



MSF R4 SIP-ISUP IA

MSF-IA-SIP.019-FINAL

MultiService Forum Implementation Agreement

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Abstract:

The MultiService Forum (MSF) is responsible for developing Implementation Agreements or Architectural Frameworks which can be used by developers and network operators to ensure interoperability between components from different vendors. MSF Implementation Agreements are formally ratified via a Straw Ballot and then a Principal Member Ballot.

Draft MSF Implementation Agreements or Architectural Framework may be published before formal ratification via Straw or Principal Member Ballot. In order for this to take place, the MSF Technical Committee must formally agree that a draft Implementation Agreement or Architectural Framework should be progressed through the balloting process. A Draft MSF Implementation Agreement or Architectural Framework is given a document number in the same manner as an Implementation Agreement.

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The use of capitalization of the key words "MUST", "SHALL", "REQUIRED", "MUST NOT", "SHOULD NOT", "SHOULD", "RECOMMENDED", "NOT RECOMMENDED", "MAY" or "OPTIONAL" is as described in section V-B of the MSF Technical Committee Operating Procedures.

The goal of the MSF is to promote multi-vendor interoperability as part of a drive to accelerate the deployment of next generation networks. To this end the MSF looks to adopt pragmatic solutions in order to maximize the chances for early deployment in real world networks.

To date the MSF has defined a number of detailed Implementation Agreements and detailed Test Plans for the signaling protocols between network components and is developing additional Implementation Agreements and Test Plans addressing some of the other technical issues such as QoS and Security to assist vendors and operators in deploying interoperable solutions.

The MSF welcomes feedback and comment and would encourage interested parties to get involved in this work program. Information about the MSF and membership options can be found on the MSF website <http://www.msforum.org/>

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I. The MultiService Forum

The MultiService Forum (MSF) is a global association of service providers, system suppliers and other organizations committed to developing and promoting open-architecture, multiservice communication systems. Founded in 1998, the MSF is an open-membership organization comprised of the world's leading telecommunications companies.

The MSF's activities include developing implementation agreements, promoting worldwide compatibility and interoperability, and encouraging input to appropriate national and international standards bodies.

As part of MSF's effort to drive and promote interoperability, the MSF has created a number of programs geared toward accelerating real world network deployments:

1. Global MSF Interoperability (GMI) events. GMI events provide a real-world setting for vendors to test their solutions and provide evidence that vendor products meet the interoperability standards set forth by MSF Implementation Agreements. Each MSF GMI event is built around a set of capabilities defined for a given release of the MSF Architecture.
2. Next Generation Network (NGN) Test Bed. The NGN test bed provides a facility to enable carriers and vendors to perform in-depth testing of a specific interface as defined in a given release of the MSF architecture.
3. Certification Programs. For more mature technologies the MSF can provide Certification of compliance to a given Implementation Agreement where MSF members believe that it is of value to the industry to do so.

II. An introduction to MSF documentation and GMI 2008

This document is part of the MSF Release 4 set of architectural, protocol and test documentation.

The MSF Release 4 Architecture is a physical implementation of the functional architectures that have been proposed by the key Standards Development Organizations. As such the MSF Release 4 Architecture represents the current state of the industry and it identifies current open interfaces between physically separate network elements.

MSF Implementation Agreements define the protocols to be used over specific open interfaces. Where possible MSF Implementation Agreements are based on industry standard protocols augmented with additional information so as to ensure interoperability between communicating network elements. This level of interoperability is achieved by closing any gaps and tightening any optional capabilities in those industry standards to remove the danger of mutually incompatible selections by vendors. An MSF

Implementation Agreement is targeted at a given release of the MSF architecture but can be used in any circumstance where an operator wishes to deploy the open interface and its functionality within their own network.

The MSF Release 4 architecture and its associated implementation agreements are used as the basis for GMI 2008. GMI 2008 is a global test event executed to demonstrate multi-vendor, multi-service interoperability based around IMS and includes IPTV and web based services.

As part of GMI 2008 a number of detailed test scenarios have been developed and a number of test plans defined. Test plans contain the set of test cases required to demonstrate a given MSF Release 4 capability and serve to exercise and validate the set of Implementation Agreements required to realize the capability.

Following the completion of GMI 2008 the MSF Release 4 architecture and individual implementation agreements will be updated if the testing identifies any deficiencies in the documents.

For more information about the scope of GMI2008 please go to <http://www.msforum.org>

III. Impact on previously published MSF documents

This specification is for MSF release 4 and GMI 2008. It replaces the following earlier MSF documents

MSF-IA-SIP.004-FINAL

I. Scope

This Implementation Agreement covers the SIP-ISUP Inter-working IA for a CA/MGCF within the MSFR4 architecture in preparation for the GMI2008 event. This IA is an endorsement of ETSI TISPAN ES 283 027 (v2.4.0) which in turn is an endorsement of 3GPP TS 29.163 (Inter-working between IP Multimedia Core Network Subsystem and Circuit Switched Networks). The specification ES 283 027 v2.4.0 is part of TISPAN NGN R2.

This IA corresponds to the MSF SESS1 interface in the MSF R4 Architecture (see [27]) and is applicable to the following MSFR4 entities:-

- CA

Note that the MSF CA is regarded as being a co-located MGC(MSF)/AGC(MSG)/S-CSC(MSF)/AS.

II. Relationship to corresponding MSFR3 IA

This IA is applicable to a MGCF which is inter-working SIP on its core n/w side and ISUP on its PSTN side. In MSFR3, there was no explicit MGCF. Rather, the MGCF functionality was represented by a R2 CA entity. Therefore, the corresponding IA in MSFR3 was MSF-IA-SIP.004-FINAL – “IA for an Inter-CA SIP Profile”, which was inherited from GMI2004. The previous IA was a stand-alone document as opposed to being an endorsement of another standard. More specifically, the previous IA was actually written for MSFR2 for the GMI2004 event and was simply carried forward to MSFR3. With the benefit of hindsight, it is probable that an endorsement of 3GPP 29.163 ought to have been created for the GIM2006 event, but this was not done due to GMI2006 being heavily geared toward IMS itself with the inter-working with the PSTN being a lower priority scenario. However, it is now appropriate to replace the historical Inter-CA IA with an endorsement of ETSI TISPAN ES 283 027.

III. MSF Endorsement Notice

The present document endorses ES 283 027 (V2.4.0): "Inter-working between SIP and BICC / ISUP " [ITU-T Q.1912.5, modified].

Where implementation detail is not provided within this IA, the implementation specification as described in ES 283 027 shall take precedence over that in other specifications.

Except where otherwise indicated, each section that follows has the same number as the section of ES 283 027 which it modifies.

The following notation is used to identify the differing types of changes or modifications used compared to the specification in ES 283 027.

- <AP> - indicates a provision which adds precision, but no new normative content.
- <NEW> - indicates new normative content. <NEW> in a section header indicates that the section heading is new relative to ES 283 027 v2.4.0.
- <CHG> - indicates changed normative content.

Note: The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", "OPTIONAL", "CONDITIONAL" and "IF" in this document are to be interpreted as described in the Technical Committee Operating Procedures.

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MSF Release 4 **Endorsement of ETSI TISPAN ES 283 027 (v2.4.0)**

1 Scope

<CHG> Replace the current text with the following.

The present document provides the MSF endorsement of the ETSI ES 283 027 [28].

The present document specifies the principles of interworking between MSFR4 and ISUP based legacy CS (Circuit Switched) networks, in order to support IM basic voice calls, therefore all the references to interworking between IMS and BICC are out of scope of this standard.

This document is a protocol interworking specification; therefore all references to the user plane interworking are out of scope of this standard.

<AP> This document relates to the SESS1 interface in [27].

2 References

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

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

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

- [1] 3GPP TS 29.163 : "3rd Generation Partnership Project; Technical Specification Group Core Network and Terminals; Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (Release 7)".
- [2] ETSI EN 300 356-1 (V4.2.1): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 1: Basic services [ITU-T Recommendations Q.761 to Q.764 (1999) modified]".
- [3] ETSI ES 283 003: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IP Multimedia Call Control Protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) Stage 3 [3GPP TS 24.229 (Release 7), modified]".
- [4] ETSI ES 282 007: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IP Multimedia Subsystem (IMS); Functional architecture".
- [5] ETSI EN 300 356-3 (V4.2.1): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 3: Calling Line Identification Presentation (CLIP) supplementary service [ITU-T Recommendation Q.731, clause 3 (1993) modified]".
- [6] ETSI EN 300 356-4 (V4.2.1): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 4: Calling Line Identification Restriction (CLIR) supplementary service [ITU-T Recommendation Q.731, clause 4 (1993) modified]".

- [7] ETSI EN 300 356-5 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 5: Connected Line Identification Presentation (COLP) supplementary service [ITU-T Recommendation Q.731, clause 5 (1993) modified]".
- [8] ETSI EN 300 356-6 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 6: Connected Line Identification Restriction (COLR) supplementary service [ITU-T Recommendation Q.731, clause 6 (1993) modified]".
- [9] ETSI EN 300 356-7 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 7: Terminal Portability (TP) supplementary service [ITU-T Recommendation Q.733, clause 4 (1993) modified]".
- [10] ETSI EN 300 356-8 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 8: User-to-User Signalling (UUS) supplementary service [ITU-T Recommendation Q.737, clause 1 (1997) modified]".
- [11] ETSI EN 300 356-9 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 9: Closed User Group (CUG) supplementary service [ITU-T Recommendation Q.735, clause 1 (1993) modified]".
- [12] ETSI EN 300 356-10 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 10: Subaddressing (SUB) supplementary service [ITU-T Recommendation Q.731, clause 8 (1992) modified]".
- [13] ETSI EN 300 356-11 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 11: Malicious Call Identification (MCID) supplementary service [ITU-T Recommendation Q.731, clause 7 (1997) modified]".
- [14] ETSI EN 300 356-12 (V4.2.1): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 12: Conference call, add-on (CONF) supplementary service [ITU-T Recommendation Q.734, clause 1 (1993) and implementors guide (1998) modified]".
- [15] ETSI EN 300 356-14 (V4.2.1): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 14: Explicit Call Transfer (ECT) supplementary service [ITU-T Recommendation Q.732, clause 7 (1996) and implementors guide (1998) modified]".
- [16] ETSI EN 300 356-15 (V4.2.1): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 15: Diversion supplementary service [ITU-T Recommendation Q.732, clauses 2 to 5 (1999) modified]".
- [17] ETSI EN 300 356-16 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 16: Call Hold (HOLD) supplementary service [ITU-T Recommendation Q.733, clause 2 (1993) modified]".
- [18] ETSI EN 300 356-17 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 17: Call Waiting (CW) supplementary service [ITU-T Recommendation Q.733, clause 1 (1992) modified]".
- [19] ETSI EN 300 356-18 (V4.1.2): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 18: Completion of Calls to Busy Subscriber (CCBS) supplementary service [ITU-T Recommendation Q.733, clause 3 (1997) modified]".

- [20] ETSI EN 300 356-19 (V4.2.1): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 19: Three-Party (3PTY) supplementary service [ITU-T Recommendation Q.734, clause 2 (1996) and implementors guide (1998) modified]".
- [21] ETSI EN 300 356-20 (V4.3.1): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 20: Completion of Calls on No Reply (CCNR) supplementary service [ITU-T Recommendation Q.733, clause 5 (1999) modified]".
- [22] ETSI EN 300 356-21: "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 21: Anonymous Call Rejection (ACR) supplementary service [ITU-T Recommendation Q.731, clause 4 (1993)]".
- [23] ETSI EN 300 485 (V1.2.3): "Integrated Services Digital Network (ISDN); Definition and usage of cause and location in Digital Subscriber Signalling System No. one (DSS1) and Signalling System No.7 ISDN User Part (ISUP) [ITU-T Recommendation Q.850 (1998), modified]".
- [25] ETSI EN 300 403-1: "Integrated Services Digital Network (ISDN); Digital Subscriber Signalling System No. one (DSS1) protocol; Signalling network layer for circuit-mode basic call control; Part 1: Protocol specification [ITU-T Recommendation Q.931 (1993), modified]".
- [26] RFC 3966 (December 2004): "The tel URI for Telephone Numbers".
- [27] [MSF-ARCH-004.00-FINAL - MSF Release 4 Architecture](#)
- [28] [ETSI ES 283 027 v2.4.0 \(2007-11\) – TISPAN; Endorsement of the SIP-ISUP Interworking between the IP Multimedia Core Network and Circuit Switched Networks. \[3GPP TS 29.163 \(Release 7\), modified\]](#)

3 Definitions and abbreviations

For the purposes of the present document, the terms, definitions and abbreviations given in 3GPP TS 29.163 [1], clauses 7.2.3 and 7.4 apply.

Endorsement notice

3GPP TS 29.163 [1] applies, with the following modifications:

Clauses 6.3, 7.3, 8, 9.2.2.1, 9.2.2.2, 9.2.3.1, 9.2.3.2, 9.2.4.1, 9.2.5.1, 9.2.6.1, 9.2.7.1, 9.2.8.1, 9.2.8.3 and Annex B do not apply to this recommendation.

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

Replace references as shown below.

Reference in 3GPP TS 29.163 [1]	Modified reference
ITU-T Recommendation Q.761	EN 300 356-1 [Error! Reference source not found.]
ITU-T Recommendation Q.762	EN 300 356-1 [Error! Reference source not found.]
ITU-T Recommendation Q.763	EN 300 356-1 [Error! Reference source not found.]
ITU-T Recommendation Q.764	EN 300 356-1 [Error! Reference source not found.]
3GPP TS 24.229	ES 283 003 [3]
3GPP TS 23.228	ES 282 007 [4]
ITU-T Recommendation Q.850 (1998)	EN 300 485 [Error! Reference source not found.]
ITU-T Recommendation Q.731.3	EN 300 356-3 [Error! Reference source not found.]
ITU-T Recommendation Q.731.4	EN 300 356-4 [Error! Reference source not found.]
ITU-T Recommendation Q.731.5	EN 300 356-5 [Error! Reference source not found.]
ITU-T Recommendation Q.731.6	EN 300 356-6 [Error! Reference source not found.]
ITU-T Recommendation Q.731.7	EN 300 356-11 [Error! Reference source not found.]
ITU-T Recommendation Q.731.8	EN 300 356-10 [Error! Reference source not found.]
ITU-T Recommendation Q.732.2	EN 300 356-15 [Error! Reference source not found.]
ITU-T Recommendation Q.732.3	EN 300 356-15 [Error! Reference source not found.]
ITU-T Recommendation Q.732.4	EN 300 356-15 [Error! Reference source not found.]
ITU-T Recommendation Q.732.5	EN 300 356-15 [Error! Reference source not found.]
ITU-T Recommendation Q.732.7	EN 300 356-14 [Error! Reference source not found.]
ITU-T Recommendation Q.733.1	EN 300 356-17 [Error! Reference source not found.]
ITU-T Recommendation Q.733.2	EN 300 356-16 [Error! Reference source not found.]
ITU-T Recommendation Q.733.3	EN 300 356-18 [Error! Reference source not found.]
ITU-T Recommendation Q.733.4	EN 300 356-7 [Error! Reference source not found.]
ITU-T Recommendation Q.733.5	EN 300 356-20 [Error! Reference source not found.]
ITU-T Recommendation Q.734.1	EN 300 356-12 [Error! Reference source not found.]
ITU-T Recommendation Q.734.2	EN 300 356-19 [Error! Reference source not found.]
ITU-T Recommendation Q.735.1	EN 300 356-9 [Error! Reference source not found.]
ITU-T Recommendation Q.737.1	EN 300 356-8 [Error! Reference source not found.]

Modifications to 3GPP TS 29.163 Release 7

Clause 7.2.3.1.2.5
Modify as follows:

7.2.3.1.2.5 *Transmission medium requirement*

The I-MGCF may either transcode the selected codec(s) to the codec on the PSTN side or it may attempt to interwork the media without transcoding. If the I-MGCF transcodes, it shall select the TMR parameter to "3.1 kHz audio". If the I-MGCF does not transcode, it should map the TMR, USI and Access Transport parameters from the selected codec according to Table 2a. The support of any of the media listed in Table 2a is optional. The SDP for the data transfer with 64 kbit/s clearmode shall be mapped to the TMR "64 kbit/s unrestricted".

If the XML BearerCapability element is received in the INVITE the mapping of the USI shall be taken from the BearerCapability XML, else the USI shall be derived from the SDP attributes as shown in Table2a .

Add Clause 7.2.3.1.2.1.5a

7.2.3.1.2.5a *Transmission medium requirement prime and USI prime (optional)*

The Fallback mechanism shall only apply if a XML Bearer Capability element appears within the INVITE Request.

When the INVITE request includes multiple audio codecs with codec appearing in Table 2a then the I-MGCF shall:

- If the first stated codec in the INVITE is a codec appearing in Table 2a and is the equivalent as stated within the second Bearer Capability in the XML Bearer Capability element then the I-MGCF shall map the XML Bearer Capability element into the TMR and USI prime.
- If the second stated codec in the INVITE is a codec appearing in Table 2a and is the equivalent as stated within the first Bearer Capability in the XML Bearer Capability element then the I-MGCF shall map the XML Bearer Capability element into the TMR prime and USI.
- If the compared codec stated within the INVITE is not equivalent as stated within XML Bearer Capability element then the XML Bearer Capability element shall be discarded

If the Fallback mechanism is not supported by the terminating network the procedures as described within clause 7.2.3.1.2.5 shall apply

In cases where the fallback appears within the terminating entity and sends back a codec to which is fallen back then the I-MGCF shall only apply the cut-though to the chosen codec.

In cases where the fallback appears within the terminating entity and sends back a SDP Answer that is equivalent with the TMR prime codec (fallback codec) then the I-MGCF shall only apply the cut-though to the fallback codec.

In cases where preconditions are used the I-MGCF has to wait for the SDP answer where the preconditions are met and fallback codec is sent back.

Add Clause 7.2.3.1.2.10

7.2.3.1.2.10 *Access Transport Parameter and User Tele Service*

When an INVITE was received containing a PSTN XML body as defined in ES 283 003 [3], an available "ProgressIndicator" element can be as a network option mapped into a Progress Indicator in the Access Transport Parameter of the sent IAM

Table 7a0.1 : Mapping of PSTN XML elements with ISUP Parameters

INVITE →	IAM →	
PSTN XML	ISUP Parameter	Content
ProgressIndicator	Access Transport Parameter	Progress indicator
HighLayerCompatibility		High layer compatibility (Note)
LowLayerCompatibility		Low layer compatibility
BearerCapability	User Service Information	
HighLayerCompatibility	User Tele Service	High layer compatibility (Note)
NOTE: If two high layer compatibility information elements are received, they are transferred in the same order as received in the INVITE message in the access transport parameter of the initial address message.		

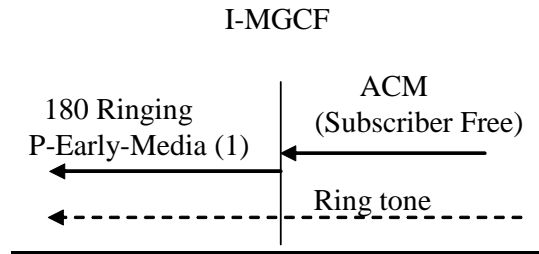
Clause 7.2.3.1.4

Modify as follows:

7.2.3.1.4 Sending of 180 ringing

The I-MGCF shall send the SIP 180 Ringing when receiving any of the following messages:

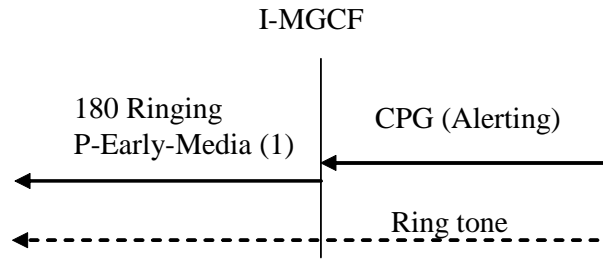
- ACM with Called party's status indicator set to subscriber free.



NOTE 1: Including the P-Early-Media Header is a network option for a speech call.

Figure 6: The receipt of ACM

- CPG with Event indicator set to alerting



NOTE 1: Including the P-Early-Media Header is a network option for a speech call.

Figure 7: Receipt of CPG (Alerting)

For a speech call, if the I-MGCF supports the P-Early-Media header as a network option, and if the INVITE request includes the P-Early-Media header, the I-MGCF shall include in the SIP 180 Ringing response a P-Early-Media header authorizing early media, except when

- the I-MGCF has already sent a reliable provisional response including a P-Early-Media header, as defined in IETF draft-ejzak-sipping-p-em-auth-04.txt [89], and
- the most recently sent P-Early-Media header authorized early media.

NOTE: If the I-MGCF signals the P-Early-Media header authorizing early media, then the IMS can expect tones or announcements to the calling party to flow from the CS network via an MGW controlled by the I-MGCF. In particular, once the I-MGCF sends the 180 Ringing response, ringback is expected in media from the CS network.

Received PSTN XML elements shall be mapped as shown in Table 7a0.2

Table 7a0.2 : ISUP Parameters with Mapping of PSTN XML elements

←18x	←CPG or ACM	
<u>PSTN XML</u>	<u>ISUP Parameter</u>	<u>Content</u>
ProgressIndicator	Access Transport Parameter	Progress indicator
HighLayerCompatibility		High layer compatibility (Note)
LowLayerCompatibility		Low layer compatibility
NOTE: If two high layer compatibility information elements are received, they are transferred in the same order as received in the INVITE message in the access transport parameter of the initial address message.		

Table 7a0.3 : Sending criteria of the XML with Progress indicator X

← 180 Ringing	←ACM
<u>PSTN XML body with Progress indicator X</u>	<u>Content</u>
No. 1 <i>(Call is not end-to-end ISDN: further progress information may be available in-band)</i>	Backward call indicators parameter ISDN User Part indicator 0 ISDN User Part not used <i>all the way</i>
No. 2 <i>(Destination address is non-ISDN)</i>	Backward call indicators parameter ISDN User Part indicator 1 ISDN User Part used <i>all the way</i> ISDN access indicator 0 <i>Terminating access non-ISDN</i>
No. 4 <i>(Call is returned to ISDN)</i>	Backward call indicators parameter ISDN User Part indicator 1 ISDN User Part used <i>all the way</i> ISDN access indicator 1 <i>Terminating access ISDN</i>

Progress indicator

Progress indicator information elements possibly present in the access transport parameter of the Address Complete Message (ACM) are transferred into the PSTN XML body (as defined in ES 283 003 [3]) of the 180 Ringing sent to the calling user.

Clause 7.2.3.1.4A

Modify as follows:

7.2.3.1.4A Sending of 183 Session Progress for early media scenarios

If SIP preconditions are used, the first 183 Session Progress will be sent after the reception of the INVITE request, before any ISUP message has been received from the CS network. The I-MGCF shall not include the P-Early-Media header in any SIP message before it receives an ISUP ACM.

On receipt of an ACM with the options described in table 7a2 the I-MGCF can be sent, as a network option, a 183 Session Progress response with following options:

Table 7a2 : Sending criteria of the XML with Progress indicator X

← 183 Session Progress	←ACM
PSTN XML body with Progress indicator X	Content
No. 1 <i>(Call is not end-to-end ISDN: further progress information may be available in-band)</i>	Backward call indicators parameter ISDN User Part indicator 0 <i>ISDN User Part not used all the way</i>
No. 2 <i>(Destination address is non-ISDN)</i>	Backward call indicators parameter ISDN User Part indicator 1 <i>ISDN User Part used all the way</i> ISDN access indicator 0 <i>Terminating access non-ISDN</i>
No. 4 <i>(Call is returned to ISDN)</i>	Backward call indicators parameter ISDN User Part indicator 1 <i>ISDN User Part used all the way</i> ISDN access indicator 1 <i>Terminating access ISDN</i>

Progress indicator

Progress indicator information elements possibly present in the access transport parameter of the Address Complete Message (ACM) are transferred into the PSTN XML body (as defined in ES 283 003 [3]) of the 183 Session Progress sent to the calling user.

For a speech call upon receipt of one of the following messages, if the I-MGCF supports the P-Early-Media header as a network option, and if the I-MGCF has received the P-Early-Media header in the INVITE request, and has not already sent a provisional response including a P-Early-Media header with parameters indicating authorization of early media, then the I-MGCF shall send the 183 Session Progress response with a P-Early-Media header authorizing early media:

- ACM with the value of the called party’s status indicator “no indication” and one of the options described in table 7a1. Based on local configuration, the I-MGCF may also send a 183 Session Progress response with a P-Early-Media header authorizing early media if it receives an ACM with other parameters than described in table 7a1.

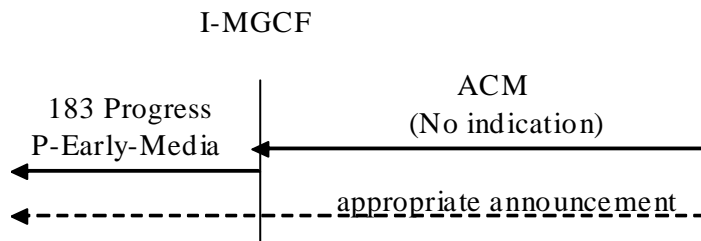


Figure 7c: Receipt of ACM "No indication"

Table 7a.1: ACM Parameters that trigger the 183 Session Progress response

←183 Session Progress	←ACM
183 Session Progress response including a P-Early-Media header authorizing early media, if not already sent	1) Optional backward call indicators parameter In-band information indicator 1 <i>In-band info...</i> 2) Backward call indicators parameter ISDN User Part indicator 0 ISDN User Part not used all the way

NOTE: As a network option the I-MGCF can also map ACM into 183 in other cases than those described in table 7a1.

- 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 7b1.

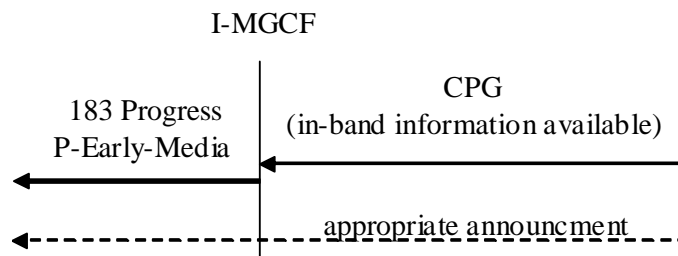


Figure 7d: Receipt of CPG (in-band information available)

Table 7b.1: CPG Parameters that trigger the 183 Session Progress response

←183 Session Progress	←CPG
183 Session Progress response including a P-Early-Media header authorizing early media, if not already sent	Event indicator
	000 0010 (<i>progress</i>)
	1) Optional backward call indicators parameter
	In-band information indicator
	0 <i>In-band info ...</i>
	2) Backward call indicators parameter
	ISDN User Part indicator
	0 ISDN User Part not used all the way

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: As a network option the I-MGCF can also map CPG into 183 in other cases than those described in table 7a1.

Received PSTN XML elements shall be mapped as shown in Table 7a0.2

On receipt of an CPG with the options described in table 7a3 the I-MGCF can be sent, as a network option, 183 Session Progress response with following options:

<u>← 183 Session Progress</u>	<u>←CPG</u>
<u>PSTN XML body with Progress indicator X</u>	<u>Content</u>
No. 1 <i>(Call is not end-to-end ISDN: further progress information may be available in-band)</i>	Backward call indicators parameter ISDN User Part indicator 0 ISDN User Part not used <i>all the way</i>
No. 2 <i>(Destination address is non-ISDN)</i>	Backward call indicators parameter ISDN User Part indicator 1 ISDN User Part used all <i>the way</i> ISDN access indicator 0 Terminating access non- <i>ISDN</i>
No. 4 <i>(Call is returned to ISDN)</i>	Backward call indicators parameter ISDN User Part indicator 1 ISDN User Part used all <i>the way</i> ISDN access indicator 1 Terminating access ISDN

Progress indicator

Progress indicator information elements possibly present in the access transport parameter of the Call Progress Message (CPG) are transferred into the PSTN XML (as defined in ES 283 003 [3]) body of the 183 Session Progress to the calling user.

Clause 7.2.3.1.5

Modify as follows:

7.2.3.1.5 Sending of the 200 OK (INVITE)

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

- The reception of the ANM.

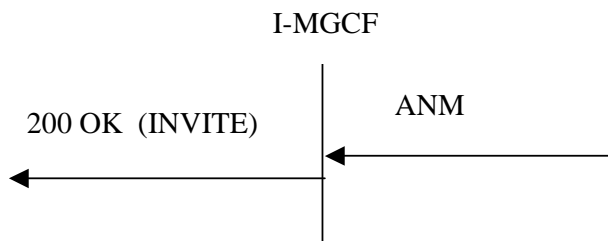


Figure 8: Receipt of ANM

- The reception of the CON message.

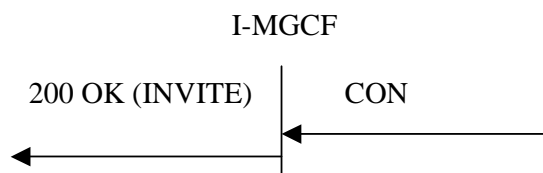


Figure 9: Receipt of CON

Received PSTN XML elements shall be mapped as shown in Table 7a0.2

Clause 7.2.3.1.6

Modify as follows:

7.2.3.1.6 Sending of the Release message (REL)

The following are possible triggers for sending the Release message:

- Receipt of the BYE method.

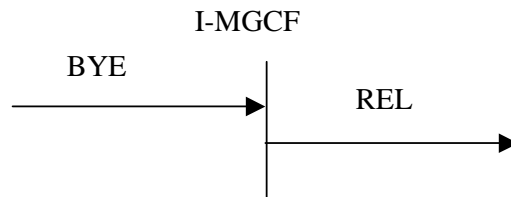


Figure 10: Receipt of the Bye method

- Receipt of the CANCEL method

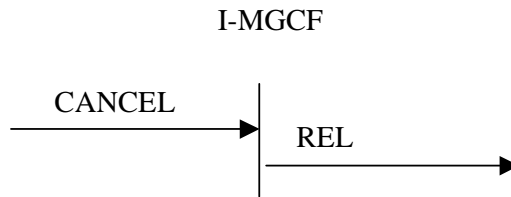


Figure 11: Receipt of Cancel method

Additional triggers are contained in table 10.

Received PSTN XML elements shall be mapped as shown in Table 8b

Clause 7.2.3.1.7

Modify as follows:

7.2.3.1.7 Coding of the REL

If the Reason header field with Q.850 Cause Value is included in the BYE or CANCEL request, then the Cause Value shall be mapped to the ISUP Cause Value field in the ISUP REL . The mapping of the Cause Indicators parameter to the Reason header is shown in Table 8a. Table 8 shows the coding of the Cause Value in the REL if it is not available from the Reason header field. In both cases, the Location Field shall be set to "*network beyond interworking point*".

Table 8: Coding of REL

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

Table 8a – Mapping of SIP Reason header fields into Cause Indicators parameter

Component of SIP Reason header field	Component value	BICC/ISUP Parameter field	Value
Protocol	"Q.850"	Cause Indicators parameter	–
protocol-cause	"cause = XX" (NOTE 1)	Cause Value	"XX" (NOTE 1)
–	–	Location	"network beyond interworking point"

NOTE 1: "XX" is the Cause Value as defined in ITU-T Rec. Q.850.

Table 8b : Mapping of PSTN XML elements with ISUP Parameters

BYE or CANCEL →	REL →	
PSTN XML	ISUP Parameter	Content
ProgressIndicator	Access Transport Parameter	Progress indicator
HighLayerCompatibility		High layer compatibility (Note)
LowLayerCompatibility		Low layer compatibility
HighLayerCompatibility	User Tele Service	High layer compatibility (Note)

NOTE: If two high layer compatibility information elements are received, they are transferred in the same order as received in the INVITE message in the access transport parameter of the initial address message.

Clause 7.2.3.1.8

Modify as follows:

7.2.3.1.8 Receipt of the Release Message

If the REL message is received and a final response (i.e. 200 OK (INVITE)) has already been sent, the I-MGCF shall send a BYE message.

NOTE: According to SIP procedures, 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 request has been received) on the incoming side of the I-MGCF then the I-MGCF does not send a 487 Request terminated response and instead waits until the ACK request is received before sending a BYE message.

If the REL message is received and the final response (i.e. 200 OK (INVITE)) has not already been sent, the I-MGCF shall send a Status-Code 4xx (Client Error) or 5xx (Server Error) response. The Status code to be sent is determined by examining the Cause code value received in the REL message. Table 9 specifies the mapping of the cause code values, as defined in ITU-T Recommendation Q.850 [38], to SIP response status codes. Cause code values not appearing in the table shall have the same mapping as the appropriate class defaults according to ITU-T Recommendation Q.850 [38].

Table 9: Receipt of the Release message (REL)

←SIP Message	← REL
Status code	Cause parameter
404 Not Found	Cause value No. 1 (unallocated (unassigned) number)
500 Server Internal error	Cause value No 2 (no route to network)
500 Server Internal error	Cause value No 3 (no route to destination)
500 Server Internal error	Cause value No. 4 (Send special information tone)

←SIP Message	← REL
Status code	Cause parameter
404 Not Found	Cause value No. 5 (Misdialed 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)
480Temporarily unavailable	Cause value No 21 (call rejected)
410 Gone	Cause value No 22 (number changed)
433 Anonymity Disallowed.(NOTE 1)	Cause value No. 24 (call rejected due to ACR supplementary service)
480 Temporarily unavailable	Cause value No 25 (Exchange routing error)
502 Bad Gateway	Cause value No 27 (destination out of order)
484 Address Incomplete	Cause value No. 28 invalid number format (address incomplete)
500 Server Internal error	Cause value No 29 (facility rejected)
480 Temporarily unavailable	Cause value No 31 (normal unspecified) (class default) (NOTE 2)
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)
500 Server Internal error	Cause value in the Class 010 (resource unavailable, Cause value No's. 38, 41, 42, 43, 44, & 47) (47 is class default)
500 Server Internal error	Cause value No 50 (requested facility no subscribed)
500 Server Internal error	Cause value No 57 (bearer capability not authorised)
500 Server Internal error	Cause value No 58 (bearer capability not presently)
500 Server Internal error	Cause value No 63 (service option not available, unspecified) (class default)
500 Server Internal error	Cause value in the Class 100 (service or option not implemented, Cause value No's. 65, 70 & 79) 79 is class default
500 Server Internal error	Cause value No 88 (incompatible destination)
404 Not Found	Cause value No 91 (invalid transit network selection)
500 Server Internal error	Cause value No 95 (invalid message) (class default)
500 Server Internal error	Cause value No 97 (Message type non-existent or not implemented)
500 Server Internal error	Cause value No 99 (information element/parameter non-existent or not implemented))
480 Temporarily unavailable	Cause value No. 102 (recovery on timer expiry)
500 Server Internal error	Cause value No 110 (Message with unrecognised Parameter, discarded)
500 Server Internal error	Cause value No. 111 (protocol error, unspecified) (class default)
480 Temporarily unavailable	Cause value No. 127 (interworking unspecified) (class default)

← SIP Message	← REL
Status code	Cause parameter
NOTE 1: Anonymity Disallowed, draft-ietf-sip-acr-code-02 [77] refers NOTE 2: Class 1 and class 2 have the same default value.	

A Reason header field containing the received (Q.850) Cause Value of the REL shall be added to the SIP final response or BYE request sent as a result of this clause. The mapping of the Cause Indicators parameter to the Reason header is shown in Table 9a.

Table 9a – Mapping of Cause Indicators parameter into SIP Reason header fields

Cause indicators parameter field	Value of parameter field	component of SIP Reason header field	component value
–	–	protocol	"Q.850"
Cause Value	"XX" (NOTE 1)	protocol-cause	"cause = XX" (NOTE 1)
–	–	reason-text	FFS
NOTE 1: "XX" is the Cause Value as defined in ITU-T Rec. Q.850.			

Table 9aa : ISUP Parameters with Mapping of PSTN XML elements

←4xx,5xx,6xx PSTN XML	←REL	
	ISUP Parameter	Content
ProgressIndicator	Access Transport Parameter	Progress indicator
HighLayerCompatibility		High layer compatibility (Note)
LowLayerCompatibility		Low layer compatibility
NOTE: If two high layer compatibility information elements are received, they are transferred in the same order as received in the INVITE message in the access transport parameter of the initial address message.		

Add Clause 7.2.3.2.1.5

7.2.3.2.1.5 Fallback (optional)

When the IAM includes a TMR and a TMR prime parameter and a USI and USI Prime parameter then the O-MGCF shall:

- Map the USI Prime into the second Bearer Capability stated in the XML Bearer Capability element and
- For the TMR and USI Prime the mapping shall apply according to the procedures described within clause 7.2.3.2.2.2 for the first offered codec.
- Map the USI into the first Bearer Capability stated in the XML Bearer Capability element and
- For the TMR prime and USI the mapping shall apply according to the procedures described within clause 7.2.3.2.2.2 for the second offered codec.

NOTE: If the Fallback mechanism is not supported the ISUP Fallback procedures apply.

In cases where the fallback appears within the terminating entity and sends back a SDP Answer that is equivalent with the TMR prime codec (fallback codec) then the O-MGCF shall only apply the cut-through to the fallback codec.

In cases where preconditions are used the O-MGCF has to wait for the SDP answer where the preconditions are met and fallback codec is sent back.

Clause 7.2.3.2.2

Modify as follows:

7.2.3.2.2.2 *SDP Media Description*

The SDP media description will contain precondition information as per RFC 3312 [37]. Depending on the coding of the continuity indicators different precondition information (RFC 3312 [37]) is included. If the continuity indicator indicates "continuity performed on a previous circuit" or "continuity required on this circuit", and the INVITE is sent before receiving a COT, then the O-MGCF shall indicate that the preconditions are not met. Otherwise the MGCF shall indicate whether the preconditions are met, dependent on the possibly applied resource reservation within the IMS.

If the O-MGCF determines that a speech call is incoming, the O-MGCF shall include the AMR codec transported according to RFC 3267 [23] with the options listed in clause 5.1.1 of 3GPP TS 26.236 [32] in the SDP offer, unless the Note below applies. Within the SDP offer, the O-MGCF should also provide SDP RR and RS bandwidth modifiers specified in IETF RFC 3556 [59] to disable RTCP, as detailed in Clause 7.4 of 3GPP TS 26.236 [32]. The O-MGCF may include other codecs according to operator policy.

NOTE: If the O-MGCF is deployed in an IMS network that by local configuration serves no user equipment that implements the AMR codec, then the AMR codec may be excluded from the SDP offer.

To avoid transcoding or to support non-speech services, the O-MGCF may add media derived from the incoming ISUP information according to Table 10b. The support of the media listed in Table 10b is optional.

If the XML BearerCapability element is supported the mapping of the ISUP USI Parameter shall be mapped to the XML BeareCapability element as shown in Table 17b.

Add Clause 7.2.3.2.2.7

7.2.3.2.2.7 PSTN XML elements

Table 17b : Mapping of ISUP Parameters with PSTN XML elements

IAM →		INVITE →
ISUP Parameter	Content	PSTN XML
<u>Access Transport Parameter</u>	<u>Progress indicator</u>	<u>ProgressIndicator</u>
	<u>High layer compatibility (Note 1)</u>	<u>HighLayerCompatibility</u>
	<u>Low layer compatibility</u>	<u>LowLayerCompatibility</u>
<u>User Service Information</u>		<u>Bearer Capability</u>
<u>User Tele Service</u>	<u>High layer compatibility (Note)</u>	<u>HighLayerCompatibility</u>
NOTE 1: <u>If two high layer compatibility information elements are received, they are transferred in the same order as received in the INVITE message in the access transport parameter of the initial address message.</u>		
NOTE 2: <u>In the normal case, the High layer compatibility information in the ATP is equal to the High layer compatibility in the User Tele Service parameter. It is network dependent which information is mapped into the XML instance in the INVITE. In the XML instance, no two identical High layer compatibility information are present.</u>		

Add Clause 7.2.3.2.2.8

7.2.3.2.2.8 Progress indicator

Table 17A – Coding of the progress indicator

IAM →		Access transport parameter	INVITE →
Forward call indicators parameter			PSTN XML body with Progress indicator X
<u>ISDN User Part indicator</u>	<u>ISDN access indicator</u>		
0 <i>(ISDN User Part not used all the way)</i>	Value non-significant	Value non-significant	no mapping
1 <i>(ISDN User Part used all the way)</i>	0 <i>(originating access non-ISDN)</i>	Value non-significant	No. 3
1 <i>(ISDN User Part used all the way)</i>	1 <i>(originating access ISDN)</i>	p.i. No. x	No. x

Add Clause 7.2.3.2.5.3

7.2.3.2.5.3 Access Transport Parameter

Table 17c : Mapping of PSTN XML elements with ISUP Parameters

ISUP Parameter	← ACM	← 180
	Content	PSTN XML
<u>Access Transport Parameter</u>	<u>Progress indicator</u>	<u>ProgressIndicator</u>
	<u>High layer compatibility (Note)</u>	<u>HighLayerCompatibility</u>
	<u>Low layer compatibility</u>	<u>LowLayerCompatibility</u>
<u>User Tele Service</u>	<u>High layer compatibility (Note)</u>	<u>HighLayerCompatibility</u>
NOTE: If two high layer compatibility information elements are received, they are transferred in the same order as received in the INVITE message in the access transport parameter of the initial address message.		

Add Clause 7.2.3.2.5.4

7.2.3.2.5.4 Progress indicator

Table 7.2.3.2.5.4-1 Handling of of the progress indicator

← ACM	← 183
Access transport parameter	PSTN XML body with Progress indicator X
p.i. No. x	p.i. No. x

Table 7.2.3.2.5.4-2 Handling of of the progress indicator

← ACM	← 180
Access transport parameter	PSTN XML body with Progress indicator X
p.i. No. x	p.i. No. x

Add Clause 7.2.3.2.6.1

7.2.3.2.6.1 Handling of of the progress indicator

Table 7.2.3.2.5.6-1 Handling of of the progress indicator

← CPG	← 183
Access transport parameter	PSTN XML body with Progress indicator X
p.i. No. x	p.i. No. x

Table 7.2.3.2.5.6-1 Handling of of the progress indicator

←CPG	←180
<u>Access transport parameter</u>	<u>PSTN XML body with Progress indicator X</u>
p.i. No. x	p.i. No. x

Add Clause 7.2.3.2.7.2

7.2.3.2.7.2 Access Transport Parameter

Received PSTN XML elements shall be mapped as shown in Table 17c

Add Clause 7.2.3.2.9.2

7.2.3.2.9.2 Access Transport Parameter

Received PSTN XML elements shall be mapped as shown in Table 17c

Add Clause 7.2.3.2.11.2

7.2.3.2.11.2 Access Transport Parameter

Received PSTN XML elements shall be mapped as shown in Table 17c

Clause 7.2.3.2.11.2

Modify as follows

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

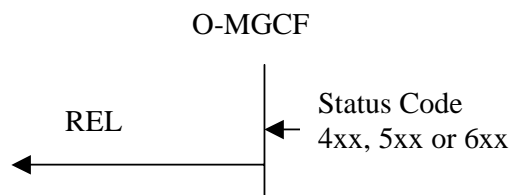


Figure 21: Receipt of Status codes 4xx, 5xx or 6xx

If a Reason header is included in a 4XX, 5XX, 6XX response, then the Cause Value of the Reason header shall be mapped to the ISUP Cause Value field in the ISUP REL message. The mapping of the Reason header to the Cause Indicators parameter is shown in Table 8a (see 7.2.3.1.7). Otherwise coding of the Cause parameter value in the REL message is derived from the SIP Status code received according to table 18. The Cause Parameter Values are defined in ITU-T Recommendation Q.850 [38].

In all cases where SIP itself specify additional SIP side behaviour related to the receipt of a particular INVITE response these procedures should be followed in preference to the immediate sending of a REL message to BICC/ISUP.

If there are no SIP side procedures associated with this response, the REL shall be sent immediately.

NOTE: If an optional Reason header is included in a 4XX, 5XX, 6XX, then the Cause Value of the Reason header can be mapped to the ISUP Cause Value field in the ISUP REL message. The mapping of the optional Reason header to the Cause Indicators parameter is out of the scope of the present specification.

NOTE Depending upon the SIP side procedures applied at the O-MGCF it is possible that receipt of certain 4xx/5xx/6xx responses to an INVITE may in some cases not result in any REL message being sent to the BICC/ISUP network. For example, if a 401 Unauthorized response is received and the O-MGCF successfully initiates a new INVITE containing the correct credentials, the call will proceed.

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

←REL (cause code)	←4xx/5xx/6xx SIP Message
127 (interworking unspecified)	400 Bad Request
127 (interworking unspecified)	401 Unauthorized
127 (interworking unspecified)	402 Payment Required
127 (interworking unspecified)	403 Forbidden
1 (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
24 (call rejected due to ACR supplementary service)	433 Anonymity Disallowed.(NOTE 1)
20 Subscriber absent	480 Temporarily Unavailable
127 (interworking unspecified)	481 Call/Transaction does not exist
127 (interworking unspecified)	482 Loop detected
127 (interworking unspecified)	483 Too many hops
28 (Invalid Number format)	484 Address Incomplete
127 (interworking unspecified)	485 Ambiguous
17 (User busy)	486 Busy Here
127 (Interworking unspecified) or not interworked. (NOTE 2)	487 Request terminated
127 (interworking unspecified)	488 Not acceptable here
127 (interworking unspecified)	493 Undecipherable

←REL (cause code)	←4xx/5xx/6xx SIP Message
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
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
NOTE 1: Anonymity Disallowed, draft-ietf-sip-acr-code-02 [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.	

Received PSTN XML elements shall be mapped as shown in Table 17c

Clause 7.2.3.2.13

Modify as follows

7.2.3.2.13 Receipt of a BYE

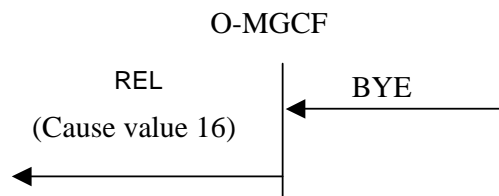


Figure 22: Receipt of BYE method

If a Reason header field with Q.850 Cause Value is included in the BYE request, then the Cause Value shall be mapped to the ISUP Cause Value field in the ISUP REL. The mapping of the Reason header to the Cause Indicators parameter is shown in Table 8a (see 7.2.3.1.7). On receipt of a BYE request, the O-MGCF sends a REL message with Cause Code value 16 (Normal Call Clearing).

Received PSTN XML elements shall be mapped as shown in Table 17c

Clause 7.2.3.2.14

Modify as follows

7.2.3.2.14 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-MGCF shall send a BYE request. If the final response (i.e. 200 OK (INVITE)) has not already been received the O-MGCF shall send a CANCEL method. A Reason header field containing the received (Q.850) Cause Value of the REL message shall be added to the CANCEL or BYE request. The mapping of the Cause Indicators parameter to the Reason header is shown in Table 9a (see 7.2.3.1.8).

Received PSTN XML elements shall be mapped as shown in Table 17c

Clause 7.4.21

Modify as follows:

7.4.21 User-to-User Signalling (UUS)

7.4.21.1 User-to-User Signalling (UUS) service 1 (implicit)

The coding of the User-user information element is described within ETSI EN 300 356-8 [10]. The User-to-User header is defined within ES 283 003 [3].

Note: If the draft-johnston-sipping-cc-uui is agreed then the Reference to ES 283 003 will be replaced by the agreed RFC.

Note: The IETF RFC needs more detail on encoding of the UUI as defined within ITU-T Q.763 i.e., the protocol discriminator and the encoding syntax.

The content of the uuidata field of the User-to-User header shall start with the first octet being the protocol discriminator and followed by the user information octets.

The format of the uuidata field shall be the hexadecimal representation of binary data coded in ascii alphanumeric characters. For example, the 8- bit binary value 0011- 1111 is 3F in hexadecimal. To code this in ascii, one 8- bit byte containing the ascii code for the character '3' (0011- 0011 or 033H) and one 8- bit byte containing the ascii code for the character 'F' (0100- 0110 or 046H) are required. For each byte value, the high-order hexadecimal digit is always the first digit of the pair of hexadecimal digits. The ascii letters used for the hex digits shall always be capital form.

For example:

User-to-User:

00C81031313232333334343535363637373838FA08303900064630E9E0;encoding=hex

Interworking procedures between the user-user information element and User-to-User header for the User-to-user signalling service 1 are defined in the following subsections.

Note: For 3GPP Release 8 the encoding of data should be described in more detail.

Note: For 3GPP Release 8 the interworking of the encoded data should be described in more detail.

7.4.21.1.1 Incoming Call Interworking from SIP to ISUP at I-MGCF

On the receipt of the User-to-User Header if the encoding header field parameter of the User-to-User header set to hex the I-MGCF shall map the content of the *uuidata* header field to the *protocol discriminator* and *user information* parameters of the *user-user* information element. Mapping procedures for other encoding header field values are or further study.

The *length of user-user contents* parameter shall be set by the I-MGCF according to the normal procedures.

The I-MGCF maps the messages transporting the user-user information according to the normal interworking procedures.

Table 7.4.21.1.1 Mapping of the User-to-User header to the ISUP user-to-user information parameter

<u>SIP parameter →</u>			<u>→ ISUP parameter</u>
<u>Source SIP header field and component</u>	<u>Source component value</u>	<u>ISUP Parameter field</u>	<u>Derived value of parameter field</u>
<u>User-to-User</u>	<u>uuidata</u>	<u>User-to-User</u>	<u>Protocol discriminator and User Information</u>

7.4.21.1.2 Outgoing Call Interworking from ISUP to SIP at O-MGCF

On the receipt of the user-user information element the O-MGCF shall map the protocol discriminator and user information parameters to the uuidata header field of the User-to-User header.

The O-MGCF shall set the encoding header field parameter of the User-to-User header to the “hex” value

The O-MGCF maps the messages transporting the user-user information according to the normal interworking procedures.

Table 7.4.21.1.1 Mapping of the ISUP user-to-user information parameter to the User-to-User header

	<u>→ ISUP parameter</u>		<u>→ SIP parameter</u>
<u>ISUP Parameter field</u>	<u>Source parameterfield</u>	<u>Source SIP header field and component</u>	<u>Derived value of parameter field</u>
<u>User-to-User</u>	<u>Protocol discriminator and User Information</u>	<u>User-to-User</u>	<u>uuidata</u>

7.4.21.1 User-to-User Signalling (UUS) service 1 (explicit)

The actions of the MGCF at the ISUP/BICC side are described in ITU-T Recommendation Q.737.1[42] under the clause "Interactions with other networks".

7.4.21.2 User-to-User Signalling (UUS) service 2 (implicit & explicit)

The actions of the MGCF at the ISUP/BICC side are described in ITU-T Recommendation Q.737.1[42] under the clause "Interactions with other networks".

7.4.21.3 User-to-User Signalling (UUS) service 3 (implicit & explicit)

The actions of the MGCF at the ISUP/BICC side are described in ITU-T Recommendation Q.737.1[42] under the clause "Interactions with other networks".

Clause 7.4.5

Modify as follows:

7.4.5 Sub-addressing (SUB)

~~The actions of the MGCF at the ISUP/BICC side are described in ITU-T Recommendation Q.731.8 [42] under the clause "Interactions with other networks".~~

7.4.5.1 General

The ISDN Subaddress in ISUP is transported within the Access Transport Parameter. The Coding of the Subaddress parameter within the Access Transport Parameter is described within ETSI EN 300 403-1 [xx]. The isdn-Subaddress parameter carried within a tel or sip URI is defined within RFC3966 [yy].

7.4.5.2 Incoming Call Interworking from SIP to ISUP at I-MGCF

The mapping in Table 24ba of the isdn-Subaddress parameter received within a tel or sip URI to the ISUP Access Transport Parameter encapsulating the Subaddress shall be applied.

The mapping in Table 24bb of the Subaddress received within an ANM Message containing the ISUP Access Transport Parameter to the isdn-Subaddress of a tel or sip URI to be sent within a 200 OK (INVITE) shall be applied.

Table 24ba Mapping of the Subaddress received in an initial INVITE to the Subaddress sent in the IAM

<u>SIP Message INVITE</u>		<u>ISUP Message IAM</u>	
<u>Source SIP header field and component</u>	<u>Source component value</u>	<u>ISUP Parameter field</u>	<u>Derived value of parameter field</u>
<u>Request-URI including the isdn-Subaddress</u>	<u>":isub=" 1*uric</u> <u>"uric" containing the Subaddress digits</u>	<u>Access Transport Parameter</u>	<u>called party Subaddress</u> <u>Note 1</u>
<u>P-Asserted-Identity header Field including the isdn-Subaddress</u>	<u>":isub=" 1*uric</u> <u>"uric" containing the Subaddress digits</u>	<u>Access Transport Parameter</u>	<u>calling party Subaddress</u> <u>Note 1</u>

NOTE 1: The Type of Subaddress as described within EN 300 403-1 [yy] Bits 5,6,7 and shall be set to 0 0 0 "NSAP (ITU-T Rec. X.213 [23] and ISO/IEC 8348 Add.2 [xx])"

Table 24bb Mapping of the Subaddress received in an ANM to the Subaddress sent in the 200 OK (INVITE)

<u>ISUP Message ANM</u>		<u>SIP 200 (OK)</u>	
<u>ISUP Parameter field</u>	<u>Source component value</u>	<u>Source SIP header field and component</u>	<u>Derived value of parameter field</u>
<u>connected party Subaddress</u>	<u>Subaddress encapsulated in the ISUP Access Transport parameter</u> <u>NOTE 1</u>	<u>P-Asserted-Identity including the isdn-Subaddress</u>	<u>":isub=" 1*uric</u> <u>The Subaddress digits included into the "uric" shall be derived from the Access Transport parameter</u> <u>Note 1</u>

NOTE1: The Type of Subaddress as described within EN 300 403-1 [yy] shall not be mapped

7.4.5.3 Outgoing Call Interworking from ISUP to SIP at O-MGCF

The mapping in Table 24bc of the isdn-Subaddress parameter received within a tel or sip URI to the ISUP Access Transport Parameter encapsulating the Subaddress shall be applied.

The mapping in Table 24bd of the Subaddress received within an ANM Message containing the ISUP Access Transport Parameter to the isdn-Subaddress of a tel or sip URI to be sent within a 200 OK (INVITE) shall be applied.

Table 24bc Mapping of the Subaddress received in an IAM to the Subaddress sent in the INVITE

<u>ISUP IAM Message</u>		<u>SIP INVITE Message</u>	
<u>ISUP Parameter field</u>	<u>Source component value</u>	<u>Source SIP header field and component</u>	<u>Derived value of parameter field</u>
<u>called party Subaddress</u>	<u>Access Transport parameter</u> Note 1	<u>Request-URI and To header field including the isdn-Subaddress</u>	<u>":isub=" 1*uric</u> <u>The Subaddress digits included into the "uric" shall be derived from the Access Transport parameter</u> Note 1
<u>calling party Subaddress</u>	<u>Access Transport parameter</u> Note 1	<u>P-Asserted-Identity header field and From header field including the isdn-Subaddress</u>	<u>":isub=" 1*uric</u> <u>The Subaddress digits included into the "uric" shall be derived from the Access Transport parameter</u> Note 1

NOTE 1: The "Type of Subaddress" as described within EN 300 403-1 [yy] shall not be mapped

Table 24bd Mapping of the Subaddress received in a 200OK to the Subaddress sent in the ANM

<u>200OK</u>		<u>ANM</u>	
<u>Source SIP header field and component</u>	<u>Source component value</u>	<u>ISUP Parameter field</u>	<u>Derived value of parameter field</u>
<u>P-Asserted-Identity including the</u>	<u>":isub=" 1*uric</u>	<u>connected party Subaddress</u>	<u>Access Transport parameter</u> Note 1

Note 1: The "Type of Subaddress as described" within EN 300 403-1 [yy] Bits 5,6,7 and shall be set to 0 0 0 "NSAP (ITU-T Rec. X.213 [23] and ISO/IEC 8348 Add.2 [xx])"

*Annex D (informative):
Bibliography*

draft-johnston-sipping-cc-uu-02 "Transporting User to User Information for Call Centers using SIP" 2007

History

Document history		
V0.0	January 2008	Initial version.
V0.1	January 2008	Textual error corrected. Input to MSF San Francisco meeting.
V0.2	March 2008	Straw ballot comment resolution – see msf2008.086.00.
V0.3	April 2008	Changes applied during ballot resolution at Shanghai.
V0.4	July 2008	Boiler plate text from msf2008.200 added.