NICC ND 1037 V1.1.1 (2022-05)

NICC Document

SIP - ISUP Interworking

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Contents

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Foreword

This NICC Document (ND) has been produced by NICC SIP TG

1 Scope

The present document specifies the interworking function to be performed between C7 ISUP, as specified in ND1007 [N2] and SIP, as specified in ND1035 [N1].

This specification is only applicable for networks where a transit capability is provided between points of interconnect where, for any particular call, one interconnect is via C7 ISUP and the other is via SIP, i.e. where the transit network is providing the C7 ISUP<-> SIP protocol conversion explicitly between the two points of interconnect.

It is acknowledged that within networks similar interworking functions may be employed, e.g. for calls originating from a network that uses SIP as its internodal signalling and inter-works this to C7 ISUP for interconnect to other networks. For such situations this specification is recommended by NICC as guidance to the network operators to assist in their network-specific solutions; however, network-specific service requirements may cause variations to this.

2 References

References in this section have been added for the UK endorsement of 3GPP TS 29.163 [N5] and use the prefix letter N. For any references in endorsement text that do not have the prefix N refer to the 3GPP TS 29.163 [N5] section 2.

2.1 Normative references

The following referenced documents are indispensable for the application of this document. For dated references, only the edition cited applies. For non-specific references, the latest edition of the referenced document (including any amendments) applies.

[N1] ND1035 SIP Network to Network Interface Signalling

[N2] ND1007 ISDN User Part (ISUP)

[N3] ND1016 Requirements on Communications Providers in relation to Customer Line Identification display services and other related services

[N4] ND1704 End-to-End Network Performance Rules & Objectives for the Interconnection of NGNs

[N5] 3GPP TS 29.163 V13.9.0

[N6] RFC5031 SOS URN

[N7] RFC3261 SIP: Session Initiation Protocol

[N8] ND1647 SIP-NNI Basic Voice Architecture

3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

3GPP	3rd Generation Partnership Project
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
NNI	Network to Network Interface
SIP	Session Initiation Protocol
URN	Uniform Resource Name
UK	United Kingdom

NOTE: Any abbreviations found in endorsement text that are not found in the above list can be found in section 3.2 of 3GPP TS 29.163 [N5].

4 General

The present document is constructed as an endorsement of 3GPP TS 29.163 [N5] adapted for use in the UK when interworking between SIP and ISUP.

As described in section 5, the UK architecture is not restricted to IMS architectures.

The main section of 3GPP TS 29.163 [N5] that is of relevance is 7.2.3 which describes the mapping of signalling parameters between SIP and ISUP. To aid interpretation NICC has developed transition diagrams for the SIP to ISUP and ISUP to SIP directions where each transition represents receipt of a stimulus signalling message. These diagrams are presented in sections 6.1 and 6.2 along with tables that indicate the sections of 3GPP TS 29.163 [N5] pertinent to the transition. The intent of this is that the reader can easily identify the required mapping when a given stimulus message is received by referencing the endorsement text for the indicated sections in Annex A.

Interpretation of sections in 3GPP TS 29.163 [N5] other than 6, 7.2.3, 7.4 and 7.5 is left for individual implementations. However, it should be noted that any UK implementation must comply with Ofcom General Conditions, and relevant NICC ND documents in the Normative References of the present document including ND1007 [N2], ND1035 [N1], ND1016 [N3] and ND1704 [N4].

4.1 Exclusions from UK Endorsement of TS 29.163

Signalling of IMS Centralised Services (ICS) is not supported across UK NNIs. Therefore, any reference to ICS within the endorsed text has been deleted.

5 Architecture

5.1 Endorsement of TS 29.163 Section 6

TS 29.163 section	Title	Comment
6.1	Interworking reference model	Insert the following note under Figure 1: "UK NOTE on the interpretation of the reference architecture in Figure 1: • The reference architecture in Figure 1 is defined in terms of 3GPP IMS functional entities but the application of this specification in UK networks is not restricted to an IMS and does not depend on such a functional architecture. • ND1647 [N8] specifies the UK SIP NNI architecture for basic voice calls but does not explicitly cover interworking between SIP and ISUP/BICC. • For the control plane, ND1647 [N8] defines the use of the Qs interface (based on SIP as specified by ND1035 [N1]) between interconnected signalling border functional entities, but this may also be applied between other SIP entities within a network. This corresponds to the Mg/Mj interfaces in Figure 1. In 3GPP the Mg/Mj interfaces are internal to a single IMS but for UK purposes the same interworking requirements apply whether the interface is internal or external. • For the user plane, ND1647 [N8] defines the use of the Qm interface between interconnected media border functional entities, but this may also be applied between other media entities within a network. This corresponds to the Mb interface in Figure 1. In 3GPP the Mb interface is internal to a single IMS but for UK purposes the same interworking requirements apply whether the interface is internal or external. • The required interworking function between UK SIP and UK ISUP corresponds to the signalling interworking functionality of the MGCF in Figure 1."
6.1.1	Interworking reference points and interfaces	Insert the following paragraph under the paragraph which begins 'Protocol for Mg reference point...' "The SIP profile specified by ND1035 [N1] shall apply. This defines the minimum capability required for UK voice calls and does not preclude support for additional SIP capabilities e.g. in accordance with 3GPP TS 24.229 [9]." Insert the following paragraph under the paragraph which begins 'Protocol for Mn reference point...' "Mn is out of scope for the UK endorsements." Insert the following paragraph under the paragraph which begins 'Protocol for Mb reference point...' "The UK SIP NNI profile allows for IPV4."
6.1.2.2	Media Gateway Control Function (MGCF)	Insert the following paragraph below the existing text:

		“The required interworking function between SIP [N1]and ISUP [N2] corresponds to the signalling interworking functionality of the MGCF.”

6 Interworking

Sections 6.1 and 6.2 describe interworking of SIP to ISUP and ISUP to SIP respectively through endorsement of 3GPP TS 29.163 [N5]. The transition diagrams and tables described in section 4 are presented prior to the endorsement text.

Parameter/header field values not described in 6.1.1 and 6.2.1 shall be assigned by normal protocol procedures.

6.1 SIP to ISUP Interworking

The UK Transition diagram for SIP to ISUP is shown in Figure 6.1 below:

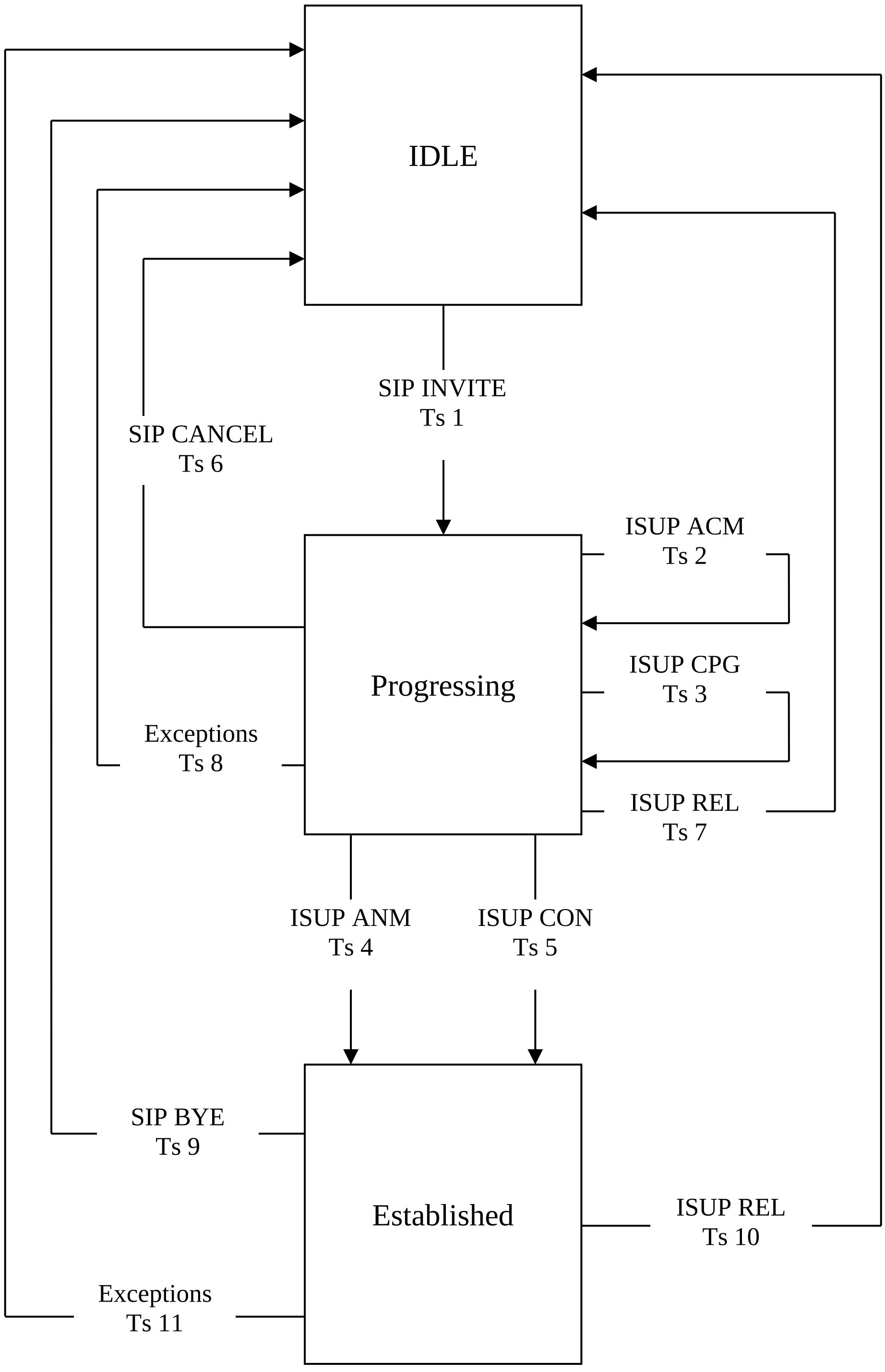

Figure 6.1 – SIP to ISUP Transition Diagram

The references relevant to Figure 6.1 are given in Table 6.1

Transition	Received Signalling Message	3GPP TS 29.163 references
Ts 1	SIP INVITE	7.2.3.1.1, 7.2.3.1.2
Ts 2	ISUP ACM	7.2.3.1.4.0, 7.2.3.1.4A, 7.2.3.1.4B
Ts 3	ISUP CPG	7.2.3.1.4.0, 7.2.3.1.4A, 7.2.3.1.4B
Ts 4	ISUP ANM	7.2.3.1.5
Ts 5	ISUP CON	7.2.3.1.5
Ts 6	SIP CANCEL	7.2.3 1.6
Ts 7	ISUP REL *(before answer)*	7.2.3.1.8
Ts 8	NONE (*Nodal exceptions before answer)*	7.2.3.1.9, 7.2.3.1.10
Ts 9	SIP BYE	7.2.3 1.6
Ts 10	ISUP REL *(after answer)*	7.2.3.1.8
Ts 11	NONE (*Nodal exceptions after answer)*	7.2.3.1.9, 7.2.3.1.10

Table 6.1 – TS 29.163 references for SIP to ISUP interworking

6.1.1 Endorsement of TS 29.163 Section 7.2.3.1 Incoming call interworking from SIP to ISUP at I-MGCF

<table>
<tr><th>TS 29.163 section</th><th>Title</th><th>Comment</th></tr>
<tr><td>7.2.3.1.1</td><td>Sending of IAM</td><td>Add the following below the note under paragraph 2.

“UK NOTE: A number of factors mean that it is not possible to guarantee the end to end routing of calls in the UK. The implication of this is that some calls may signal preconditions but SIP to UK ISUP mapping will occur at an MGCF that does not support them. Where this occurs callers may encounter unintended behaviour e.g. a portion of a pre-answer announcement is not audible.”

Delete the paragraph which begins “If a Continuity Check is supported….”

Delete Figure 3

Modify the paragraph which begins “If a Continuity Check is not supported….” as follows

If ~~Continuity Check procedure is not supported in the ISUP network and~~the SIP precondition extension is included, and the SDP in the received INVITE request contains preconditions not met, the I-MGCF shall delay sending the IAM until the SIP preconditions are met ~~and set the continuity indicators in the resulting IAM to “Continuity check not required”~~.

Add the following Note below the paragraph which begins “If a Continuity Check is not supported….”

“UK Note: ISUP Continuity Check is not supported in UK ISUP”

Modify the paragraph which begins “If an MGCF discovers…” as follows:
“If an MGCF ~~discovers~~<u>identitfies</u> an emergency call it shall~~, depending on national requirements, map that to appropriate indication in ISUP.~~ set the ISUP Calling Party’s Category parameter to:
0 0 0 0 1 0 1 1 Calling subscriber with priority”

Add the following text below the text “0 0 0 0 1 0 1 1 Calling subscriber with priority”.

“An Emergency call shall be identified if a Resource-Priority header field, populated as described in Annex A of ND1035 [N1], is received.

Alternatively an emergency call may be identified if either of the following are received:
• An sos-URN as described in RFC5031 [N6]
• An emergency call service address e.g. 999, 112”</td></tr>
<tr><td>7.2.3.1.2.0</td><td>General</td><td>Modify the text as follows:

“The following ISDN user part parameters description can</td></tr>
</table>

<table>
<tr><td></td><td></td><td>be found in ~~ITU-T Recommendation Q.763~~ ND1007 [N2].”</td></tr>
<tr><td>7.2.3.1.2.1</td><td>Called party number</td><td>Modify the first paragraph as follows:

“The ~~E.164~~address encoded in the Request-URI shall be mapped to the called party number parameter of the IAM message.”

Replace table 2 with below

Table 2: Coding of the called party number

<table>
<tr><th>INVITE→
Request-URI</th><th>IAM→
Called Party Number</th></tr>
<tr><td rowspan="5">E.164 address (format +CC NDC SN) (e.g. as User info portion of a SIP URI with user=phone, or as tel URI)</td><td>Address Signal:
Analyse the information contained in received E.164 address.
If CC is country code of the network in which the next hop terminates, then remove "+CC" and use the remaining digits to fill the Address signals. (NOTE 2)
If CC is not the country code of the network in which the next hop terminates, then remove "+" and use the remaining digits to fill the Address signals. (NOTE 2)
(NOTE 1)</td></tr>
<tr><td>Odd/even indicator: set as required</td></tr>
<tr><td>Nature of address indicator:
Analyse the information contained in received E.164 address.
If CC is country code of the network in which the next hop terminates, then set Nature of Address indicator to "National (significant) number.
If CC is not the country code of the network in which the next hop terminates, then set Nature of Address indicator to "International number".</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 rowspan="5">Non E.164 numbers (as a local-number with phone-context=+44 in the User Info portion in a SIP URI with user=phone, or as a local number with phone-context=+44 in a tel URI)</td><td>Address Signal:
Use received non E.164 number to fill the Address signals. (NOTE 2)</td></tr>
<tr><td>Odd/even indicator: set as required</td></tr>
<tr><td>Nature of address indicator:
set Nature of Address indicator to 1111 1110 “UK Specific Address” "network-specific number"~~.~~</td></tr>
<tr><td>Internal Network Number Indicator:
1 routing to internal network number not allowed</td></tr>
<tr><td>Numbering plan Indicator:
Set Numbering Plan indicator to “001 ISDN (Telephony) numbering plan (Rec. E.164)”</td></tr>
<tr><td colspan="2">NOTE 1: If PSTN XML and ISUP Sending Terminated (ST) signal are supported as a network option, then the PSTN XML sendingCompleteIndication, if present, is mapped to the sending terminated digit (hexadecimal digit F) in the address signals field of the Called Party Number parameter.
NOTE 2: Overlap sending is not supported in UK SIP networks so an ISUP sending terminated (ST) signal shall be appended to the address digits once constructed</td></tr>
</table>
</td></tr>
<tr><td>7.2.3.1.2.2</td><td>Nature of connection indicators</td><td>Modify the text for the setting of bits DC as follows:

“bits DC Continuity check indicator

0 0 *continuity check not required*~~, if the continuity check procedure is not supported in the succeeding network (figure 4).~~
~~0 1 *continuity check required*, if a continuity check shall be carried out on the succeeding circuit. (figure 3)~~
~~1 0 *continuity check performed on a previous circuit otherwise*, if the continuity check procedure is supported in the succeeding network, but shall not be carried out on the succeeding circuit otherwise. (figure 3)~~”</td></tr>
<tr><td>7.2.3.1.2.3</td><td>Forward call indicators</td><td>Modify the text for the setting of bit D as follows:

“bit D Interworking indicator
~~1~~0 no interworking encountered

~~As a network operator option, the value D = 0 "No interworking encountered" is used if the TMR = 64 kBit/s unrestricted is used.~~
~~NOTE: This avoids sending of a progress indicator~~</td></tr>
</table>

		~~with progress information 0 0 0 0 0 0 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 by an ISDN terminal."~~
7.2.3.1.2.4	Calling party's category	Replace the existing text with the following: "The ISUP CPC shall set to '10, Ordinary Calling Subscriber' unless the call is an emergency call (see endorsement of TS.29.163 [N5] section 7.2.3.1.1)."
7.2.3.1.2.4A	Originating Line Information	Void the section
7.2.3.1.2.5	Transmission medium requirement	Replace Table 2A with the Table 2A below.

Table 2a: Coding of TMR/USI/HLC from SDP: SIP to ISUP

	m= line		**b= line (NOTE 3)**	**a= line**	**TMR parameter**	**USI parameter (optional)**		**HLC IE in the ATP parameter (optional)**
<media>	<transport>	<fmt-list>	<modifier>:<bandwidth-value> (NOTE 4)	rtpmap:<dynamic-PT> <encoding name> <clock rate>[<encoding parameters>]	TMR codes	Information Transfer Capability	User Information Layer 1 Protocol Indicator	High Layer Characteristics Identification
audio	RTP/AVP	8	N/A or AS: up to (64 kbit/s+ RTP/UDP/IP overhead)	N/A	"3.1KHz audio"			(NOTE 2)
audio	RTP/AVP	Dynamic PT	N/A or AS: up to (64 kbit/s + RTP/UDP/IP overhead)	rtpmap:<dynamic-PT> PCMA/8000	"3.1KHz audio"			(NOTE 2)
audio	RTP/AVP	Dynamic PT	AS: (64 kbit/s+ RTP/UDP/IP overhead)	rtpmap:<dynamic-PT> CLEARMODE/8000 (NOTE 1)	"64 kbit/s unrestricted"	"Unrestricted digital information"		

NOTE 1: CLEARMODE is specified in RFC4040 [69].

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 MGCF should return an b:AS bandwidth modifier with a bandwidth of 64kbit/s + RTP/UDP/IP overhead in the SDP answer to request that the peer does not send with a higher bandwidth. If the received b=line indicates a bandwidth greater than 64kbit/s + RTP/UDP/IP overhead, the MGCF should also accept the incoming call.

NOTE 4 :<bandwidth value> for <modifier> of AS is in units of kbit/s.

7.2.3.1.2.6	Calling party number	Replace Tables 3 & 4 with the Tables 3 & 4 below. Delete Table 5

Table 3 - Mapping of the "P-Asserted-Identity" header field and privacy indications to ISUP Calling party number parameter and National Forward Call Indicators (NFCI) parameter CLI Blocking Indicator (CBI)

Has a "P-Asserted-Identity" header field containing a URI that contains an E.164 address (NOTE 1, NOTE 2) been received?	**Has a "Privacy" header field containing priv-values "id" or "header" been received?**	**Has a "Privacy" header field containing priv-value "user", or a "From" header field containing 'anonymous' been received?**	**Calling Party Number parameter**		**NFCI parameter**
			Address Signals	**APRI**	**CBI**
No (Note 3)	No	No	Inject a network provided E.164 number (See Table 4)	Set to "presentation restricted by network".	Set to "*network number may not be disclosed to the called user"* (0).
			As a network option the IWU may omit the Calling Party Number parameter and inject a PCLI parameter as described in ND1007 [N2]		
No (Note 3)	Yes	No	Inject a network provided E.164 number (See Table 4)	Set to "presentation restricted".	Set to "Network Number may (subject to interaction with CLIR) be disclosed to the called user" (1)
			As a network option the IWU may omit the Calling Party Number parameter and inject a PCLI parameter as described in ND1007 [N2]		
No (Note 3)	No	Yes	Inject a network provided E.164 number (See Table 4)	Set to "presentation restricted".	Set to "*Network Number may (subject to interaction with CLIR) be disclosed to the called user*" (1)
			As a network option the IWU may omit the Calling Party Number parameter and inject a PCLI parameter as described in ND1007 [N2]		
No (Note 3	Yes	Yes	Inject a network provided E.164 number (See Table 4)	Set to "presentation restricted".	Set to "*Network Number may (subject to interaction with CLIR) be disclosed to the called user*" (1)
			As a network option the IWU may omit the Calling Party Number parameter and inject a PCLI parameter as described in ND1007 [N2]		
Yes	No	No	Derived from "P-Asserted-Identity" (see Table 4)	Set to "presentation allowed".	Set to "*Network Number may (subject to interaction with CLIR) be disclosed to the called user*" (1)
Yes	Yes	No	Derived from "P-Asserted-Identity" (see Table 4)	Set to "presentation restricted by network".	Set to "*network number may not be disclosed to the called user"* (0).
Yes	No	Yes	Derived from "P-Asserted-Identity" (see Table 4)	Set to "presentation restricted".	Set to "*Network Number may (subject to interaction with CLIR) be disclosed to the called user*" (1)
Yes	Yes	Yes	Derived from "P-Asserted-Identity" (see Table 4)	Set to "presentation restricted".	set 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. In this case, the sip URI with "user=phone" and a specific host portion, should be used.					
NOTE 2: CLIs received by the MGCF have previously been validated within the UK IP domain so are assumed to be E.164 in global number format.					
NOTE 3: Only calls where a Resource-Priority header field is present and populated in accordance ND1035 [N1] shall be					

Has a "P-Asserted-Identity" header field containing a URI that contains an E.164 address (NOTE 1, NOTE 2) been received?	Has a “Privacy” header field containing priv-values “id” or “header” been received?	Has a “Privacy” header field containing priv-value “user”, or a “From” header field containing ‘anonymous’ been received?	Calling Party Number parameter		NFCI parameter
			Address Signals	APRI	CBI
mapped. Otherwise the call shall be failed with a 603 response.					

Table 4 - BICC/ISUP Calling Party Number parameter derivation

BICC/ISUP Calling Party Number Parameter field	Derivation
Odd/even indicator	Set according to the number of address signals
Nature of Address indicator	(NOTE 1) If “CC” encoded in the P-Asserted-Identity URI is equal to the “CC” of the country where the MGCF is logically located then set to "national (significant) number" else set to "international number" In the case where the Address Signals are injected by the MGCF the NOA is set to "national (significant) number".
Number incomplete indicator	set to “complete”
Numbering plan indicator	set to “ISDN/Telephony (E.164)”
Address Presentation Restricted indicator	see Table 3
Screening indicator	set to “network provided”
Address signals	(NOTE 1) If NOA is "national (significant) number" then set to "NDC" + "SN" encoded in the URI of the received P-Asserted-Identity. If NOA is "international number" Then set to "CC"+"NDC"+"SN" encoded in the URI of the received P-Asserted-Identity. In the case where the Address Signals are injected by the MGCF will be in the form "NDC" + "SN".
NOTE 1: The E.164 numbers considered within the present document are composed by a Country Code (CC), followed by a National Destination Code (NDC), followed by a Subscriber Number (SN). On the IP side, the numbers are international public telecommunication numbers ("CC"+"NDC"+"SN") and are prefixed by a "+" sign. On the CS side CC is omitted when the ISUP Nature of Address is set to 'National (Significant) Number'.	

7.2.3.1.2.7	Generic number	Replace Table 6 with the Tables 6 & 6a below.

Table 6 - Mapping of SIP “From” and “Privacy” header to ISUP Generic Number parameter

Has a "From" header field containing a URI with an E.164 address been received?	Has a Privacy header been received containing priv-value “user”	Generic Number (Additional Calling Party Number) parameter address signals	Generic Number parameter APRI
No	-	Parameter not included	-
Yes	No	Derived from “From” header URI (see Table 6)	set to “presentation allowed”.
Yes	Yes	Derived from “From” header URI (see Table 6)	set to “presentation restricted ”.

Table 6a - ISUP Generic Number parameter derivation

BICC/ISUP Generic Number Parameter field	Derivation
Number qualifier indicator	set to “additional calling party number”
Odd/even indicator	Set according to the number of address signals
Nature of Address indicator (NOA)	(NOTE 1) If “CC” encoded in the URI is equal to the “CC” of the country where the MGCF is logically located then set to "national (significant) number" else set to "international number"
Number incomplete indicator	set to “complete”
Numbering plan indicator	set to “ISDN/Telephony (E.164)”
Address Presentation Restricted indicator	see Table 6
Screening indicator	set to 'user provided, not verified'
Address signals	if NOA is "national (significant) number" then set to "NDC" + "SN" If NOA is "international number" Then set to "CC"+"NDC"+"SN"
NOTE 1: The E.164 numbers considered within the present document are composed by a Country Code (CC), followed by a National Destination Code (NDC), followed by a Subscriber Number (SN). On the IP side, the numbers are international public telecommunication numbers ("CC"+"NDC"+"SN") and are prefixed by a "+" sign. On the CS side CC is omitted when the ISUP Nature of Address is set to 'National (Significant) Number'.	

7.2.3.1.2.9	Hop counter (national option)	Delete “(national option)” from the title Append the following text to the paragraph under Table 7. “The factor used in the calculation shall equal ‘2’. If after processing the Max Forward value utilising the algorithm with a factor of 2 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.”
7.2.3.1.2.11	Location number	Insert the following below the section title: “SIP TG editor’s note: FFS”
7.2.3.1.2.12	Support of ICS	Void the section
7.2.3.1.2.13	UID capability indicators (National option)	Delete “(national option)” from the title
7.2.3.1.2.14	Called IN number and original called IN number (optional)	Void the section
7.2.3.1.2A	Coding of the IAM when Number Portability is supported	Void the section
7.2.3.1.2B	Coding of the IAM for Carrier Routeing	Void the section
7.2.3.1.3	Sending of COT	Void the section
7.2.3.1.3A	Sending of SAM	Void the section
7.2.3.1.4.0	General	Modify the text in the paragraph which begins “For a speech call that is not identified as an "ICS call"….” as follows.. ~~“For a speech call that is not identified as an "ICS call" as specified in clause 7.2.3.1.2.12, i~~If the INVITE request includes the P-Early-Media header field, the I-MGCF shall include in the SIP 180 Ringing response a P-Early-Media header field authorizing backward early media, except when - the I-MGCF has already sent a reliable provisional response

		(see IETF RFC 3262 [54]) including a P-Early-Media header, as defined in IETF RFC 5009 [89], and - the most recently sent P-Early-Media header field authorized backward early media." Modify the text in the paragraph which begins "If the INVITE request doesn't include…" as follows.. "If the INVITE request doesn't include the P-Early-Media header field with the "supported" parameter ~~and if the speech call is not identified as the "ICS call" as specified in clause 7.2.3.1.2.12~~, then as a network option the I-MGCF may include in the SIP 180 Ringing response a P-Early-Media header field authorizing backward early media." Modify the text in the paragraph which begins "As a network option, an I-MGCF may…" as follows.. "As a network option, an I-MGCF may, in addition, generate a Call-Info header field, or an Alert-Info header field according to rules and procedures of IETF RFC 3261 [19] to indicate an optional alternative to provide media instead of the in-band media received from the PSTN." Delete the following text: "For the speech call identified as the "ICS call" as specified in clause 7.2.3.1.2.12, if the I-MGCF has received the P-Early-Media header field in the initial INVITE request, the I-MGCF shall include in the SIP 180 Ringing response the P-Early-Media header field indicating the backward early media is not authorised, except when: - the I-MGCF has already sent a reliable provisional response (see IETF RFC 3262 [54]) including a P-Early-Media header, as defined in IETF RFC 5009 [89]; and - the most recently sent P-Early-Media header field did not authorize the backward early media. If the P-Early-Media header field with the "supported" parameter was not included in the initial INVITE request, then as a network option the I-MGCF may include in the SIP 180 Ringing response the P-Early-Media header field indicating the backward early media is not authorised for the "ICS call".
7.2.3.1.4A	Sending of 183 Session Progress for early media scenarios	Modify the text in the paragraph which begins "For a speech call, which is not identified…." as follows: ~~For a speech call, which is not identified as an "ICS call" as specified in clause 7.2.3.1.2.12, u~~Upon receipt of one of the following messages and if the I-MGCF has received the P-Early-Media header field in the INVITE request~~, and has not already sent a provisional response including a P-Early-Media header field with parameters indicating authorization of early media with the same directionality as determined by table 7.2.3.1.4A.1 or table 7.2.3.1.4A.2,~~ then the I-MGCF shall send the 183 Session Progress response with a P-Early-Media header field authorizing early backward media as indicated: - ACM with the value of the called party's status indicator "no indication" ~~and one of the options described in table~~

~~7.2.3.1.4A.1~~. If the I-MGCF supports the PSTN XML body as a network option, the I-MGCF shall map parameters within the ACM into the PSTN XML body within the 183 as indicated in table 7.2.3.1.4A.1. ~~Based on local configuration, the I-MGCF may also send a 183 Session Progress response with a P-Early-Media header field authorizing early media if it receives an ACM with other parameters than described in table 7.2.3.1.4A.1.~~

Add the following immediately above Table 7.2.3.1.4A.1:

"UK Note: In the UK this table is not used to decide when to send a 183. It has been retained solely for the mapping of ACM parameter indicators to pstn-xml."

Replace Table 7.2.3.1.4A.1 with the following:

←183 Session Progress	←ACM
PSTN XML with ProgressIndicator with "Progress Description" value No. 8 ("In-band information or appropriate pattern is now available") (NOTE 1)	Optional backward call indicators parameter In-band information indicator 1 "In-band info or an appropriate pattern is now available"
PSTN XML with ProgressIndicator with "Progress Description" value No. 1 ("Call is not end-to-end ISDN: further call progress information may be available in-band") (NOTE 1)	Backward call indicators parameter ISDN User Part indicator 0 "ISDN User Part not used all the way"
PSTN XML with ProgressIndicator with "Progress Description" value No. 2 ("Destination address is non-ISDN") (NOTE 1)	Backward call indicators parameter ISDN User Part indicator 1 "ISDN User Part used all the way" ISDN access indicator 0 "Terminating access non-ISDN"
PSTN XML with ProgressIndicator with "Progress Description" value No. 7 ("Terminating access ISDN") (NOTE 1)	Backward call indicators parameter ISDN User Part indicator 1 "ISDN User Part used all the way" ISDN access indicator 1 "Terminating access ISDN"
NOTE 1: The ProgressIndicator "Coding Standard" parameter shall be set to "00 (ITU-T standardized coding)". The default value for the ProgressIndicator "Location" parameter is "0011 (Transit Network)".	

Delete Note 1 from under Table 7.2.3.1.4A.1

Modify the text which begins "- CPG message, when:…" as follows:

“- CPG message, when:

1. Event indicator is set to "in-band information or an appropriate pattern is now available", or
2. Event indicator is set to "Progress"~~ and one of the options described in table 7.2.3.1.4A.2~~.”

Add the following text immediately above Table 7.2.3.1.4A.2:

“UK Note: In the UK this table is not used to decide when to send a 183. It has been retained solely for the mapping of CPG parameter indicators to pstn-xml.”

Replace Table 7.2.3.1.4A.2 with the following:

<table>
<tr><th>←183 Session Progress</th><th>←CPG</th></tr>
<tr><td>PSTN XML with ProgressIndicator with "Progress Description" value No. 8 ("In-band information or appropriate pattern is now available") (NOTE 3)</td><td>Event indicator
000 0010 (progress)

Optional backward call indicators parameter
In-band information indicator
1 "In-band info or an appropriate pattern is now available"</td></tr>
<tr><td>PSTN XML with ProgressIndicator with "Progress Description" value No. 1 (Call is not end-to-end ISDN: further progress information may be available in-band") (NOTE 3)</td><td>Event indicator
000 0010 (progress)

Backward call indicators parameter
ISDN User Part indicator
0 "ISDN User Part not used all the way"</td></tr>
<tr><td>PSTN XML with ProgressIndicator with "Progress Description" value No. 2 ("Destination address is non-ISDN") (NOTE 3)</td><td>Backward call indicators parameter
ISDN User Part indicator
1 "ISDN User Part used all the way"

ISDN access indicator
0 "Terminating access non-ISDN"</td></tr>
<tr><td>PSTN XML with ProgressIndicator with "Progress Description" value No. 7 ("Terminating access ISDN") (NOTE 3)</td><td>Backward call indicators parameter
ISDN User Part indicator
1 "ISDN User Part used all the way"
ISDN access indicator
1 "Terminating access ISDN"</td></tr>
<tr><td colspan="2">NOTE 1: The mapping of the contents in the CPG message is only relevant if the information received in the message is different compared to earlier received information, e.g., in the ACM message or a CPG message received prior to this message.
NOTE 2: 183 Session Progress message including a P-Early-Media header authorizing early media may only be sent for a speech call.
NOTE 3: The ProgressIndicator "Coding Standard" parameter shall be set to "00 (ITU-T standardized coding)". The default value for the ProgressIndicator "Location" parameter is "0011 (Transit Network)".</td></tr>
</table>

Delete Note 2 from under Table 7.2.3.1.4A.2.

Modify the text in the paragraph which begins “If the INVITE request doesn’t include…” as follows:

		“If the INVITE request doesn't include the P-Early-Media header field with the "supported" parameter, then as a network option the I-MGCF may include in the SIP 18~~0~~3 ~~Ringing~~ Session Progress response a P-Early-Media header field authorizing early media.” Insert the following text below the paragraph which begins “As a network option, an I-MGCF may generate a Call-Info header field…” “UK Note: If selected, this network option provides for an I-MGCF to indicate an optional alternative to the in-band media from the PSTN. In any case, the I-MGCF must ensure that the in-band media is made available through the procedures specified in the rest of this clause.”
7.2.3.1.4B	Sending of 181 Call is being forwarded	Modify The text which begins “- ACM with call diversion information…” as follows: “ - ACM with call diversion information not indicating that presentation is not allowed~~ and optional backward call indicators indicate that in-band information is available~~.” Modify The text which begins “- CPG with call diversion information…” as follows: “- CPG with call diversion information not indicating that presentation is not allowed~~ and optional backward call indicators indicate that in-band information is available~~.” Modify the text in the paragraph which begins “For a speech call…” as follows: “~~For a speech call, and i~~If the INVITE request includes the P-Early-Media header, the I-MGCF shall include in the SIP 181 Call is being forwarded response a P-Early-Media header authorizing early media, except when” Modify the text in the paragraph which begins “If the INVITE request doesn't include …” as follows: “If the INVITE request doesn't include the P-Early-Media header field with the "supported" parameter, then as a network option the I-MGCF may include in the SIP 18~~0~~1 Call is being forwarded~~Ringing~~ response a P-Early-Media header field authorizing early media.”
7.2.3.1.4C	Sending of 183 Session Progress for overlap signalling using the in-dialog method	Void the section
7.2.3.1.4D	Sending of 183 Session Progress to carry ISUP Cause	Void the section
7.2.3.1.4E	Sending of 183 Session Progress for ICS call	Void the section
7.2.3.1.5	Sending of the 200 OK (INVITE)	Add the following text under the paragraph which begins “On receipt of an ANM/CON message containing the ATP…” “UK NOTE: The preceding two paragraphs are linked and only apply where the I-MGCF supports the PSTN XML body as a network option. The first paragraph describes the mapping from ISUP ATP to PSTN XML when a TMU parameter is received

		and the second paragraph describes a specific case for setting of the BearerCapability element of PSTN XML when no TMU is received.”

7.2.3.1.8	Receipt of the Release Message	Replace Table 9 with Table 9 below:

Table 9: Receipt of the Release message (REL)

←SIP Message Status code	← REL Cause indicators parameter
404 Not Found	Cause value No 1 Unallocated (unassigned) number
404 Not Found	Cause value No 2 No route to specified transit network (national use)
404 Not Found	Cause value No 3 No route to destination
604 Does Not Exist Anywhere	Cause value No 4 Send special information tone
404 Not Found	Cause value No 5 Misdialled trunk prefix (national use)
480 Temporarily Unavailable	Cause value No 8 Preemption
480 Temporarily Unavailable	Cause value No 9 Preemption - circuit reserved for reuse
410 Gone	Cause value No 14 QoR: ported number
480 Temporarily Unavailable	Cause value No 16 Normal call clearing
600 Busy Everywhere	Cause value No 17 User busy
408 Request Timeout	Cause value No 18 No user responding
480 Temporarily Unavailable	Cause value No 19 No answer from user (user alerted)
480 Temporarily Unavailable	Cause value No 20 Subscriber absent
603 Decline	Cause value No 21 Call rejected
410 Gone	Cause value No 22 Number changed
302 Moved Temporarily	Cause value No 23 Redirection to new destination
433 Anonymity Disallowed (NOTE 1)	Cause value No 24 Call rejected due to ACR supplementary service
483 Too Many Hops	Cause value No 25 Exchange Routing Error
480 Temporarily unavailable	Cause value No 26 (Non-selected user clearing)
480 Temporarily Unavailable	Cause value No 27 Destination out of order
484 Address Incomplete	Cause value No 28 Invalid number format (address incomplete)
403 Forbidden	Cause value No 29 Facility rejected
480 Temporarily Unavailable	Cause value No 31 Normal unspecified
486 Busy Here/ 600 Busy Everywhere. (Determined by Location value. "User " = 600, others = 486)	Cause value No 34 No circuit/channel available.
500 Server Internal Error	Cause value No 38 Network out of order
500 Server Internal Error	Cause value No 41 Temporary failure
503 Service Unavailable	Cause value No 42 Switching equipment congestion
500 Server Internal Error	Cause value No 43 Access information discarded
500 Server Internal Error	Cause value No 44 Requested circuit/channel not available
500 Server Internal Error	Cause value No 46 Precedence call blocked
500 Server Internal Error	Cause value No 47 Resource unavailable, unspecified
403 Forbidden	Cause value No 50 Requested Facility Not Subscribed
403 Forbidden	Cause value No 53 Outgoing calls barred within CUG
403 Forbidden	Cause value No 55 Incoming calls barred within CUG
488 Not Acceptable Here	Cause value No 57 Bearer capability not authorized
403 Forbidden	Cause value No 58 Bearer capability not presently available
403 Forbidden	Cause value No 62 Inconsistency in designated outgoing access information and subscriber class
403 Forbidden	Cause value No 63 Service or option not available, unspecified
501 Not Implemented	Cause value No 65 Bearer capability not implemented
501 Not Implemented	Cause value No 69 Requested facility not implemented
488 Not Acceptable Here	Cause value No 70 Only restricted digital information bearer capability is available
501 Not Implemented	Cause value No 79 Service or option not implemented, unspecified
403 Forbidden	Cause value No 87 User not member of CUG
488 Not Acceptable Here	Cause value No 88 Incompatible destination
404 Not Found	Cause value No 90 Non-existent CUG
404 Not Found	Cause value No 91 Invalid transit network selection
502 Bad Gateway	Cause value No 95 Invalid message, unspecified
502 Bad Gateway	Cause value No 97 Message type non-existent or not implemented
501 Not Implemented	Cause value No 98 (Message not compatible with call state or message type non-existent or not implemented)
502 Bad Gateway	Cause value No 99 Information element / parameter non-existent or not implemented
504 Server Time-out	Cause value No 102 Recovery on timer expiry
502 Bad Gateway	Cause value No 103 Parameter non-existent or not implemented, passed on
502 Bad Gateway	Cause value No 110 Message with unrecognized parameter, discarded
502 Bad Gateway	Cause value No 111 Protocol error, unspecified
502 Bad Gateway	Cause value No 127 Interworking, unspecified
NOTE 1: Anonymity Disallowed, IETF RFC 5079 [77] refers NOTE 2: No Retry-After header field shall be included.	

7.2.3.1.11	Internal through connection of the bearer path	Modify the text as follows: ~~The through connection procedure is described in clause 9.2.2.3.5.~~ During call setup a bothway speech path shall be established. Both way speech shall be established on receipt of an ISUP ANM or CON message.

6.2 ISUP to SIP Interworking

The UK Transition diagram for ISUP to SIP is shown in Figure 5.2 below:

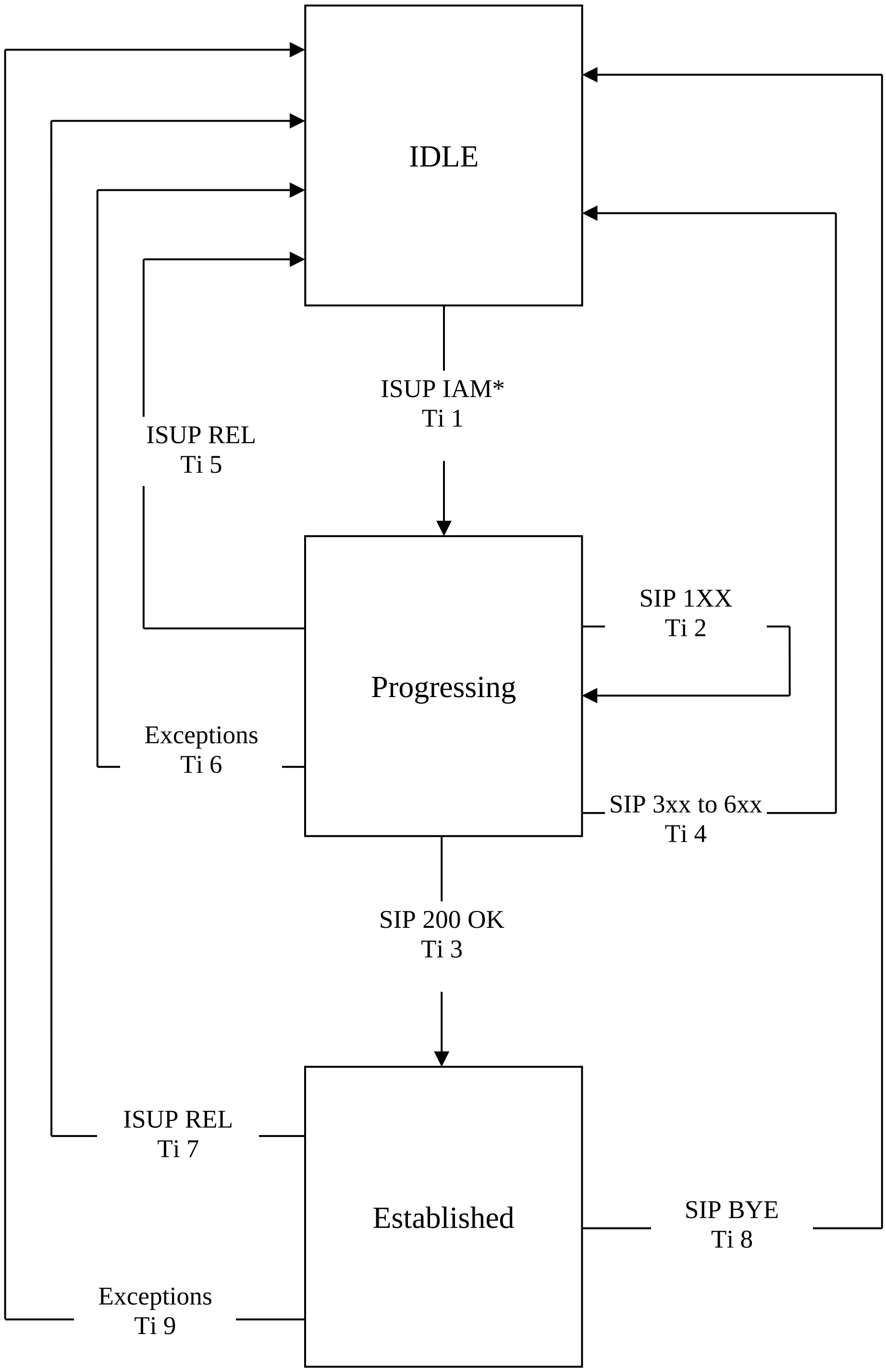

Figure 5.2 – ISUP to SIP Transition Diagram

The references relevant to Figure 6.2 are given in Table 6.2

Transition	Received Signalling Message	3GPP TS 29.163 references
Ti 1	ISUP IAM	7.2.3.2.1, 7.2.3.2.2
Ti 2	SIP 18X	7.2.3.2.4, 7.2.3.2.5, 7.2.3.2.6, 7.2.3.2.7
Ti 3	SIP 200 OK (INVITE)	7.2.3.2.7a, 7.2.3.2.7b, 7.2.3.2.8, 7.2.3.2.9, 7.2.3.2.10, 7.2.3.2.11
Ti 4	SIP 3xx/4xx/5xx/6xx	7.2.3.2.12, 7.2.3.2.17, 7.2.3.2.19
Ti 5	ISUP REL (before answer)	7.2.3.2.14
Ti 6	NONE (Nodal exceptions before answer)	7.2.3.2.15, 7.2.3.2.16
Ti 7	ISUP REL (after answer)	7.2.3.2.14
Ti 8	SIP BYE	7.2.3 2.13
Ti 9	NONE (Nodal exceptions after answer)	7.2.3.2.15, 7.2.3.2.16

Table 6.2 – TS 29.163 references for ISUP to SIP interworking

6.2.1 Endorsement of TS 29.163 Section 7.2.3.2 Outgoing Call Interworking from ISUP to SIP at O-MGCF

<table>
<tr><th>TS 29.163 section</th><th>Title</th><th>Comment</th></tr>
<tr><td>7.2.3.2.1.1</td><td>General</td><td>Insert the following text under the note:

“If an MGCF identifies an emergency call it shall construct a Resource-Priority header field as described in ND1035 [N1].

A call shall be identified as an emergency call if the ISUP CPC set to 11 is received.

Alternatively, a call may be identified as an emergency call if the called address contains an emergency call service address e.g. 999, 112.”</td></tr>
<tr><td>7.2.3.2.1.2</td><td>Interaction with continuity check</td><td>Void the section</td></tr>
<tr><td>7.2.3.2.1.3</td><td>IAM without calling party number</td><td>Void the section</td></tr>
<tr><td>7.2.3.2.1a</td><td>Sending of INVITE without determining the end of address signalling</td><td>Void the section</td></tr>
<tr><td>7.2.3.2.2.0</td><td>Overview</td><td>Replace Table 10aa with the table below:
<table>
<tr><th>IAM→</th><th>INVITE→</th></tr>
<tr><td>Called Party Number</td><td>Request-URI
To</td></tr>
<tr><td rowspan="3">Calling Party Number</td><td>P-Asserted-Identity</td></tr>
<tr><td>Privacy</td></tr>
<tr><td>From</td></tr>
<tr><td rowspan="2">Generic Number ("additional calling party number")</td><td>From</td></tr>
<tr><td>Privacy</td></tr>
<tr><td>Hop Counter</td><td>Max-Forwards</td></tr>
<tr><td>TMR/USI</td><td>Message Body (application/SDP)</td></tr>
<tr><td>Location Number
UK note: FFS</td><td>P-Access-Network-Info</td></tr>
</table>
</td></tr>
<tr><td>7.2.3.2.2.1</td><td>Request-URI and To header field</td><td>Replace Table 10a with the table below:</td></tr>
</table>

Table 10a: Mapping ISUP Called Party Number to SIP Request-URI and To header field

IAM		INVITE	
BICC/ISUP Parameter / field	**Value**	**SIP component**	**Value**
Called Party Number (NOTE 3)		Request-URI and To header field	addr-spec derived from Called Party Number parameter address signals. UK Note: the provision of the display-name field is not required
Nature of Address Indicator	"national (significant) number"	Tel URI or SIP URI	Insert "+44" before the Address signals
	"international number"		Insert "+" before the Address signals
	"UK specific number"		local number (with a phone-context = +44)
NOTE 3: If the address signals received in the ISUP Called Party Number contain a sending terminated signal (hexadecimal digit F), then this shall be discarded or if the O-MGCF supports the PSTN XML body as a network option then the PSTN XML sendingCompleteIndication shall be set.			

7.2.3.2.2.2	SDP Media Description	Replace paragraphs 1, 2 and the note with the following: "If the O-MGCF indicates support of the SIP preconditions in the initial INVITE request then it shall indicate whether preconditions are met dependent on the status of the local resource reservation. The media path used shall comply with ND1704 [N6]. Specifically, the guidance in ND1704 [N6] regarding the use of G.711 A-law as the default codec for interoperability shall apply. NOTE: RFC 3261 [N7] supports sending a 488 (Not Acceptable Here) response which, if agreed bilaterally, offers an alternative to including G.711 A-law in the initial INVITE." Replace Table 10b with the table below:

Table 10b: Coding of SDP media description lines from TMR/USI: ISUP to SIP

TMR parameter	**USI parameter (Optional)**		**HLC IE in ATP (Optional)**	**m= line**			**b= line**	**a= line**
TMR codes	Information Transfer-Capability	User Information Layer 1 Protocol Indicator	High Layer Characteristics Identification	<media>	<transport>	<fmt-list>	<modifier>: <bandwidth-value>	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>
"speech"	"Speech"	"G.711 A-law"	Ignore	audio	RTP/AVP	8	AS: (64 + RTP/UDP/IP overhead)	rtpmap:8 PCMA/8000
"speech"	"Speech"	"G.711 A-law"	Ignore	audio	RTP/AVP	Dynamic PT	AS: (64 + RTP/UDP/IP overhead)	rtpmap:<dynamic-PT> PCMA/8000
"3.1 KHz audio"	USI Absent		Ignore	audio	RTP/AVP	8	AS: (64 + RTP/UDP/IP overhead)	rtpmap:8 PCMA/8000
"3.1 KHz audio"	"3.1 KHz audio"	"G.711 A-law"	(NOTE 3)	audio	RTP/AVP	8	AS: (64 + RTP/UDP/IP overhead)	rtpmap:8 PCMA/8000
"3.1 KHz audio"	"3.1 KHz audio"		"Facsimile Group 2/3"	image (NOTE 9)	Udptl [73]	t38[73]	AS: (64 + UDP/IP overhead)	Based on ITU-T T.38 [72]. (NOTE 8)
"3.1 KHz audio"	"3.1 KHz audio"		"Facsimile Group 2/3"	image (NOTE 9)	Tcp (NOTE 7)	t38[73]	AS: (64 + TCP/IP overhead)	Based on ITU-T T.38 [72]. (NOTE 8)
"3.1 KHz audio"	"3.1 KHz audio"		"Facsimile Group 2/3"	Audio (NOTE 9)	RTP/AVP	8	AS: (64 + RTP/UDP/IP overhead)	rtpmap:8 PCMU/8000
"64 kbit/s preferred"	"Speech/ 3.1KHz audio" (NOTE 6)	N/A	Ignore	audio	RTP/AVP	Dynamic PT	AS: (64 + RTP/UDP/IP overhead)	rtpmap:<dynamic-PT> CLEARMODE/8000 (NOTE 2)(NOTE 4)
"64 kbit/s unrestricted"	"Unrestricted digital information"	N/A	Ignore	audio	RTP/AVP	Dynamic PT	AS: (64 + RTP/UDP/IP overhead)	rtpmap:<dynamic-PT> CLEARMODE/8000 (NOTE 2)(NOTE 5)

NOTE 2: CLEARMODE is specified in IETF RFC 4040 [69].

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

NOTE 4: After the CLEARMODE codec, additional speech codecs such as AMR and/or G.722 and/or G.711 available via transcoding or reframing should be offered in the same m-line.

NOTE 5: As alternative or in addition to the m-line containing the CLEARMODE codec, an MGCF supporting the multimedia interworking detailed in Annex E may add an m-line for speech codecs and an m-line for video codecs as detailed in this Annex.

NOTE 6: In this case, the USI prime parameter will also be present and will indicate "Unrestricted Digital Information with tones/announcements".

NOTE 7: This value is not recommended in a network supporting MTSI clients as it is not supported by an MTSI client (see TS 24.173 [88]).

NOTE 8: Annex K describes recommended values.

NOTE 9: FAX can either be transported according to ITU-T recommendation T.38 [72] using the "image" media type or as inband Voice band data over IP using the "audio" media type.

TMR parameter	USI parameter (Optional)		HLC IE in ATP (Optional)	m= line			b= line	a= line
TMR codes	Information Transfer-Capability	User Information Layer 1 Protocol Indicator	High Layer Characteristics Identification	<media>	<transport>	<fmt-list>	<modifier>:<bandwidth-value>	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>
Use of the G.711 A-law codec is recommended for transmission of fax information. When required, support for ITU-T Rec. T.38 [72] Fax transport shall be indicated in the SDP offer per RFC 3362 [73]. NOTE: Calls encountering multiple T.38 hops (.e.g. due to multiple TDM<->IP interworkings) have been shown to suffer quality degradation.								

7.2.3.2.2.3	P-Asserted-Identity, From and Privacy header fields	Insert the following text immediately under the section title; "In accordance with Ofcom General Condition C.6 and ND1016 [xx] the ISUP parameters received by the MGCF that would be used for mapping should contain valid CLI. Calls with invalid CLI information should be rejected within the UK TDM domain." Replace Tables 12, 13, 14 & 15 with the Tables 12, 13, 14 & 15 below Delete Table 16.

Table 12 - Mapping ISUP Calling Party Number parameter to SIP P-Asserted-Identity header field

Has a Calling Party Number parameter been received with a complete E.164 number, with SI = UPVP or NP?	Has a Calling Party Number parameter with APRI = "presentation restricted by the network", or a NFCI parameter with CBI = 0 been received?	P-Asserted-Identity header field	Privacy header
N (NOTE 2)	-	Inject a network number CLI or Header field not included	If a network number has been injected then include priv-value "id".
Y	N	Map the Calling Party Number parameter to the P-Asserted-Identity header field (see Table 14 for formatting) (Note 1)	See Table 15
Y	Y	Map the Calling Party Number parameter to the P-Asserted-Identity header field (see Table 14 for formatting) (Note 1)	Include priv-value "id".
NOTE 1: The presence of a display-name field causes possible presentation conflicts with the URI so the field should not be provided by the MGCF.			
NOTE 2: In the case of a non-emergency call the Calling Party Number parameter received by the IWU should have been validated previously within the UK TDM domain so is assumed to be an E.164 number			

Table 13 - Mapping 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 and PNP = "PN preferred for mapping to legacy (IUP) ISDN services"?	Has a Generic Number parameter been received with Number Qualifier "*additional calling party number*" and a complete E.164 number, with SI = "UPNV"?	Has a NFCI parameter been received with CBI = 0? (NOTE 4)	Has a Calling Party Number parameter been received with a complete E.164 number, with SI = UPVP or NP?	CgPN APRI	"From" header field
Y	-	-	Y	-	Map the Presentation Number parameter to the "From" header field URI (see Table 14 for formatting) (NOTE 2)
Y (NOTE 6)	-	-	N (NOTE 5)	-	Insert a sip URI with the user field containing 'unavailable'
N	Y	-	Y	-	Map the Generic Number parameter to the "From" header field URI (see Table 14 for formatting) (NOTE 2)
N (NOTE 6)	Y	-	N (NOTE 5)	-	Insert a sip URI with the user field containing 'unavailable'
N (NOTE 6)	N	Y	-	-	Insert a sip URI with the user field containing 'unavailable'
N (NOTE 6)	N	N	N	-	Insert a sip URI with the user field containing 'unavailable'
N	N	N	Y	"Presentation allowed"	Map the CgPN parameter to the "From" header field URI (see Table 14 for formatting) (NOTE 2) (NOTE 4)
N	N	N	Y	"Presentation restricted" / "Address not available" (NOTE 1)	Insert a sip URI with the user field containing 'anonymous'
N (NOTE 6)	N	N	Y	"Restricted by the network"	Insert a sip URI with the user field containing 'unavailable' (NOTE 3)
NOTE 1: The APRI value "Not Available" has been included in this table. However, this value is defined for national use in Q.763 and should not normally be received.					
NOTE 2: The presence of a display-name field causes possible presentation conflicts with the URI so the field should not be provided by the IWU.					
NOTE 3: A network provided CLI in the CgPN parameter may occur on a call from any type of access line. Therefore, in order to allow the "display" of this network provided CLI at a SIP UAS it must be mapped into the SIP From header.					
NOTE 4: The NFCI CBI indicator value of '0' is only valid where Presentation Number or Generic Number parameter is present.					
NOTE 5: The presence of a Presentation Number and/or Generic Number parameter is only valid if a CgPN parameter was received containing CLI information that could be mapped to a P-Asserted-Identity header field. Mapping does not take place where this is not the case.					
NOTE 6: This mapping option is only allowed for emergency calls					

Table 14 - Mapping of ISUP Line Identity parameter fields to SIP header fields

BICC/ISUP parameter field		SIP header field (Note)
Nature of Address Indicator	**Address Signals**	**Addr-spec**
"national (significant) number"	“NDC” + “SN”	Prefix "+" CC (of the country where the IWU is logically located) to the address signals, then map [as “+”+ “CC” + “NDC” + “SN”] to the URI scheme used.
"international number"	“CC” + “NDC” + “SN”	Prefix "+" to the address signals, then map [as “+”+ “CC” + “NDC” + “SN”] to the URI scheme used.
Note: The presence of a display-name field causes possible presentation conflicts with the URI so the field should not be provided by the MGCF.		

Table 15 - Mapping of ISUP APRI values into SIP Privacy header field

BICC/ISUP Parameter	APRI Value	Privacy header field (Note 1)
Presentation Number (when used for the population of the “From” header field – see Table 13)	“Presentation allowed”	Do not include priv-value ‘user’.
	“Presentation restricted”	include the priv-value ‘user’. (NOTE 3)
Generic Number (when used for the population of the “From” header field – see Table 13)	“Presentation allowed”	Do not include priv-value ‘user’.
	“Presentation restricted”	include the priv-value ‘user’. (NOTE 3)
Calling Party Number	“Presentation allowed”	Do not include priv-value ‘id’ or ‘header’.
	“Presentation restricted” / “Address not available” (Note 2)	include the value ‘id’, and as a network option ‘header’.
	“Presentation restricted by the network”	include the value ‘id’, and as a network option ‘header’.
NOTE 1: After the mappings in this table have been performed if the result is that no priv-value has to be sent then the Privacy header field is omitted.		
NOTE 2: The APRI value “Not Available” has been included in this table. However, this value is defined for national use in Q.763 and should not normally be received.		
NOTE 3: As a network option the “From” header field can be set to an anonymous URI with the Privacy header field not containing priv-value ‘user’.		

7.2.3.2.2.3A	"cpc" URI Parameter in P-Asserted-Identity Header	Void the section
7.2.3.2.2.3B	"oli" URI Parameter in P-Asserted-Identity Header	Void the section
7.2.3.2.2.4	Max Forwards header	Replace the paragraph immediately under Table 17 with the following: “The factor used in the calculation shall equal ‘2’. 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.”
7.2.3.2.2.6	P-Early-Media header field	Replace ‘may’ with ‘shall’ in the first paragraph.
7.2.3.2.2A	Coding of the INVITE when Number Portability is supported	Void the section
7.2.3.2.2B	Coding of the INVITE for Carrier Routeing	Void the section
7.2.3.2.2C	Coding of INVITE with instance-id in form of IMEI URN	Void the section
7.2.3.2.2.10	PSAP Call-back indication	Void the section
7.2.3.2.3	Receipt of CONTINUITY	Void the section
7.2.3.2.5.1	Backward call indicators	Modify the text immediately under ‘Bit I’ as follows: “0 no interworking encountered (SS7 used all of the way)”
7.2.3.2.7b	Internal through connection of the bearer path	Replace the text with the following: “During call setup a both way through connection shall be established. Both way speech shall be established on sending of an ISUP ANM or CON message.”
7.2.3.2.8	Sending of the Answer Message (ANM)	Delete the note. Insert the following text under paragraph 1: “The MGCF shall stop the awaiting answer indication (if previously applied) and establish a both way media connection.”
7.2.3.2.10	Sending of the Connect message (CON)	Insert the following text under paragraph 1: “The MGCF shall stop the awaiting answer indication (if previously applied) and establish a both way media connection.”
7.2.3.2.12	Receipt of Status Codes 4xx, 5xx or 6xx	Replace table 18 with the table below:

Table 18: 4xx/5xx/6xx Received on SIP side of O-MGCF

←REL		←4xx/5xx/6xx SIP Message (NOTE 4)
(location value)	**(cause value)**	
(BI)	95 (Invalid message, unspecified)	400 Bad Request
(BI)	63 Service or option not available, unspecified	401 Unauthorized
(BI)	63 Service or option not available, unspecified	402 Payment Required
(BI)	63 Service or option not available, unspecified	403 Forbidden
(BI)	1 (Unallocated (unassigned) number)	404 Not Found
(BI)	63 Service or option not available, unspecified	405 Method Not Allowed
(BI)	79 (Service or option not implemented, unspecified)	406 Not Acceptable
(BI)	63 Service or option not available, unspecified	407 Proxy authentication required
(BI)	18 No User Responding	408 Request Timeout
(BI)	22 (Number changed)	410 Gone
(BI)	111 (Protocol error, unspecified)	413 Request Entity too long
(BI)	111 (Protocol error, unspecified)	414 Request-URI too long
(BI)	79 (Service or option not implemented, unspecified)	415 Unsupported Media type
(BI)	127 (Interworking, unspecified)	416 Unsupported URI scheme
(BI)	79 (Service or option not implemented, unspecified)	417 Unknown Resource-Priority
(BI)	79 (Service or option not implemented, unspecified)	420 Bad Extension
(BI)	79 (Service or option not implemented, unspecified)	421 Extension required
(BI)	31 (Normal, unspecified)	422 Session Interval Too Small
(BI)	63 Service or option not available, unspecified	423 Interval Too Brief
(User)	24 (Call rejected due to feature at the destination)	433 Anonymity Disallowed (NOTE 1)
(BI)	127 (Interworking, unspecified)	440 Max-Breadth Exceeded
(User)	31 (Normal, unspecified)	480 Temporarily Unavailable
(BI)	95 (Invalid message, unspecified)	481 Call/Transaction does not exist
(BI)	25 (Exchange routing error)	482 Loop detected
(BI)	25 (Exchange routing error)	483 Too many hops
(BI)	28 (Invalid number format (address incomplete))	484 Address Incomplete
(BI)	1 (Unallocated (unassigned) number)	485 Ambiguous
(BI)	17 User Busy or 34 No circuit/channel available as a network option for consistency with ND1017	486 Busy Here
(BI)	31 (Normal, unspecified) (NOTE 2)	487 Request terminated
(BI)	79 (Service or option not implemented, unspecified)	488 Not acceptable here
	No mapping	491 Request Pending
(BI)	127 (Interworking, unspecified)	493 Undecipherable
(BI)	47 Resource unavailable, unspecified	500 Server Internal error
(BI)	79 (Service or option not implemented, unspecified)	501 Not implemented
(BI)	111 (Protocol error, unspecified)	502 Bad Gateway
(BI)	42 Switching equipment congestion	503 Service Unavailable
(BI)	102 (Recovery on timer expiry)	504 Server timeout
(BI)	127 (Interworking, unspecified)	505 Version not supported
(BI)	111 (Protocol error, unspecified)	513 Message too large
(TN)	34 No circuit/channel available	580 Precondition failure
(User)	17 (User busy)	600 Busy Everywhere
(User)	21 (Call rejected)	603 Decline
(User)	4 Send special information tone	604 Does not exist anywhere
(BI)	79 (Service or option not implemented, unspecified)	606 Not Acceptable
	NOTE 1: Anonymity Disallowed, IETF RFC 5079 [77] refers. NOTE 2: No interworking if the O-MGCF previously issued a CANCEL request for the INVITE. NOTE 3: The 4xx/5xx/6xx SIP responses that are not covered in this table are not interworked. NOTE 4: Receipt of any SIP Status code which is not present in this table shall be mapped to ISUP REL Cause 31, Location BI	

7.2.3.2.12.1	Special handling of 404 Not Found and 484 Address Incomplete	Void the section

	responses after sending of INVITE without determining the end of address signalling	
7.2.3.2.18	Sending of CANCEL	Void the section
7.2.3.2.20	Sending of INFO for overlap signalling using the in-dialog method	Void the section
7.2.3.3	Timers	Replace Table 19 with the Table 19 below:

Table 19: Timers for interworking

Symbol	Time-out value	Cause for initiation	Normal termination	At expiry	Reference
Ti/w1	4 s to 6 s (default of 4 s)	When last address message is received and the minimum number of digits required for routing the call have been received.	At the receipt of fresh address information.	Send INVITE, send the address complete message	7.2.3.2.1 7.2.3.2.4 (NOTE 1)
Ti/w2	4 s to 20 s (default of 4 s)	When INVITE is sent unless the ACM has already been sent.	On reception of any of the following: • 180 Ringing, or • 183 Session Progress (and, when supported, preconditions have been met), or • 181 Call is Being Forwarded, or • 200 OK (INVITE).	Send ACM (no indication)	7.2.3.2.4 7.2.3.2.1 (NOTE 2)
NOTE 1: This timer is used when overlap signalling is received from BICC/ISUP network and converted to en-block signalling at the MGCF. NOTE 2: This timer is used to send an early ACM if a delay is encountered in receiving a response from the subsequent SIP network.					

7 Supplementary Services

Sections 7.1 and 7.2 describe interworking with ISDN and IMS supplementary services respectively through endorsement of TS 29.163 sections 7.4 and 7.5.

7. 1 Endorsement of TS 29.163 Section 7.4 CS Supplementary Services

TS 29.163 section	**Title**	**Comment**
7.4.1	Calling line identification presentation/restriction (CLIP/CLIR)	Insert the following text under paragraph 3: "As SIP does not implement an equivalent of the ISUP screening indicator when interworking the SIP From header to ISUP Generic Number the Screening indicator is set to User Provided Not Verified by default as described in table 6 in clause 7.2.3.1.2.6."
7.4.2	Connected line presentation and restriction (COLP/COLR)	Insert the following text: "UK: For Further Study"
7.4.5	Subaddressing (SUB)	Void the section.
7.4.8	Explicit Call Transfer (ECT)	Insert the following text as paragraph 1: "UK: For Further Study"
7.4.10.1	Session hold initiated from the IM CN subsystem side	Insert the following text under figure 30a: "As a UK option, networks may choose to handle the SIP signalling hold/resume as described but perform no interworking to ISUP."
7.4.10.2	Session hold initiated from the CS network side	Insert the following text under figure 30b: "As a UK option, networks may choose to handle the SIP signalling hold/resume as described but perform no interworking to ISUP."
7.4.11	Call Completion on busy subscriber	Insert the following text as paragraph 1: "UK: For Further Study"
7.4.12	Completion of Calls on No Reply (CCNR)	Insert the following text as paragraph 1: "UK: For Further Study"
7.4.14	Conference calling (CONF) / Three-Party Service (3PTY)	Insert the following text as paragraph 1: "UK: For Further Study. It is not clear that there are any UK implementations that use ISDN or IMS implementations of network conferencing."
7.4.17	Multi-Level Precedence and Pre-emption (MLPP)	Insert the following text as paragraph 1: "UK: For Further Study"
7.4.18	Global Virtual Network Service (GVNS)	Insert the following text as paragraph 1: "UK: For Further Study"
7.4.21	User-to-User Signalling (UUS)	Void the section.

7. 2 Endorsement of TS 29.163 Section 7.5 IMS Supplementary Services

TS 29.163 section	**Title**	**Comment**
7.5.2	Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR)	Insert the following text: "UK: For Further Study"
7.5.6	Conference call (CONF)	Insert the following text: "UK: For Further Study. It is not clear that there are any UK implementations that use ISDN or IMS implementations of network conferencing."
7.5.8	Message Waiting Indication (MWI)	Insert the following text below the existing text: "UK NOTE: There is no interworking of SIP MWI to UK-ISUP."
7.5.10	Closed User Group (CUG)	Void the section
7.5.11	CCBS/CCNR	Insert the following text: "UK: For Further Study"
7.5.12	Communication Waiting (CW)	Void the section

History

Document history		
Version	Date	Milestone
1.1.1	4th May 2022	Initial Publication