Calling Party Number parameters to SIP From header field

Has a Presentation Number parameter been received, with a complete E.164 number, with PNP = “PN preferred for mapping to legacy (IUP) ISDN services” and with APRI = “presentation allowed”?	Has, a Generic Number parameter been received with Number Qualifier "*additional calling party number*", with a complete E.164 number, with SI = "UPNV", and with APRI = "presentation allowed"?	Has a NFCI parameter been received with CBI = 0?	Has a Calling Party Number parameter with complete E.164 number been received, and with SI = UPVP or NP?	CgPN APRI	From header field: display-name (optional) and addr-spec
Y	Y or N	Y or N	Y or N	-	Map the Presentation Number parameter to the From header field as described in Table 28.
N	Y	Y or N	Y or N (If N see Note 2)	-	Map the Generic Number parameter to the From header field as described in Table 28.
	N	Y	Y or N	-	unavailable@hostportion
		N	N	-	unavailable@hostportion
			Y	Presentation allowed	Map the CgPN parameter to the From header field as described in Table 28.
				Presentation restricted / Not available (Note 3)	The display-name is omitted and the addr-spec is set to the "Anonymous URI" (Note 1).
				Restricted by Network	unavailable@hostportion

NOTE 1 – 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 2 – The combination where there is a GN(AcgPN) but no CgPN is an error case but the mapping shown here shall apply to ensure consistency across different implementations.

NOTE 3 – Note that the APRI value “Not Available” is shown. However this value is defined for national use in Q.763 and should not normally be received.

Table 28/Q.1912.5 – Mapping of ISUP Line Identity parameter fields to SIP header fields

BICC/ISUP parameter field		SIP header field	
Nature of Address Indicator	**Address Signals**	**Addr-spec**	**Display-name (Note 1)**
"national (significant) number"	NDC + SN	Add "+" CC (of the country where the IWU is located) to the address signals, then map (as "+" CC NDC SN) to the userinfo portion of the URI scheme used.	display-name shall be mapped from the address signals, if possible and network policy allows it.
"international number"	CC + NDC + SN	Add "+" to the address signals, then map (as "+" CC NDC SN) to the userinfo portion of the URI scheme used.	display-name shall be mapped from the address signals, if possible and network policy allows it.

Note 1 – 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 29/Q.1912.5 – Not Used

Table 30/Q.1912.5 – Not Used

Table 31/Q.1912.5 – Mapping of BICC/ISUP <u>CgPN</u> APRIs into SIP Privacy header field

<table>
<tr><th>BICC/ISUP Parameter/field</th><th>Value</th><th>SIP component</th><th>Value</th></tr>
<tr><td>Calling Party Number</td><td></td><td>Privacy header field</td><td>priv-value</td></tr>
<tr><td rowspan="2">APRI
~~(See Table 27 to determine which APRI to use for this mapping)~~</td><td>"presentation restricted"</td><td>priv-value</td><td>"id"
("id" included only if the P-Asserted-Identity header is included in the SIP INVITE)</td></tr>
<tr><td>"presentation allowed" or "presentation restricted by network"</td><td>priv-value</td><td>Omit Privacy header
or Privacy header without "id" if other privacy service is needed)</td></tr>
<tr><td colspan="4">~~NOTE – When Calling Party Number parameter is received, P-Asserted-Identity header is always derived from it as in Table 27.~~</td></tr>
</table>

<table>
<tr><th>Q.1912.5 Paragraph</th><th>Title</th><th>Comment</th></tr>
<tr><td>7.1.4</td><td>Hop Counter (Optional)</td><td>
E: Modify title 7.1.4:
Hop Counter ~~(Optional)~~

E: Modify clause 7.1.4.

~~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
<table>
<tr><th>Hop Counter value</th><th>Max-Forwards value</th></tr>
<tr><td>X</td><td>Y = Integer part of (X * Factor)</td></tr>
</table>
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.

UK: Replace the final paragraph of 7.1.4.1 with the following:
“The factor used in the calculation shall equal ‘1’. If the Hop Counter value is not available then a Hop Counter value of 30 shall be assumed and used to calculate the value of the Max Forwards.”
</td></tr>
</table>

<table>
<tr><th>Q.1912.5 Paragraph</th><th>Title</th><th>Comment</th></tr>
<tr><td>7.3.1.1</td><td>Setting for ACM Backward Call Indicators (mandatory) (Profiles A and B only)</td><td>
E: Replace clause 7.3.1.1 4th and 5th paragraphs and Table 34

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

~~Table 34/Q.1912.5 – Default Backward Call Indicators settings for Profile A~~
<table>
<tr><th>~~Parameter~~</th><th>~~Bits~~</th><th>~~Codes~~</th><th>~~Meaning~~</th></tr>
<tr><td>~~Interworking Indicator~~</td><td>~~I~~</td><td>~~1~~</td><td>~~"interworking encountered"~~</td></tr>
<tr><td>~~ISDN User part/BICC Indicator~~</td><td>~~K~~</td><td>~~0~~</td><td>~~"ISDN user part/BICC not used all the way"~~</td></tr>
<tr><td>~~ISDN Access Indicator~~</td><td>~~M~~</td><td>~~0~~</td><td>~~"terminating access non-ISDN"~~</td></tr>
</table>
~~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 BA Charge indicator
10 charge
bits DC Called party's status indicator
01 subscriber free if the 180 Ringing has been received.
00 no indication otherwise
bits FE Called party's category indicator
00 no indication
bits HG End-to-end method indicator
00 no end-to-end method available
bit I Interworking indicator
1 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
</td></tr>
</table>

Q.1912.5 Paragraph	Title	Comment
		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 [2] "Destination access is non-ISDN", so the call will not be released for that reason at an ISDN terminal. bit N Echo control device indicator 0 incoming echo control device not included 1 incoming echo control device included Bits PO SCCP method indicator 00 no indication **UK:** Modify the mapping as follows: bit I Interworking indicator 0 no interworking encountered (SS7 used all of the way) bit L Holding indicator ~~(national use)~~ 0 holding not requested
7.7.1	**Receipt of forward REL**	**E**: 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.
7.7.2	**Receipt of backward BYE**	**E:** 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~~. **UK:** Modify clause 7.7.2 5th paragraph, add new final sentence: The Location Field shall be set to "*user*".
7.7.3	**Autonomous release at O-IWU**	**E:** 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.

Q.1912.5 Paragraph	Title	Comment
7.7.4	**Receipt of RSC, GRS or CGB (ISUP)**	**E:** 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 <u>(Note 1)</u> SIP messages may be sent on receipt of a single GRS or CGB message. <u>Note 1: I.e. In the case where the Range subfield of the Range and Status Parameter contains a value equal to or greater than *"1"* multiple SIP messages will be sent on receipt of a single GRS or CGB message"</u> **UK**: Delete ETSI modification to clause 7.7.4, 2nd paragraph **E:** Modify clause 7.7.4, 4th paragraph: ~~Depending on local policy, a~~ <u>A</u> Reason header field containing the (Q.850) Cause Value of the REL message sent by the O-IWU ~~may~~ <u>shall</u> be added to the SIP message (BYE or CANCEL) to be sent by the SIP side of the O-IWU.
7.7.5	**Receipt of RSC or GRS (BICC)**	**E:** Modify clause 7.7.5, 2nd paragraph: "On receipt of a GRS message, one SIP message is sent for each call association. Therefore, multiple <u>(Note 1)</u> SIP messages may be sent on receipt of a single GRS message. <u>Note 1: I.e. In the case where the Range subfield of the Range and Status Parameter contains a value equal to or greater than *"1"* multiple SIP messages will be sent on receipt of a single GRS message"</u> **UK**: Delete ETSI modification to clause 7.7.5, 2nd paragraph **E:** Modify clause 7.7.5, 4th paragraph, 1st sentence: ~~Depending on local policy, a~~ <u>A</u> Reason header field containing the (Q.850) Cause Value of the REL message sent by the O-IWU ~~may~~ <u>shall</u> be added to the SIP message (BYE or CANCEL) to be sent by the SIP side of the O-IWU.
7.7.6	**Receipt of 4XX, 5XX, 6XX responses to INVITE**	**UK:** Replace Table 40 with the following:

Table 40/Q1912.5 Receipt of 4XX, 5XX or 6XX at O-IWU

SIP Response received	ISUP Cause Value	ISUP Location
400 Bad Request	95 Invalid message, unspecified	(BI)
401 Unauthorized	63 Service or option not available, unspecified	(BI)
402 Payment Required	63 Service or option not available, unspecified	(BI)
403 Forbidden	63 Service or option not available, unspecified	(BI)
404 Not Found	1 Unallocated (unassigned) number	(BI)
405 Method Not Allowed	63 Service or option not available	(BI)
406 Not Acceptable	79 Service or option not implemented, unspecified	(BI)
407 Proxy Authentication Required	63 Service or option not available, unspecified	(BI)
408 Request Time-out	18 No User Responding	(BI)
410 Gone	22 Number changed	(BI)
413 Request Entity Too Large	111 Protocol error, unspecified	(BI)
414 Request-URI Too Long	111 Protocol error, unspecified	(BI)
415 Unsupported Media Type	79 Service or option not implemented, unspecified	(BI)

SIP Response received	ISUP Cause Value	ISUP Location
416 Unsupported URI Scheme	127 Interworking, unspecified	(BI)
420 Bad Extension	79 Service or option not implemented, unspecified	(BI)
421 Extension Required	79 Service or option not implemented, unspecified	(BI)
423 Interval Too Brief	63 Service or option not available, unspecified	(BI)
433 Anonymity Disallowed	24 Call rejected due to ACR supplementary service	(User)
480 Temporarily Unavailable	31 Normal, unspecified	(User)
481 Call Leg/Transaction Does Not Exist	95 Invalid message, unspecified	(BI)
482 Loop Detected	25 Exchange routing error	(BI)
483 Too Many Hops	25 Exchange routing error	(BI)
484 Address Incomplete	28 Invalid number format (address incomplete)	(BI)
485 Ambiguous	1 Unallocated (unassigned) number	(BI)
486 Busy Here	34 No circuit/channel available	(BI)
487 Request Terminated	31 Normal, unspecified	(BI)
488 Not Acceptable Here	79 Service or option not implemented, unspecified	(BI)
491 Request Pending	No mapping	
493 Undecipherable	127 Interworking, unspecified	(BI)
500 Server Internal Error	47 Resource unavailable, unspecified	(BI)
501 Not Implemented	79 Service or option not implemented, unspecified	(BI)
502 Bad Gateway	111 Protocol error, unspecified	(BI)
503 Service Unavailable	42 Switching equipment congestion	(BI)
504 Server Time-out	102 Recovery on timer expiry	(BI)
505 Version Not Supported	127 Interworking, unspecified	(BI)
513 Message Too Large	111 Protocol error, unspecified	(BI)
580 Precondition Failure	34 No circuit/channel available	(TN)
600 Busy Everywhere	17 User busy	(User)
603 Decline	21 Call rejected	(User)
604 Does Not Exist Anywhere	4 Send special information tone	(User)
606 Not Acceptable	79 Service or option not implemented, unspecified	(BI)

1.2 Additions

Add new paragraphs as shown below:

Q.1912.5 Paragraph	Title	Comment
6.zz	**Receipt of Information and Information Request messages**	ISUP Information and Information Request messages shall be handled as follows: **a) for Profiles A and B** On receipt of an INR message from the ISUP procedures the IWU shall instruct the ISUP procedures to respond by generating an INF message containing a National Information indicators parameter with all indicators set to value "0". On receipt of an INF message from the ISUP procedures the IWU shall discard the message. **b) for Profile C** On receipt of an INR message from the ISUP procedures the IWU shall instruct the SIP procedures to encapsulate the INR message in an INFO or 183 SESSION PROGRESS message as appropriate. On receipt of an INF message from the ISUP procedures the IWU shall instruct the SIP procedures to encapsulate the INF message in an INFO or 183 SESSION PROGRESS message as appropriate. On receipt of a SIP INFO message containing an INR/INF, the INR/INF shall be de-encapsulated and passed to the ISUP procedures.
7.1.6	**P-Charging Vector header field**	The O-IWU shall include a P-Charging Vector header field (see RFC 3455 [35]) in the INVITE. The format shall be as described in section 4.2.3 of this document.
7.2.2	**P-Charging Vector header field**	The O-IWU shall include the same P-Charging Vector header field (see RFC 3455 [35]), as generated in the initial INVITE, in subsequent INVITEs. The format shall be as described in section 4.2.3 of this document.
7.ww	**Receipt of Information and Information Request messages**	ISUP Information and Information Request messages shall be handled as follows: **a) for Profiles A and B** On receipt of an INR message from the ISUP procedures the IWU shall instruct the ISUP procedures to respond by generating an INF message containing a National Information indicators parameter with all indicators set to value "0". On receipt of an INF message from the ISUP procedures the IWU shall discard the message. **b) for Profile C** On receipt of an INR message from the ISUP procedures the IWU shall instruct the SIP procedures to encapsulate the INR message in an INFO message. On receipt of an INF message from the ISUP procedures the IWU shall instruct the SIP procedures to encapsulate the INF message in an INFO message. On receipt of a SIP INFO or 183 SESSION PROGRESS message containing an INR/INF, the INR/INF shall be de-encapsulated and passed to the ISUP procedures.
7.xx	**Receipt of Network Suspend (SUS)/Network Resume (RES) message**	If the O-IWU is the controlling exchange for the Suspend/Resume procedure, the actions taken on the ISUP side upon receipt of the Suspend (SUS) or Resume (RES) message are described in TSG/SPEC/007 Section 4.2.2.1. SUS/RES is not interworked in Profile A or B operation. In the Profile C (SIP-I) case, the SUS/RES is encapsulated in the MIME body of an INFO message.

END OF TSG/SPEC/017§1

2 Q.1912.5 Annex A BICC specific interworking for basic call

2.1 Introduction

This annex contains additional interworking to/from SIP which are particular to the BICC protocol.

2.2 Exceptions

UK: Annex not required.

END OF TSG/SPEC/017§2

3 Q.1912.5 Annex B Interworking for ISDN supplementary services

3.1 Exceptions

Q.1912.5 Annex B Paragraph	**Title**	**Comment**
B.1	**Interworking of CLIP/CLIR supplementary service to SIP networks**	**E:** Modify Annex B.1 2nd paragraph 2nd sentence: 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 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. **UK:** Replace last sentence of ETSI modification to Annex B.1 last paragraph with the following: The information contained in the following encapsulated ISUP parameters shall be de-encapsulated and used: Calling Party Number Generic Number Presentation Number. National forward call indicators, CBI bit
B.2	**Interworking of COLP/COLR supplementary service to SIP networks**	**UK:** Modify as shown: "Profiles A and B ~~FFS.~~ Interworking not possible."
B.3	**Interworking of Direct-Dialling-In (DDI) supplementary service to SIP networks**	**UK:** Modify as shown: "Profiles A and B ~~FFS.~~ As this service relies only on the Called Party Number information, interworking is as described in sections 6.1.3.1 and 7.1.2 above."
B.5	**Interworking of Sub-addressing (SUB) supplementary service to SIP networks**	**UK:** Modify as shown: "Profiles A and B ~~FFS.~~ Interworking not possible."
B.6	**Interworking of Call Forwarding Busy (CFB)/Call Forwarding No Reply (CFNR)/Call Forwarding Unconditional (CFU) supplementary services to SIP networks**	**UK:** Profiles A and B The LDLI parameter will not be interworked
B.7	**Interworking of Call Deflection (CD) supplementary service to SIP networks**	**UK:** Profiles A and B The LDLI parameter will not be interworked

Q.1912.5 Annex B Paragraph	**Title**	**Comment**
B.10	**Interworking of Call Hold (HOLD) supplementary service to SIP networks**	**E:** 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~~ <u>recvonly</u> media stream, or the attribute may be omitted, since sendrecv is the default • "a=recvonly", if the stream was previously an inactive media stream
B.17	**Interworking of Multi-Level Precedence and Preemption (MLPP) supplementary service to SIP networks**	**E:** Not supported
B.18	**Interworking of Global Virtual Network Service (GVNS) supplementary service to SIP networks**	**E:** “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.”
B.19	**Interworking of International Telecommunication Charge Card (ITCC) supplementary service to SIP networks**	**E:** Not supported
B.20	**Interworking of Reverse Charging (REV) supplementary service to SIP networks**	**E:** Not supported
B.21	**Interworking of User-to-User supplementary service to SIP networks**	**UK: Modify text as shown** Profiles A and B The IWU shall act in accordance with the procedures described within ITU-T Rec. Q.737.1, under the clause<u>s</u> head<u>ed</u>~~ing~~ "Interactions with other networks"<u>. In the UK, SIP networks shall be treated as “No. 7 network not supporting the service”</u>.
B.22	<u>**Interworking of Anonymous Call Rejection (ACR) supplementary service to SIP networks**</u>	**E: Add Annex B.22** This section describes the interworking of the ETSI ACR service as described in 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.

Q.1912.5 **Annex B Paragraph**	**Title**	**Comment**
B.23	**Interworking of Call Forwarding on Subscriber Not Reachable (CFNRc)**	**UK:** Add Profiles A and B The IWU shall act in accordance with the procedures described within 2.7/Q.732.2-5, under the clause heading "Interactions with other networks". The LDLI parameter will not be interworked Profile C (SIP-I) Call forwarding in the PSTN requires no additional interworking beyond use of ISUP encapsulation.

3.2 Additions

The following UK supplementary services require no additional interworking for all profiles other than that specified for basic call:

Indirect Access

UK Carrier Pre-selection

UK Number Portability

Targeted Transit

The following UK supplementary services will interwork for profile C only. They require no additional interworking beyond use of ISUP encapsulation.

Enhanced Operator Services

Ring Back When Free

END OF TSG/SPEC/017§3

4 Q.1912.5 Annex C

4.1 Introduction

This annex contains references to normative Internet Engineering Task Force (IETF) RFCs and materials originally sourced from the IETF but deemed normative to this Specification.

4.2 Exceptions

RFC 3262 shall be supported for all profiles.

The following sub-sections show the required UK modifications / clarifications (as required) to:

- the RFC's shown in the table in C.1.1.2 of Q.1912.5;

- additional RFC's required for use in the UK.

4.2.1 RFC 3966 (Obsoletes RFC 2806)

4.2.1.1 UK Local Numbers

In the UK, "local-numbers" are used to carry numbers that are not suitable for representation in the "global-number" format (e.g. 999, 118xxx, 8xxx etc). In these cases the "context" shall include a "descriptor" containing the "global-number-digits" "+44".

4.2.1.2 Modification to Sub-Section 5.1.3

Replace with the following

"5.1.3. Alphabetic, *, and # Characters as Identifiers

As called and calling terminal numbers (TNs) are encoded in BCD in ISUP, six additional values per digit can be encoded, sometimes represented as the hexadecimal characters A through F. Similarly, DTMF allows for the encoding of the symbols *, #, and A through D. However, in accordance with E.164, these may not be included in global numbers. Their meaning in UK local numbers if required by an application is defined below.

UK ISUP	SIP	UK ISUP
Hex A	ASCII character "A"	Hex A
Hex B	ASCII character "B"	Hex B
Hex C	ASCII character "C"	Hex C
Hex D	ASCII character "D"	Hex D
Hex E	ASCII character "E"	Hex E
	ASCII character "F"	Not Mapped
ST	Not Mapped	
	ASCII character "*"	Not Mapped
	ASCII character "#"	Not Mapped

4.2.2 RFC 3261

4.2.2.1 SDP Offer

RFC 3261 section	Title	Comment
13.2.1	**Creating the Initial Invite**	**UK:** Modify the SDP offer-answer rules for the initial INVITE (as shown in the bulleted list) as follows: • The initial SDP offer MUST be in ~~either~~ an INVITE ~~or, if not there, in the first reliable non-failure message from the UAS back to the UAC. In this specification that is the final 2XX response~~. • ~~If the initial offer is in an INVITE, t~~The SDP answer MUST be in a reliable non-failure message from UAS back to UAC which is correlated to that INVITE. ~~For this specification, that is only the final 2xx response to that INVITE. That same exact answer MAY also be placed in any provisional responses sent prior to the answer~~. The UAC MUST treat the first session description it receives in the first reliable non-failure message as the answer, and MUST ignore any session descriptions in subsequent responses to the initial INVITE. ~~• If the initial offer is in the first reliable non-failure message from the UAS back to UAC, the answer MUST be in the acknowledgement for that message (in this specification, ACK for a 2xx response).~~ • After having ~~sent or~~ received an SDP answer to the first SDP offer, the UAC MAY generate subsequent SDP offers in requests based on rules specified for that method, but only if it has received SDP answers to any previous SDP offers, and has not sent any SDP offers to which it has ~~hasn't gotten~~ not received an answer. • Once the UAS has sent ~~or received~~ an SDP answer to the initial SDP offer, it MUST NOT generate subsequent SDP offers in any responses to the initial INVITE. This means that a UAS based on this specification alone can never generate subsequent offers until completion of the initial transaction. Concretely, the above rules specify ~~two~~one exchange~~s~~ for UAs compliant to this specification alone - the SDP offer is in the INVITE, and the SDP answer in the 2xx (and possibly in a 1xx as well, with the same value), ~~or the offer is in the 2xx, and the answer is in the ACK~~. All user agents that support INVITE MUST support ~~these two~~ this exchange~~s~~.

4.2.3 RFC 3455

In the UK the P-Charging-Vector (section 4.6 of RFC 3455 [35]) shall be included in the SIP INVITE and shall contain the following three mandatory fields:-

icid-value

icid-generated-at

orig-ioi

where the:

icid-value (IMS Charging Identifier) is a 10 character unique identifier for the call, generated by the Session Control Function originating the first SIP INVITE. Each character shall be one of the 26 ASCII letters (case sensitive), one of the 10 ASCII digits, or the hyphen-minus.

icid-generated-at is the IP address of the Source Session Control Function.

orig-ioi (Inter Operator Identifier) is the name of the network generating the first SIP INVITE in the call establishment process, as defined in section 4.4.2.3. of ND1612 [34]

The P-Charging-Vector with contents as described above shall be inserted by the Source Session Control Function originating the SIP INVITE. It shall be passed on unchanged at all subsequent SIP aware entities if the onward path uses SIP signalling to a node within the trusted domain. If a SIP INVITE that does not contain a P-Charging-Vector is received at a Session Control Function then it shall generate one with contents as described above. The contents of the P-Charging-Vector may be used at any SIP aware entity for inclusion in Call Detail Records or for other application specific logic.

END OF TSG/SPEC/017§4

5 Q.1912.5 Appendix I Interworking scenarios between SIP and BICC

5.1 Introduction

This appendix defines typical interworking scenarios between SIP and BICC.

5.2 Exceptions

UK: Appendix not required

END OF TSG/SPEC/017§5

6 Q.1912.5 Appendix II Interworking scenarios between SIP and ISUP

6.1 Introduction

This appendix defines typical interworking scenarios between SIP and ISUP.

UK: This appendix is for information only and is not applicable to the Purple release.

END OF TSG/SPEC/017§6

7 Q.1912.5 Appendix III Interworking scenarios between Profile C (SIP-I) and ISUP

7.1 Introduction

This appendix defines typical interworking scenarios between SIP Profile C and ISUP.

UK: This appendix is for information only and may not be applicable in the UK.

END OF TSG/SPEC/017§ 7