3GPP TSG CN Plenary Meeting #18 **NP-020631**
4th - 6th December 2002. New Orleans, USA.

Source:	**LM Ericsson**
Title:	**A proposal for baseline SIP-BICC/ISUP interworking text**
Agenda item:	**9.1 IMS phase 2**
Document for:	**INFORMATION**

At the last SG 11 meeting in Geneva it was decided that the Q.1912.SIP document is kept as one document, which includes all profiles. The discussion was triggered by a Liason Statement from 3GPP CN 3, a contribution from Vodafone and a contribution from Ericsson.

The Ericsson proposal included a baseline text, which only deals with Profile A. Profile A (3GGP profile) is defined in TS 24.229. From a technical point 95% of the contribution was accepted. However, with the present structure of the document it is hard to identify all points.

Therefore, we think that the annex, which is the rest of this document and based on the Ericsson contribution, mentioned above, could be a good starting point for the work of defining a SIP-BICC/ISUP interworking specification in CN 3 (TS 29.163).

Note: The annex is just added for informational purpose.

1 Scope

This Recommendation defines the signalling interworking between the Bearer Independent Call Control (BICC) or ISDN User Part (ISUP) protocols and Session Initiation Protocol (SIP) with its associated Session Description Protocol (SDP) at an Interworking Functional entity. The services that can be supported through the use of the signalling interworking are limited to the services that are supported by BICC or ISUP and SIP based network domains.
ISUP is defined in accordance with Q.761 to Q.764 and BICC is defined in accordance with Q.1902.1 to Q.1902.4. BICC is the call control protocol used between "Serving Nodes" in a network that incorporates separate call and bearer control. An Interface Serving Node (ISN) provides the interface between BICC network domains and non-BICC network domains.
The BICC/ISUP capabilities or signalling information defined for national use is outside the scope of this Recommendation. It does not imply interworking for national-specific capabilities is not feasible.
SIP and SDP are defined by the IETF. The capabilities of SIP and SDP that are interworking with BICC or ISUP are defined in 3GPP – Technical Specification Group Core Network IP Multimedia Call Control Based on SIP and SDP," Stage 3 – Revision 5, 3GPP TS 24.229 V5.1.0 (2002-06).
Services that are common in SIP and BICC or ISUP network domains will seamlessly interwork by using the function of an Interworking Unit. The Interworking Unit will originate and/or terminate services or capabilities that do not interwork seamlessly across domains according to the relevant protocol recommendation or specification.
Table 1 lists the services seamlessly interworked and therefore within the scope of this recommendation.

Table 1- Interworking Capabilities between BICC/ISUP and SIP profile for 3GPP

Service
Speech/3.1 kHz audio
En bloc address signalling
Out of band transport of DTMF tones and information. (BICC only)
Direct-Dialling-In (DDI)
Multiple Subscriber Number (MSN)
Calling Line Identification Presentation (CLIP)
Calling Line Identification Restriction (CLIR)

2 References

3 Definitions

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

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

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

4 Abbreviations

5 Methodology

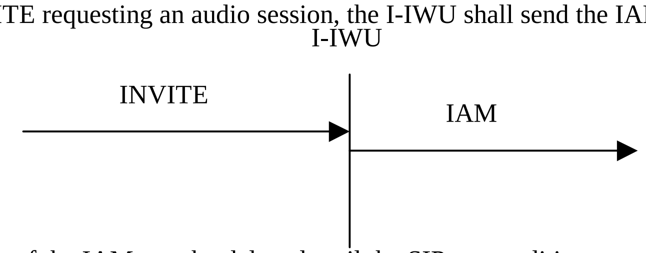
5.1 Conventions for Representation of BICC/ISUP PDU

5.2 Conventions for Representation of SIP/SDP Information

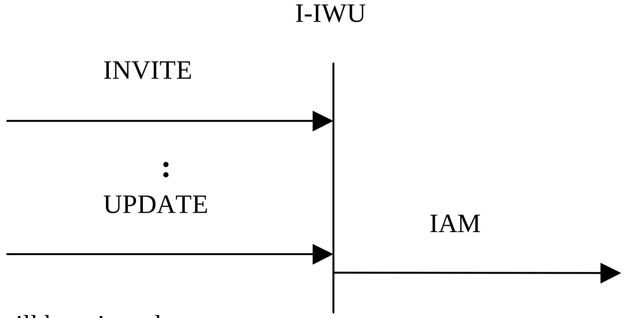
6 Incoming Call Interworking from SIP to BICC/ISUP at I-ISN

6.1 Sending of IAM

On reception of the INVITE requesting an audio session, the I-IWU shall send the IAM



Alternatively, the sending of the IAM may be delayed until the SIP preconditions are met:



Note: Non-audio sessions will be rejected

6.1.1 Coding of the IAM

6.1.1.1 Called Party Number

The information contained in the userinfo component of the Request-URI with user=phone shall be mapped to the called party number parameter of the IAM message.

Support of other URI schemas such as TEL: URI or SIPS-URI, URI component values and URI parameters is optional.

Table 2/Q.1912.SIP – Coding of the Called Party Number

INVITE→	IAM→
Request-URI	**Called Party Number**
Userinfo (SIP URI with user=phone)	Address Signal

6.1.1.2 Nature of connection indicators

bits **<u>BA</u>** **Satellite indicator**
0 1 *one satellite circuit in the connection*

bits **<u>DC</u>** **Continuity check indicator (ISUP)/Continuity indicator (BICC)**
0 0 *continuity check not required (ISUP) / no COT to be expected (BICC)* if IAM is delayed until the SIP preconditions have been met
1 0 *continuity check performed on a previous circuit (ISUP) / COT to be expected (BICC)* otherwise

bit **<u>E</u>** ***Echo control device indicator***
1 *outgoing echo control device included*

6.1.1.3 Forward call indicators

bits **<u>CB</u>** **End-to-end method indicator**
0 0 *no end-to-end method available (only link-by-link method available)*

bit **<u>D</u>** ***Interworking indicator*** **(Note 2)**

	1	*interworking encountered*
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*
bits	**HG**	**ISDN user part/BICC preference indicator**
	0 1	*ISDN user part/BICC not required all the way*
bit	**I**	**ISDN access indicator**
	0	*originating access non-ISDN*
bits	**KJ**	**SCCP method indicator (Note 2)**
	0 0	*no indication*

6.1.1.4 Calling party's category

0 0 0 0 1 0 1 0 ordinary calling subscriber

6.1.1.4 Transmission medium requirement

The TMR parameter is set to 3.1 kHz audio.[1]

6.1.3.5 Calling party number

Mapping of SIP From/P-Asserted-Identity/Privacy headers to BICC/ISUP CLI parameters

Has a “P-Asserted-Identity” header field containing a URI (Note 2) with an identity in the format “+CC”+”NCD”+”SN” been received?	Has a “From” header field (Note 3) containing a URI with an identity in the format “+CC”+”NCD”+”SN” been received?	Calling Party Number parameter Address signals	Calling Party Number parameter APRI	Generic Number (additional calling party number) address signals	Generic Number parameter APRI
No	No	Network option to either include a network provided E.164 number (See **Error! Reference source not found.**) or omit the CgPN parameter	Network option to set APRI to either “presentation restricted” or “presentation allowed” (See **Error! Reference source not found.**)	Parameter not included	Not applicable
No	Yes	Network Option to either include a network provided E.164 number (See **Error! Reference source not found.**) or omit the CgPN parameter	Network option to set APRI to either “presentation restricted” or “presentation allowed” (See **Error! Reference source not found.**)	Network Option to either derive from the “From” header or omit the parameter (See **Error! Reference source not found.**) (Note 1)	APRI = “presentation restricted” or “presentation allowed” depending on SIP Privacy header. (See **Error! Reference source not found.**)
Yes	No	Derive from P-Asserted-Identity (See **Error! Reference**	APRI = “presentation restricted” or	Not included	Not applicable

[1] Trans-coding equipment may be required.

Has a “P-Asserted-Identity” header field containing a URI (Note 2) with an identity in the format “+CC”+”NCD”+”SN” been received?	Has a “From” header field (Note 3) containing a URI with an identity in the format “+CC”+”NCD”+” SN” been received?	Calling Party Number parameter Address signals	Calling Party Number parameter APRI	Generic Number (additional calling party number) address signals	Generic Number parameter APRI
		source not found.)	“presentation allowed” depending on SIP Privacy header. (See **Error! Reference source not found.**)		
Yes	Yes	Derived from P-Asserted-Identity (See **Error! Reference source not found.**)	APRI = “presentation restricted” or “presentation allowed” depending on SIP Privacy header. (See **Error! Reference source not found.**)	Not included	Not applicable

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

Note 2: It is possible that the P-Asserted-Identity header field includes both a tel URI and a sip or sips URI. The handling of this case is for further study.

Note 3: The “From” header may contain an “Anonymous URI”. An “Anonymous URI” includes information that does not point to the calling party. RFC 3261 recommends that the display-name component contain "Anonymous". RFC [privacy] recommends that the Anonymous URI itself have the value "anonymous@anonymous.invalid".

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”*
Address signals	If NOA is *“national (significant) number”* no country code should be included. If NOA is “*international number”*, then the country code of the network-provided number should be included.

Mapping of P-Asserted-Identity and Privacy Headers to the ISUP/BICC Calling Party Number Parameter

SIP Component	Value	BICC/ISUP Parameter / field	Value
P-Asserted-Identity header field (Note 1)	name-addr / addr-spec	Calling Party Number	
		Number incomplete indicator	*"Complete"*
		Numbering Plan Indicator	*"ISDN/Telephony (E.164)"*
		Nature of Address Indicator	If +CC is equal to the CC of the country where ISN 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"*
		Address Presentation Restricted Indicator (APRI)	depends on priv-value in Privacy header.
		Screening indicator	Network Provided
addr-spec	"+CC" "NCD" "SN" from the URI	Address signal	if NOA is *"national (significant) number"* then set to "NCD" + "SN" If NOA is *"international number"* Then set to "CC"+"NCD"+"SN"
Privacy header field is not present		APRI	Presentation allowed
Privacy header field	priv-value	APRI	*"Address Presentation Restricted Indicator"*
priv-value	"header"	APRI	Presentation restricted
	"user"	APRI	Presentation restricted
	"none"	APRI	Presentation allowed
	"id" (Note 2)	APRI	Presentation restricted

Note 1: It is possible that a P-Asserted –Identity header field includes both a TEL URI and a sip or SIPS URI. The SIP URI is given priority.

6.1.3.6 Generic number

Mapping of SIP From Header Field to BICC/ISUP Generic Number (additional calling party number) parameter (Network option)

SIP Component	Value	BICC/ISUP Parameter / field	Value
From header field	name-addr or addr-spec	Generic Number Number Qualifier Indicator	*“Additional Calling Party number”*
from-spec	(name-addr / addr-spec)		
		Nature of Address Indicator	If +CC is equal to the CC of the country where IWU is located AND the next BICC/ISUP node is located in the same country then Set to *“national (significant) number”* Else set to *“international number”*
		Number incomplete indicator	*“Complete”*
		Numbering Plan Indicator	*“ISDN/Telephony (E.164)”*
		APRI	Depends on priv-value
		Screening indicator	*“user provided not verified”*
Addr-spec	“+CC” “NCD” “SN” from the URI	Address signal	if NOA is *“national (significant) number”* then set to “NCD” + “SN” If NOA is *“international number”* Then set to “CC”+”NCD”+”SN”
Privacy header field	priv-value	APRI	*"Address Presentation Restricted Indicator"*
Privacy header field is absent		APRI	*“presentation allowed”*
priv-value	"header	APRI	*“presentation restricted”*
	"user"	APRI	*“presentation restricted”*
	"none"	APRI	*“presentation allowed”*
	"id"	no mapping to Generic No	

6.1.3.8 User service information

See sub clause on TMR.

6.1.3.9 Hop counter

The I-IWU shall perform the following interworking procedure if Hop counter procedure is supported.

At the I-IWU the Max-Forwards SIP header shall be used to derive the Hop Counter parameter if applicable. Due to the different default values (that are based on network demands/provisions) of the SIP Max-Forwards header and the Hop Counter, a factor shall be used to adapt the Max Forwards to the Hop Counter at the I-IWU. For example, the following guidelines could be applied.

1. Max-Forwards for a given message should be monotone decreasing with each successive visit to a SIP entity, regardless of intervening interworking, and similarly for Hop Counter.
2. The initial and successively mapped values of Max-Forwards should be large enough to accommodate the maximum number of hops that might be expected of a validly routed call.

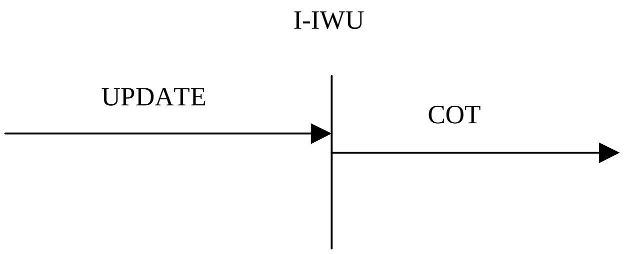
The following table shows the principle of the mapping:

Max-Forwards	= X	Hop Counter	= INTEGER part of (X /Factor)

Note: The Mapping of value X to Y should be done with the used (implemented) adaptation mechanism.

The Principle of adoption could be implemented on a basis of the network provision, trust domain rules and bilateral agreement.

6.2 Sending of COT



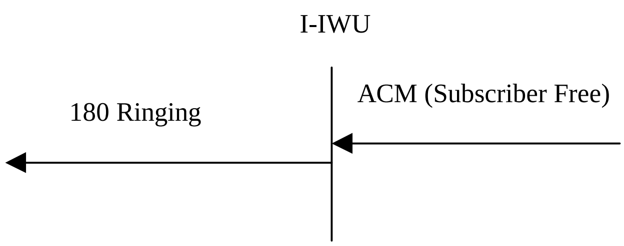
When the requested preconditions in the SIP network have been met and the IAM has already been sent, then the Continuity message is sent indicating successful

Note: If no UPDATE is received a Continuity message is sent indicating failure as a result of normal ISUP/BICC procedure.

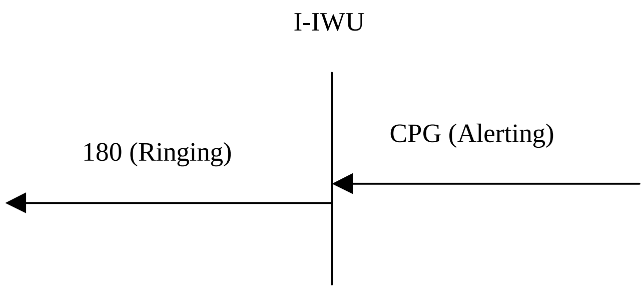
6.3 Sending of 180 Ringing

The following cases are possible trigger conditions for sending the 180 Ringing:

- The reception of ACM with **Called party's status indicator** set to *subscriber free*



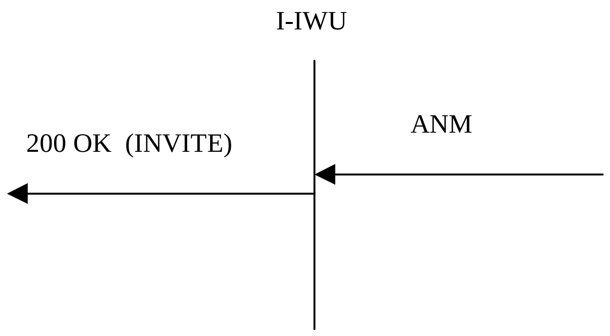
- The reception of CPG with Event information **Event indicator** set to *alerting*



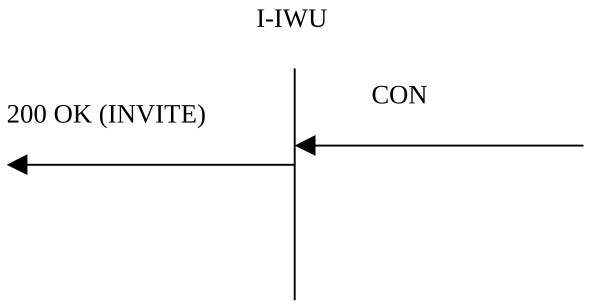
6.4 Sending of the 200 OK (INVITE)

The following cases are possible trigger conditions for sending the 200 OK (INVITE):

- The reception of the ANM



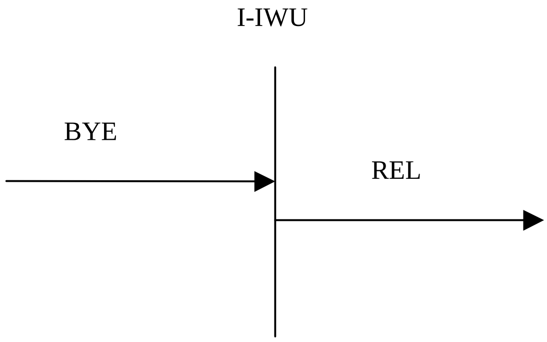
- The reception of the CON message



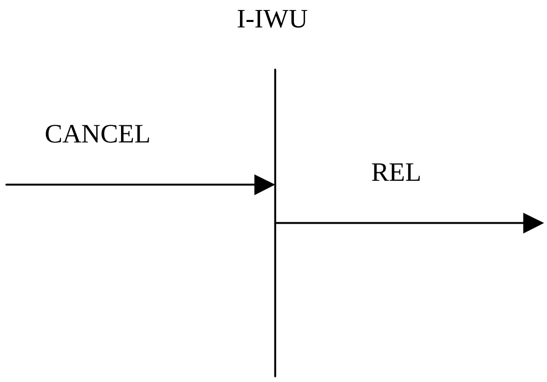
6.5 Sending of the Release message (REL)

The following are possible triggers for sending the Release message:

1. Receipt of the BYE method



2. Receipt of the CANCEL method



6.5.1 Coding of the REL

SIP Message →	**REL →**
	cause parameter
BYE	Cause value No. 16 (normal clearing)
CANCEL	Cause value No. 31 (normal unspecified)?

6.5.3 Receipt of the Release Message

In the case that the REL message is received and a final response (i.e. 200 OK (INVITE)) has already been sent, the I-IWU shall send a BYE message. If the final response (i.e. 200 OK (INVITE)) has not already been sent, the I-IWU shall send Status-Code 4xx (Client Error) or 5xx(Server Error). The Status code to be sent is determined by examining the Cause code value received in the REL message. Table XX specifies the mapping of the cause code values to Status codes.

Note: In the case that the REL message is received and a final response (e.g. 200 OK(INVITE)) has already been sent (but no ACK has been received) on the incoming side of the I-IWU then the I-IWU shall NOT send a 487 Request terminated and instead wait until the ACK is received before sending a BYE message.

Table xx

Receipt of the Release message (REL)

←SIP Message	**← REL**
	Cause parameter
404 Not Found	Cause value No. 1 (unallocated (unassigned) number)

Table xx

Receipt of the Release message (REL)

←SIP Message	← REL
	Cause parameter
503 Service unavailable	Cause value No 2 (no route to network)
503 Service unavailable	Cause value No 3 (no route to destination)
503 Service unavailable	Cause value No. 4 (Send special information tone)
404 Not Found	Cause value No. 5 (Misdialled trunk prefix)
486 Busy Here	Cause value No. 17 (user busy)
480 Temporarily unavailable	Cause value No (18 no user responding)
480 Temporarily unavailable	Cause value No 19 (no answer from the user)
480 Temporarily unavailable	Cause value No. 20 (subscriber absent)
503 Temporarily unavailable	Cause value No 21 call rejected
410 Gone	Cause value No 22 number changed
480 Temporarily unavailable	Cause value No 25 (Exchange routing error)
502 Bad Gateway	Cause value No 27 (destination out of order)
503 Service unavailable	Cause value No 29 (facility rejected)
484 Address Incomplete	Cause value No. 28 invalid number format (address incomplete)
480 Temporarily unavailable	Cause value No 31 (normal unspecified) (class default)
486 Busy here if Diagnostics indicator includes the (CCBS indicator = CCBS possible) else 480 Temporarily unavailable	Cause value in the Class 010 (resource unavailable, Cause value No 34)
503 Service unavailable	Cause value in the Class 010 (resource unavailable, Cause value No's. 38, 41, 42, 43, 44, & 47) (47 is class default)
503 Service unavailable	Cause value No 57 (bearer capability not authorized)
503 Service unavailable	Cause value No 58 (bearer capability not presently)
503 Service unavailable	Cause value No 63 (service option not available, unspecified)
503 Service unavailable	Cause value in the Class 100 (service or option not implemented, Cause value No's. 65, 70 & 79) 79 is class default
503 Service unavailable	Cause value No 88 (incompatible destination)
404 Not Found	Cause value No 91 (invalid transit network selection)
503 Service unavailable	Cause value No 95 (invalid message)
503 Service unavailable	Cause value No 97 (Message type non-existent or not implemented)
503 Service unavailable	Cause value No 99 (information element/parameter non-existent or not implemented))
503 Service unavailable	Cause value No. 102 (recovery on timer expiry)
503 Service unavailable	Cause value No 110 (Message with unrecognized Parameter, discarded)
503 Service unavailable	Cause value No. 111 (protocol error, unspecified)
503 Service unavailable	127 (interworking unspecified)

6.6 Receipt of RSC, GRS or CGB (H/W oriented)

If a RSC, GRS or CGB (H/W oriented) message is received after an initial address message has been sent for that circuit and after at least one backward message relating to that call has been received then;

1. If the final response (i.e. 200 OK (INVITE)) has already been sent, the I-IWU shall send a BYE message.
2. If the final response (i.e. 200 OK (INVITE)) has not already been sent, the I-IWU shall send Status-Code 503 Service Unavailable

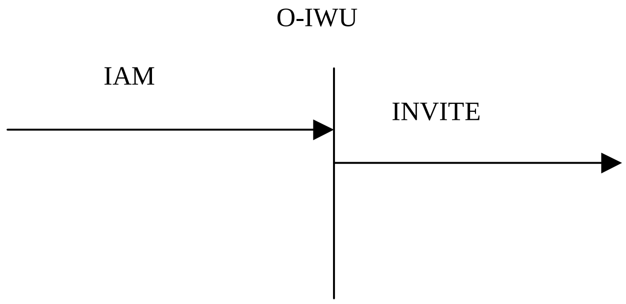
6.7 Internal through connection of the bearer path

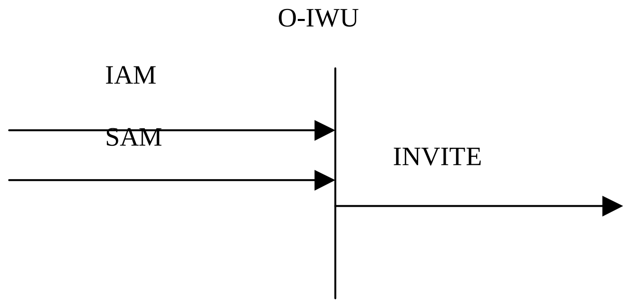
The I-IWU will through connect the internal switch path in both directions when;

1. the requested preconditions in the SIP network have been met, and
2. In the case of BICC when the BICC bearer set-up procedure is successfully completed.

7 Outgoing Call Interworking from BICC/ISUP to SIP at O-ISN

7.1 Sending of INVITE





After initiating the normal incoming BICC/ISUP call establishment procedures, determining the end of address signalling and selecting to route the call to the SIP network domain, the O-IWU shall send the initial INVITE with pre-conditions. Only calls with Transmission Requirements of speech or 3.1kHz audio will be routed to the SIP network domain, all other types of call attempts will be rejected.

The end of address signalling shall be determined:

a) by receipt of an end-of-pulsing (ST) signal; or
b) by receipt of the maximum number of digits used in the national numbering plan; or
c) by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party; or
d) 4-6 seconds (Ti/w1) after the receipt of the latest address message and the minimum number of digits required for routing the call have been received.

7.1.1 Coding of the INVITE

7.1.1.1 REQUEST URI Header

The called party number parameter of the IAM message is used to derive the userinfo component of the INVITE Request-URI.

7.1.1.2 SDP Media Description

The SDP encoding for the AMR codec[2] is specified in RFC 3267: "RTP payload format and file storage format for the Adaptive Multi-Rate (AMR) Adaptive Multi-Rate Wideband (AMR-WB) audio codecs" [XX].

7.1.1.3 P-Asserted-Identity and privacy header fields

Mapping BICC/ISUP CLI Parameters to SIP Header fields

[2] Trans-coding equipment may be required.

Has a Calling Party Number parameter with complete E.164 number, with Screening Indicator = UPVP or NP (See Note 1), and with APRI = “presentation allowed” or “presentation restricted” been received?	Has a Generic Number (additional calling party number) with a complete E.164 number, with Screening Indicator = UPNV, and with APRI = “presentation allowed” been received?	P-Asserted-Identity header field	From header field: display-name (optional) and addr-spec	Privacy header field
N	N	Header field not included	Unavailable@ Hostportion	Header field not included
N (Note 2).	Y	Header field not included	display-name derived from Generic Number (ACgPN) if possible. addr-spec derived from Generic Number (ACgPN) address signals (See **Error! Reference source not found.**) or uses network provided value	Header field not included
Y (See Note 1)	N	Derived from Calling Party Number parameter address signals (See **Error! Reference source not found.**)	if APRI = “allowed”, display-name may be derived from Calling Party Number (CgPN) if possible . if APRI = “restricted”, display-name is “Anonymous” if APRI = “allowed”, addr-spec is derived from Calling Party Number parameter address signals (See **Error! Reference source not found.**) or uses network provided value if APRI = “restricted”, addr-spec is set to the “Anonymous URI” (Note 3)	If Calling Party Number parameter APRI = “restricted” then priv-value =; “id”. For other APRI settings Privacy header is not included or if included, “id” is not included (See **Error! Reference source not found.**)
Y	Y	Derived from Calling Party Number parameter address signals (See **Error! Reference source not found.**)	display-name may be derived from Generic Number (ACgPN) if possible addr-spec is derived from Generic Number (ACgPN) address signals (See **Error! Reference source not found.**) or uses network provided value	If Calling Party Number parameter APRI = “restricted” then priv-value =; “id” . For other APRI settings Privacy header is not included or if included, “id” is not included (See **Error! Reference source not found.**)

Note-1: A Network Provided CLI in the CgPN parameter may occur on a call from an analogue access line. Therefore in order to allow the “display” of this Network Provided CLI at a SIP UAS it must be mapped into the SIP From header. It is also considered suitable to map into the P-Asserted-Identity header since it is a fully authentic CLI related exclusively to the calling line, and therefore equally as good a User Provided Verified and Passed CLI for this purpose.

Note 2: It is not clarified if the IWU is possible to set the From Header and the Display name derived form the Generic Number Parameter. This case is FFS because it may not be possible.

Note 3: The “From” header may contain an “Anonymous URI”. An “Anonymous URI” includes information that does not point to the calling party. RFC 3261 recommends that the display-name component contains "Anonymous". RFC [privacy] recommends that the Anonymous URI itself have the value "anonymous@anonymous.invalid".

Mapping of Generic Number (additional calling party number) to SIP From header fields

BICC/ISUP Parameter / field	**Value**	**SIP Component**	**Value**
Generic Number Number Qualifier Indicator	*“additional calling party number”*	From header field	display-name (optional) and addr-spec
Nature of Address Indicator	*“national (significant) number”*	Addr-spec	Add CC (of the country where the ISN is located) to GN address signals then map to SIP URI
	“international number”		Map complete GN address signals to SIP URI
Address signal	if NOA is *“national (significant) number”* then the format of the address signals is: NCD + SN	Display-name	Displayname may be mapped from Address Signal, if possible and network policy allows it.
	If NOA is *“international number”* then the format of the address signals is: CC + NCD + SN	Addr-spec	"+CC" "NCD" "SN" mapped to user portion of URI scheme used

7.1.4 Max Forwards header

The I-IWU shall perform the following interworking procedure in this clause if Hop Counter procedure is supported.

At the O-IWU the Hop Counter parameter shall be used to derive the Max-Forwards SIP header. Due to the different default values (that are based on network demands/provisions) of the SIP Max-Forwards header and the Hop Counter, an adaptation mechanism shall be used to adopt the Hop Counter to the Max Forwards at the O-IWU. For example, the following guidelines could be applied.

a) Max-Forwards for a given message should be monotone decreasing with each successive visit to a SIP entity, regardless of intervening interworking, and similarly for Hop Counter.

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

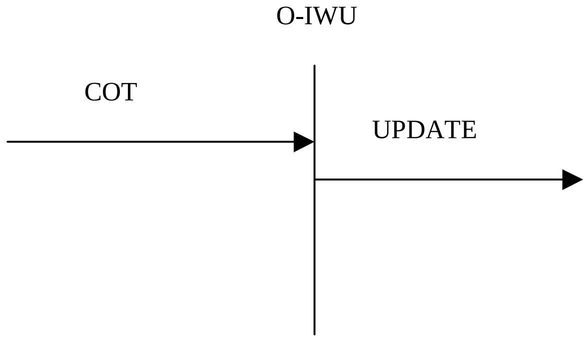
The following table shows the principle of the mapping:

Hop Counter	= Y	Max-Forwards	= X

Note: The Mapping of value X to Y should be done with the used (implemented) adaptation mechanism.

The Principle of adaptation could be implemented on a basis of the network provision, trust domain rules and bilateral agreement

7.2 Sending of UPDATE

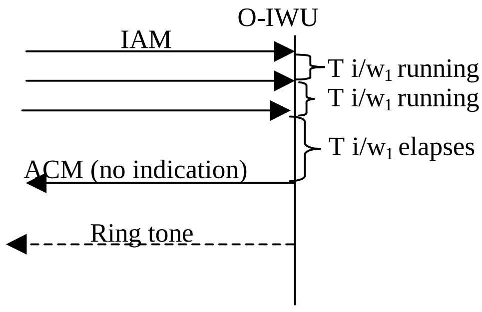


When the requested preconditions in the SIP network have been met, outstanding continuity procedures successfully completed and the incoming bearer establishment procedures (in the case of BICC) successfully completed, the UPDATE is sent confirming that all the required preconditions have been met.

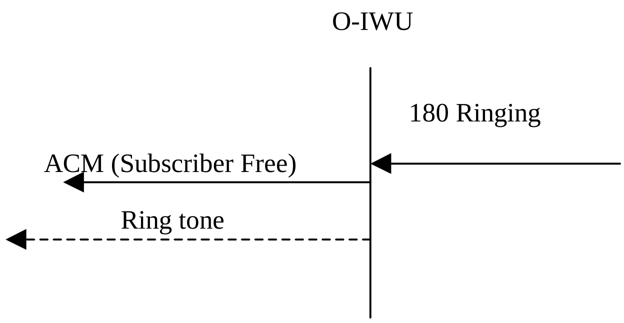
7.3 Sending of ACM and Awaiting Answer indication

The following cases are possible trigger conditions for sending the address complete message (ACM) and awaiting answer indication (e.g. ring tone):

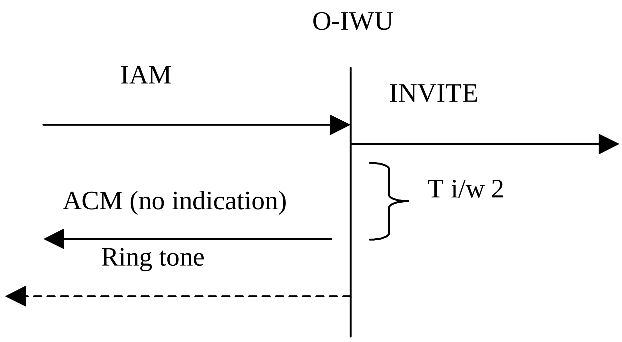
a) The detection of end of address signalling by the expiry of Timer T i/w 1 or,



b) The reception of 180 Ringing or,



c) 4-6 seconds (Ti/w 2) after the initial INVITE is sent



NOTE – In all cases, it is assumed that no Address Complete Message (ACM) has already been sent.

7.3.1 Coding of the ACM

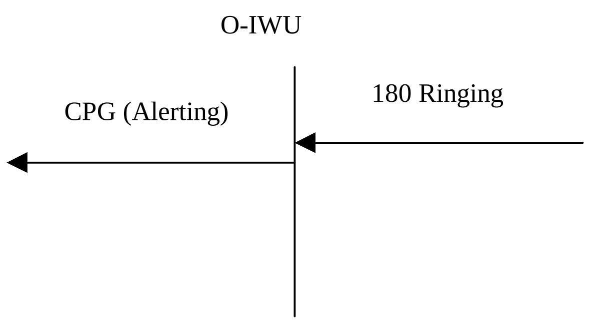
7.3.1.1 Backward call indicators

bits	**AB**	**Charge indicator Contributors**
	1 0	*charge*
bits	**DC**	**Called party's status indicator**
	0 1	*subscriber free* if the 180 Ringing has been received.
	0 0	*no indication* otherwise
bits	**FE**	**Called party's category indicator**
	0 0	*no indication*
bits	**HG**	**End-to-end method indicator**
	01	*no end-to-end method available*
bit	**I**	**Interworking indicator**
	1	*interworking encountered*
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*
bit	**L**	**Holding indicator (national use)**
	0	*holding not requested*
bit	**M**	**ISDN access indicator**
	0	*terminating access non-ISDN*

7.4 Sending of the Call Progress message (CPG)

If the Address Complete Message (ACM) has already been sent, the following cases are possible trigger conditions for sending the Call Progress message (CPG):

a) Receipt of the first the 180 Ringing



7.4.1 Coding of the CPG

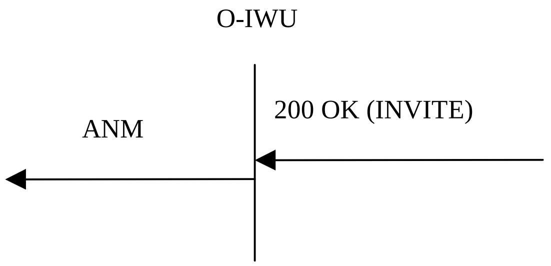
7.4.1.1 Event information

bits	G-A	Event indicator
	0 0 0 0 0 0 1	*alerting*

7.5 Sending of the Answer Message (ANM)

Upon receipt of the first 200 OK (INVITE), if the address complete message has already been sent, the interworking exchange shall:

– stop the sending of the awaiting answer indication (if any),
– through connect the internal switch path in both directions, and
– send the Answer Message (ANM) to the preceding exchange.

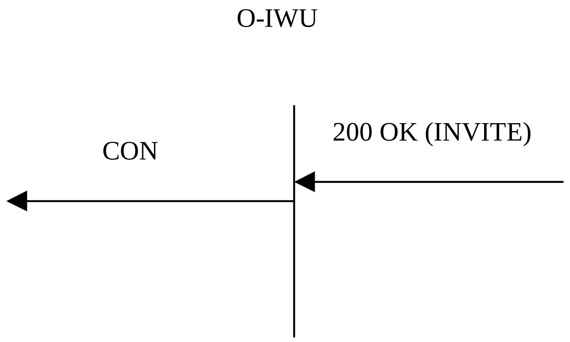


7.5.1 Coding of the ANM

The Answer Message (ANM) contains no parameters.

7.6 Sending of the Connect message (CON)

Upon receipt of the first 200 OK (INVITE), if the Address Complete Message (ACM) has not yet been sent, the O-IWU shall through connect the internal switch path in both directions and send the Connect message (CON) to the preceding exchange.

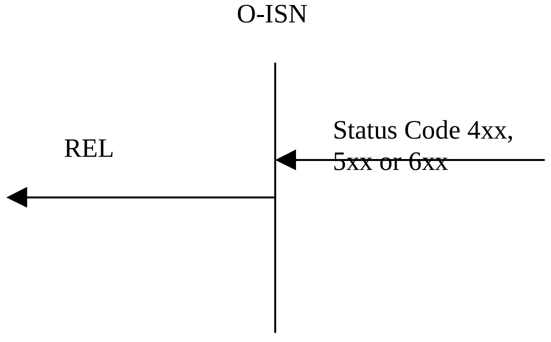


7.6.1 Coding of the CON

7.6.1.1 Backward call indicators

See Section 7.3.1.1

7.7 Receipt of Status Codes 4xx, 5xx or 6xx



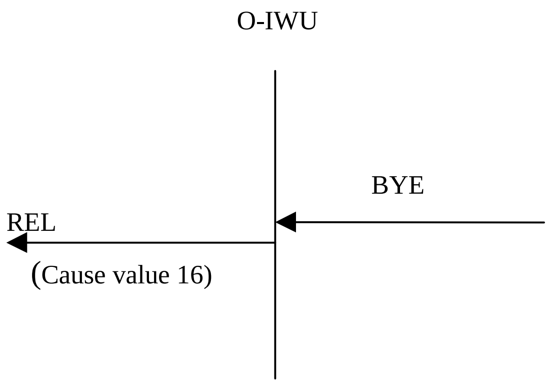
The reception of Status Codes 4xx, 5xx or 6xx will trigger the sending of a REL message. The coding of the Cause parameter value is derived from the Status code received according to table yy.

Table YY 4xx/5xx/6xx Received on SIP side of O-IWU

←REL (cause code)	←4xx/5xx/6xx SIP Message
127 (interworking unspecified)	400 Bad Request
127 (interworking unspecified)	401 Unauthorised
127 (interworking unspecified)	402 Payment Required
127 (interworking unspecified)	403 Forbidden
127 (Unallocated number)	404 Not Found
127 (interworking unspecified)	405 Method Not Allowed
127 (interworking unspecified)	406 Not Acceptable
127 (interworking unspecified)	407 Proxy authentication required
127 (interworking unspecified)	408 Request Timeout
22 (Number changed)	410 Gone
127 (interworking unspecified)	413 Request Entity too long
127 (interworking unspecified)	414 Request-uri too long
127 (interworking unspecified)	415 Unsupported Media type
127 (interworking unspecified)	416 Unsupported URI scheme
127 (interworking unspecified)	420 Bad Extension
127 (interworking unspecified)	421 Extension required
127 (interworking unspecified)	"423" ; Interval Too Brief
127 (No user responding)	480 Temporarily Unavailable
127 (interworking unspecified)	481 Call/Transaction does not exist
25 (Exchange routing error)	483 Too many hops
28 (Invalid Number format)	484 Address Incomplete
127 (interworking unspecified)	485 Ambigous
17 (User busy)	486 Busy Here
127 No mapping	487 Request terminated
127 (interworking unspecified)	488 Not acceptable here
127 (interworking unspecified)	490 Request Updated
127 (interworking unspecified)	"491" ; Request Pending
127 (interworking unspecified)	"493" ; Undecipherable
127 (interworking unspecified)	500 Server Internal error
127 (interworking unspecified)	501 Not implemented
127 (interworking unspecified)	502 Bad Gateway
127 (interworking unspecified)	503 Service Unavailable

←REL (cause code)	←4xx/5xx/6xx SIP Message
127 (interworking unspecified)	504 Server timeout
127 (interworking unspecified)	505 Version not supported
127 (interworking unspecified)	513 Message too large
127 (interworking unspecified)	580 Precondition failure
17 (User busy)	600 Busy Everywhere
21 (Call rejected)	603 Decline
1 (unallocated number)	604 Does not exist anywhere
127 (interworking unspecified)	606 Not acceptable

7.8 Receipt of a BYE



On receipt of a BYE method, the O-IWU sends a REL message with Cause Code value 16 (Normal Call Clearing).

7.9 Receipt of the Release Message

In the case that the REL message is received and a final response (i.e. 200 OK (INVITE)) has already been received the O-IWU shall send a BYE method. If the final response (i.e. 200 OK (INVITE)) has not already been received the O-IWU shall send a CANCEL method.

7.10 Receipt of RSC, GRS or CGB (H/W oriented)

If a RSC, GRS or CGB (H/W oriented) message is received and a final response (i.e. 200 OK (INVITE) has already been received the O-IWU shall send a BYE method. If a final response (i.e. 200 OK (INVITE)) has not already been received the O-IWU shall send a CANCEL method.

8 Timers

Symbol	Time-out value	Cause for initiation	Normal termination	At expiry	Reference
Ti/w1	4-6 seconds	When last address message is received in interworking situations.	At the receipt of fresh information.	Send INVITE, send the address complete message and insert ring tone	7.1 7.3
Ti/w2	4-6 seconds	When latest address message is received in interworking situations.	When ACM is sent.	Send address complete message and insert ring tone	7.3

ANNEX A

ANNEX A.1 Interworking of CLIP/CLIR Supplementary service to SIP networks.

To be added…

ANNEX B

ANNEX B.1 Interworking scenarios between SIP and BICC

To be added…