3GPP TS 29.427 V7.1.0 (2011-09)

Technical Specification

**3rd Generation Partnership Project;
Technical Specification Group Core Network and Terminals;
Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN);
Endorsement of the SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks**
[3GPP TS 29.163 (Release 7), modified]
(Release 7)

GLOBAL SYSTEM FOR
MOBILE COMMUNICATIONS

Keywords

UMTS, interworking, ISUP, SIP

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Contents

Foreword

This Technical Specification (TS) was been produced by ETSI Technical Committee Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN) and originally published as ETSI ES 283 027 [25]. It was transferred to the 3rd Generation Partnership Project (3GPP) in December 2007.

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of the present document, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

Version x.y.z

where:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

3 or greater indicates TSG approved document under change control.

y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

z the third digit is incremented when editorial only changes have been incorporated in the document.

1 Scope

The present document provides the ETSI endorsement of the 3GPP TS 29.163 (V7.1.0) [1].

The clauses 7.2.3 and 7.4 of 3GPP TS 29.163 [1] specify the signalling interworking between ISDN User Part (ISUP) protocols and SIP in order to support services that can be commonly supported by ISUP and SIP-based network domains.

The clause 9.2.3.3.5 specifies how ringing tone is applied at the gateway when interworking ISUP and SIP.

The present document specifies the principles of interworking between the ETSI TISPAN IMS and ISUP based legacy CS networks, in order to support IM basic voice calls, therefore all the references to interworking between IMS and BICC are out of scope of the present document.

The present document is a protocol interworking specification; therefore all references to the user plane interworking are out of scope of the present document.

The present document specifies the interworking between ETSI SIP profile (as specified within ES 283 003 [3]) and ISUP, as specified in EN 300 356-1 [2] respectively.

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, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

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

NOTE: While any hyperlinks included in this clause were valid at the time of publication ETSI cannot guarantee their long term validity.

[1] 3GPP TS 29.163 (V7.1.0): "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] Void.

[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]".

[24] IETF RFC 5009: "Private Header (P-Header) Extension to the Session Initiation Protocol (SIP) for Authorization of Early Media".

[25] ETSI ES 283 027 V1.4.0: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Endorsement of the SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched CS) 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

The elements of 3GPP TS 29.163 [1] apply, 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 in table 1.

Table 1: Replacement of references in 3GPP TS 29.163

References in 3GPP TS 29.163 [1]	Modified references
ITU-T Recommendation Q.761	EN 300 356-1 [2]
ITU-T Recommendation Q.762	EN 300 356-1 [2]
ITU-T Recommendation Q.763	EN 300 356-1 [2]
ITU-T Recommendation Q.764	EN 300 356-1 [2]
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 [23]
ITU-T Recommendation Q.731.3	EN 300 356-3 [5]
ITU-T Recommendation Q.731.4	EN 300 356-4 [6]
ITU-T Recommendation Q.731.5	EN 300 356-5 [7]
ITU-T Recommendation Q.731.6	EN 300 356-6 [8]
ITU-T Recommendation Q.731.7	EN 300 356-11 [13]
ITU-T Recommendation Q.731.8	EN 300 356-10 [12]
ITU-T Recommendation Q.732.2	EN 300 356-15 [16]
ITU-T Recommendation Q.732.3	EN 300 356-15 [16]
ITU-T Recommendation Q.732.4	EN 300 356-15 [16]
ITU-T Recommendation Q.732.5	EN 300 356-15 [16]
ITU-T Recommendation Q.732.7	EN 300 356-14 [15]
ITU-T Recommendation Q.733.1	EN 300 356-17 [18]
ITU-T Recommendation Q.733.2	EN 300 356-16 [17]
ITU-T Recommendation Q.733.3	EN 300 356-18 [19]
ITU-T Recommendation Q.733.4	EN 300 356-7 [9]
ITU-T Recommendation Q.733.5	EN 300 356-20 [21]
ITU-T Recommendation Q.734.1	EN 300 356-12 [14]
ITU-T Recommendation Q.734.2	EN 300 356-19 [20]
ITU-T Recommendation Q.735.1	EN 300 356-9 [11]
ITU-T Recommendation Q.737.1	EN 300 356-8 [10]

Global modifications to 3GPP TS 29.163 V7.1.0 (clauses 7.2.3 and 7.4)

Clause 7.2.3.1.1

Modify as follows.

7.2.3.1.1 Sending of IAM

On reception of a SIP INVITE requesting an audio session, the I-MGCF shall send an IAM message.

An I-MGCF shall support both incoming INVITE requests containing SIP preconditions and 100rel extensions in the SIP Supported or Require headers, and INVITE requests not containing these extensions~~, unless the Note below applies.~~

~~NOTE: If the I-MGCF is deployed in an IMS network that by local configuration serves no user requiring preconditions, the MGCF may not support incoming requests requiring preconditions.~~

The I-MGCF shall interwork forked INVITE requests with different request URIs.

If a Continuity Check procedure is supported in the ISUP network, the I-MGCF shall send the IAM immediately after the reception of the INVITE, as shown in figure 3. This procedure applies when the value of the continuity indicator is either set to "continuity check required' or "continuity check performed on a previous circuit". If the continuity indicator is set to "continuity check required" the corresponding procedures at the Mn interface described in clause 9.2.2.3 also apply.

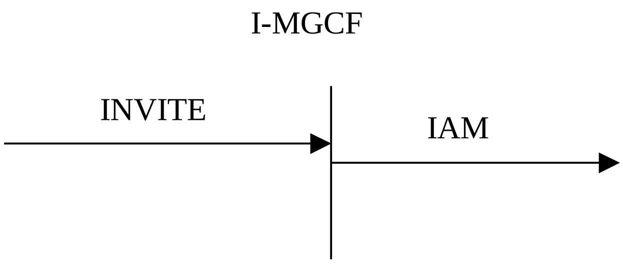


Figure 3: Receipt of an Invite request (continuity procedure supported in the ISUP network)

If no Continuity Check procedure is supported in the ISUP network, and the SDP in the received INVITE request contains preconditions not met, the I-MGCF shall delay sending the IAM until the SIP preconditions are met.

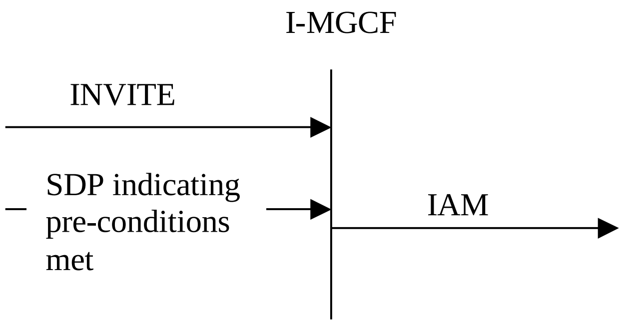


Figure 4: Receipt of an Invite request (continuity procedure not supported in the ISUP network)

The I-MGCF shall reject an INVITE request for a session only containing unsupported media types by sending a status code 500 "Server Internal error". If several media streams are contained in a single INVITE request, the I-MGCF shall select one of the supported media streams, reserve the codec(s) for that media stream, and reject the other media streams and unselected codecs in the SDP answer, as detailed in RFC 3264 [36]. If supported audio media stream(s) and supported non-audio media stream(s) are contained in a single INVITE request, an audio stream should be selected.

The I-MGCF shall include a To tag in the first backward non-100 provisional response, in order to establish an early dialog as described in RFC 3261 [19].

Clause 7.2.3.1.2.2

Modify as follows.

7.2.3.1.2.2 Nature of connection indicators

bits BA Satellite indicator

0 1 one satellite circuit in the connection

bits DC Continuity check indicator

0 0 *continuity check not required)* if the continuity check procedure is not supported in the succeeding network (figure 4).

0 1 *continuity check required*, if a continuity check shall be carried out on the succeeding circuit. (figure 3)

1 0 continuity check performed on a previous circuit otherwise, if the continuity check procedure is supported in the succeeding network, but shall not be carried out on the succeeding circuit otherwise. (figure 3)

bit E Echo control device indicator

~~1 outgoing echo control device included~~

The Echo control device indicator shall be set according to ISUP procedures.

NOTE: For certain calls e.g., TMR "64 kBit/s unrestricted", it is recommended to set the bit E to "outgoing echo control device indicator not included".

Clause 7.2.3.1.2.4

Modify as follows.

7.2.3.1.2.4 Calling party's category

~~0 0 0 0 1 0 1 0~~ ~~ordinary calling subscriber~~

See annex ZA for the normative interworking of the CPC parameter.

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"" or "speech". 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 [69] shall be mapped to the TMR "64 kbit/s unrestricted".

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

m= line			b= line (NOTE 4)	a= line	TMR parameter	USI parameter (optional) (NOTE 1)		HLC parameter (optional)
<media>	<transport>	<fmt-list>	<modifier>:<bandwidth-value> (NOTE 5)	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>	TMR codes	Information Transport Capability	User Information Layer 1 Protocol Indicator	High Layer Characteristics Identification
audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A	"3.1KHz audio"	~~"3.1KHz audio"~~	~~"G.711 □-law"~~	(NOTE 3)
audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMU/8000	"3.1KHz audio"	~~"3.1KHz audio"~~	~~"G.711 □-law"~~	(NOTE 3)
audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A	*"3.1KHz audio"*	~~*"3.1KHz audio"*~~	~~*"G.711 A-law"*~~	(NOTE 3)
audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMA/8000	*"3.1KHz audio"*	~~*"3.1KHz audio"*~~	~~*"G.711 A-law"*~~	(NOTE 3)
audio	RTP/AVP	9	AS: 64 kbit/s	rtpmap:9 G722/8000	*"64 kbit/s unrestricted"*	*"Unrestricted digital inf. w/tones/ann"*		
audio	RTP/AVP	Dynamic PT	AS: 64 kbit/s	rtpmap:<dynamic-PT> CLEARMODE/8000 (NOTE 2)	*"64 kbit/s unrestricted"*	*"Unrestricted digital information"*		
image	udptl	t38	N/A or up to 64 kbit/s	Based on T.38 [28]	*"3.1 KHz audio"*	*"3.1KHz audio"*		*"Facsímile Group 2/3"*
image	tcptl	t38	N/A or up to 64 kbit/s	Based on T.38 [28]	*"3.1 KHz audio"*	*"3.1KHz audio"*		*"Facsímile Group 2/3"*

NOTE 1 In this table the codec G.711 is used only as an example. Other codecs are possible.
NOTE 2 CLEARMODE is specified in RFC4040 [69].
NOTE 3 HLC is normally absent in this case. It is possible for HLC to be present with the value "Telephony", although 6.3.1/Q.939 indicates that this would normally be accompanied by a value of "Speech" for the Information Transfer Capability element.
NOTE 4 If the b=line indicates a bandwidth greater than 64kbit/s then the call may use compression techniques or reject the call with a 415 response indicating that only one media stream of 64kbit/s is supported.
NOTE 5 <bandwidth value> for <modifier> of AS is in units of kbit/s.

Clause 7.2.3.1.2.8

Modify as follows.

7.2.3.1.2.8 User service information

For coding of the USI see clause 7.2.3.1.2.5. ~~The Information Transfer Capability Information element is coded as "speech" or "3.1 kHz audio".~~

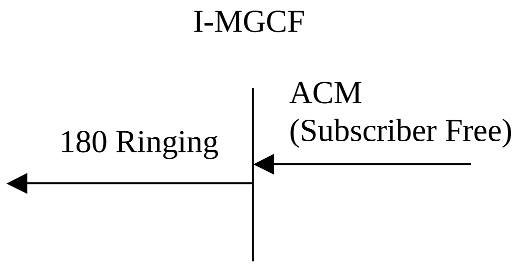
Clause 7.2.3.1.4

Modify as follows.

7.2.3.1.4 Sending of 180 ringing

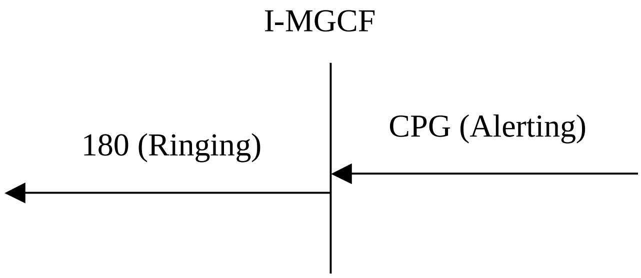
The I-MGCF shall send the SIP 180 Ringing when receiving any of the following messages:

- ACM with Called party's status indicator set to subscriber free (figure 6 is replaced by figures 6a and 6b).



~~**Figure 6: The receipt of ACM**~~

- CPG with Event indicator set to alerting (figure 7 is replaced by figures 7a and 7b).



~~**Figure 7: Receipt of CPG (Alerting)**~~

As a network option, if the INVITE request includes the P-Early-Media header and the I-MGCF supports the 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 provisional response including a P-Early-Media header, as defined in IETF RFC 5009 [24]; and
- the most recently sent P-Early-Media header authorizes early media.

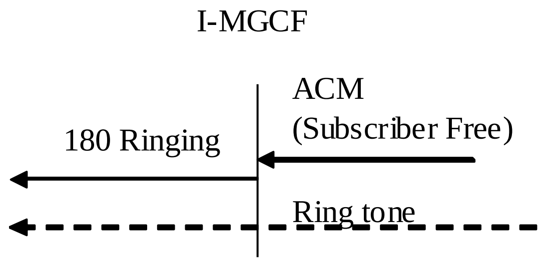


Figure 6a: The receipt of ACM without support of the P-Early-Media header

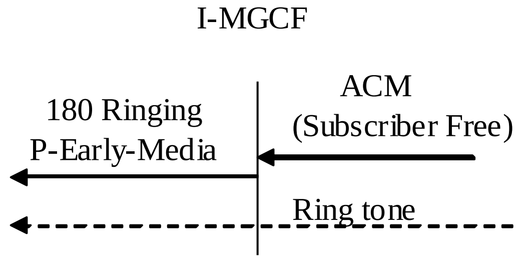


Figure 6b: The receipt of ACM with support of the P-Early-Media header

NOTE: If the I-MGCF signals the P-Early-Media header authorizing early media, then the SIP network can expect tones or announcements to the calling party to flow from the PSTN/ISDN 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 PSTN/ISDN.

- CPG with Event indicator set to alerting.

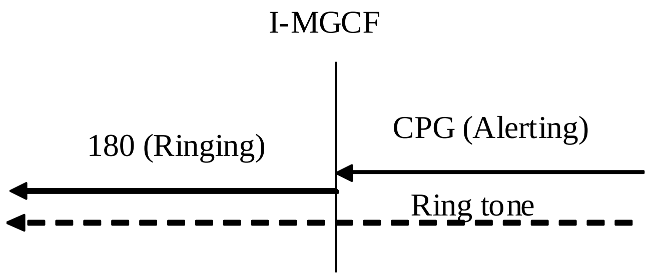


Figure 7a: Receipt of CPG (Alerting) without support of the P-Early-Media header

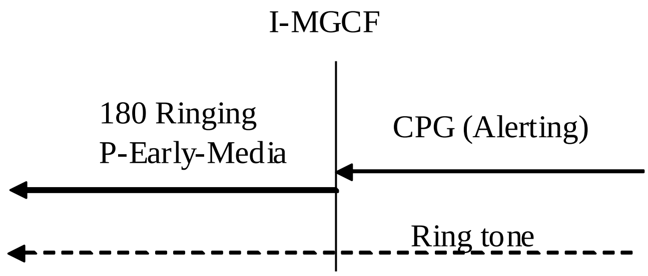


Figure 7b: Receipt of CPG (Alerting) with support of the P-Early-Media header

Add clause 7.2.3.1.4a.

7.2.3.1.4a Sending of 183 Session Progress for early media scenarios

As a network option, if the I-MGCF supports the P-Early-Media header and has received the P-Early-Media header in the INVITE request, the I-MGCF shall include a P-Early-Media header authorizing early media in the first 18x response.

As a network option, upon receiving one of the following messages and if the I-MGCF supports the P-Early-Media header 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.

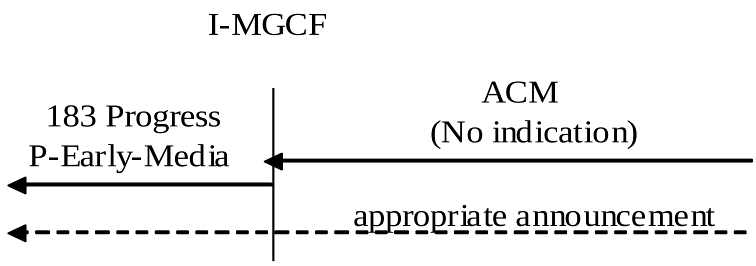


Figure 7c: Receipt of ACM No indication

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 *In-band info...* 2) **Backward call indicators parameter** ISDN User Part indicator 0 ISDN User Part not used all the way

- 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.

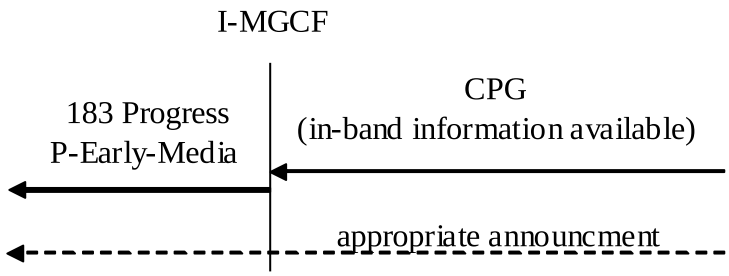


Figure 7d: Receipt of CPG (in-band information available)

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 (*progress*) **1) Optional backward call indicators parameter** In-band information indicator *In-band info ...* **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 if the USI sent in the IAM was coded 3.1 kHz audio.	

Clause 7.2.3.1.9a

Add clause 7.2.3.1.9a.

7.2.3.1.9a Receipt of REFER

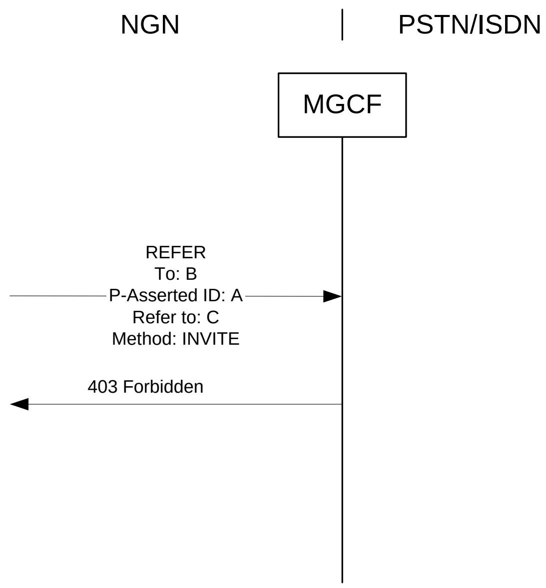


Figure 11a: Receipt of REFER method

A REFER received by the MGCF will always be rejected with a 403 Forbidden response.

Clause 7.2.3.2.1

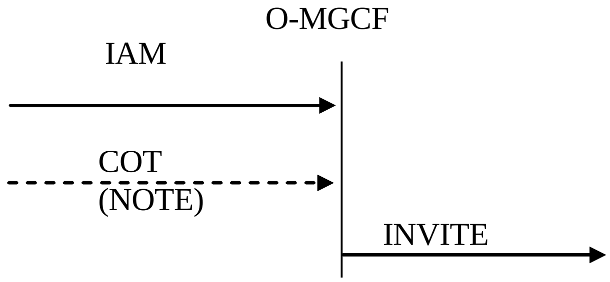
Modify as follows.

7.2.3.2.1 Sending of INVITE

An O-MGCF shall support both the SIP preconditions and 100 rel extensions and indicate the support of the SIP preconditions and 100rel extensions in the INVITE request.

~~NOTE: If the O-MGCF is deployed in an IMS network that by local configuration serves no user requiring preconditions, it may send the INVITE request without indicating support of preconditions.~~

If the Continuity Check indicator in the Nature of Connection Indicators parameter in the incoming IAM is set to indicate either '*continuity check required on this circuit*' or '*continuity check performed on previous circuit*', the O-MGCF may either defer sending the INVITE request until receiving a COT message or send the INVITE request without waiting for the COT.



NOTE: Waiting for the COT is a network option. Furthermore, it only applies if the Continuity Check indicator in the Nature of Connection Indicators parameter in the incoming IAM is set to indicate either '*continuity check required on this circuit*' or '*continuity check performed on previous circuit*'.

Figure 12: Receipt of an IAM (En bloc signalling in CS network)

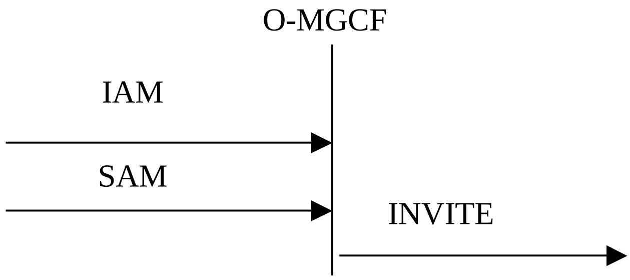


Figure 13: Receipt of an IAM (Overlap signalling in CS network)

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

The end of address signalling shall be determined by the earlier of the following criteria:

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) by observing that timer Ti/w1 has expired after the receipt of the latest address message and the minimum number of digits required for routing the call have been received.

If the end of the address signalling is determined in accordance with criteria a) b) or c), the timer Ti/w2 is started when INVITE is sent.

Add clause 7.2.3.2.2.5.

<u>7.2.3.2.2.5 P-Early-Media header</u>

<u>As a network option, if the O-MGCF supports the optional P-Early-Media header then it shall include the header without parameters in each outgoing INVITE request.</u>

Clause 7.2.3.2.4

Modify as follows.

7.2.3.2.4 Sending of ACM and awaiting answer indication

If the Address Complete Message (ACM) has not yet been sent, the following cases are possible trigger conditions that shall lead to the sending the address complete message (ACM).

- the detection of end of address signalling by the expiry of Timer T i/w_1; or

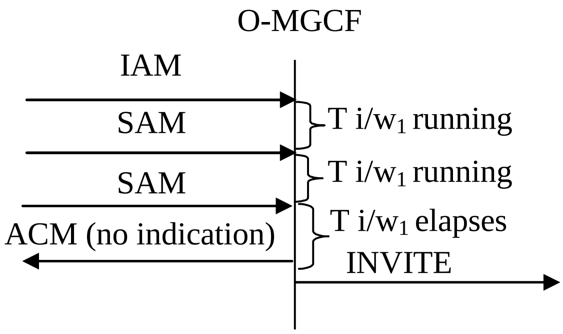


Figure 15: Sending of ACM T i/w_1 elapses

- the reception of the first 180 Ringing <u>response without a P-Early-Media header authorizing early media; or</u>

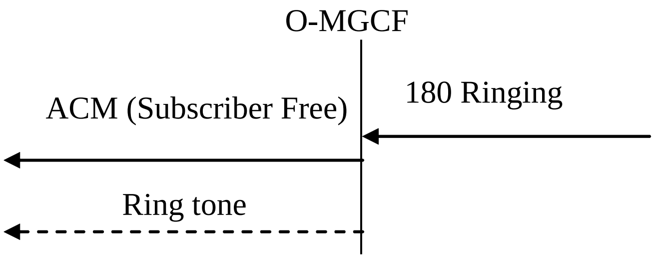
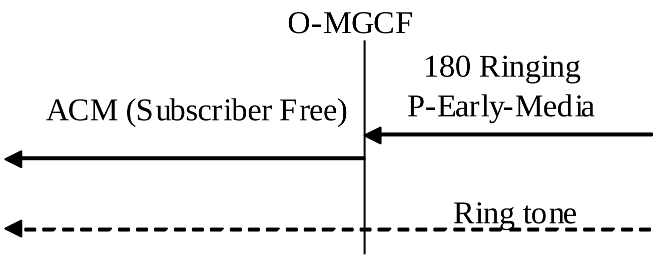


Figure 16: Sending of ACM (Receipt <u>of first 180 Ringing without authorization of early media</u>)

- <u>as a network option, the reception at an O-MGCF supporting the P-Early-Media header of the first 180 Ringing that includes a P-Early-Media header authorizing early media; or</u>



<u>**Figure 16a: Sending of ACM (Receipt of first 180 Ringing that includes authorization of early media)**</u>

<u>NOTE 1: Based on local knowledge that the call is transited to a PSTN network, the O-MGCF can make a decision not to generate the awaiting answer indication when receiving the 180 Ringing message without a P-Early-Media header.</u>

- <u>at an O-MGCF supporting the P-Early-Media header as a network option, once all the following sub-conditions have been met: {1} the reception at an O-MGCF supporting the P-Early-Media header, the first 183 Session Progress that includes a P-Early-Media header authorizing early media, {2} the SDP offer/answer procedures are completed, and {3} SDP preconditions are not used or applicable SDP preconditions have been met, or</u>

- as a network option, once all the following sub-conditions have been met: {1} the O-MGCF supporting the P-Early-Media header has received the first 183 Session Progress that includes a P-Early-Media header authorizing early media, and {2} SDP preconditions are not used or applicable SDP preconditions have been met; or

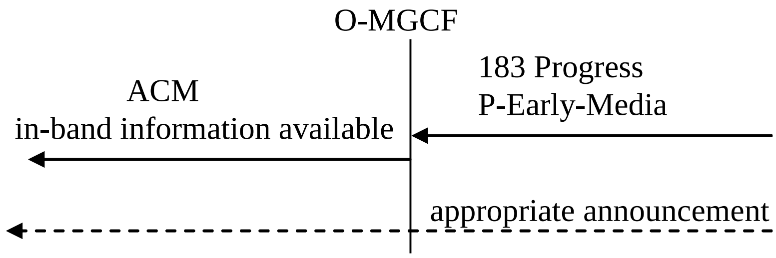


Figure 16b: Sending of ACM (Receipt of first 183 that includes authorization of early media)

- Ti/w 2 expires after the initial INVITE is sent.

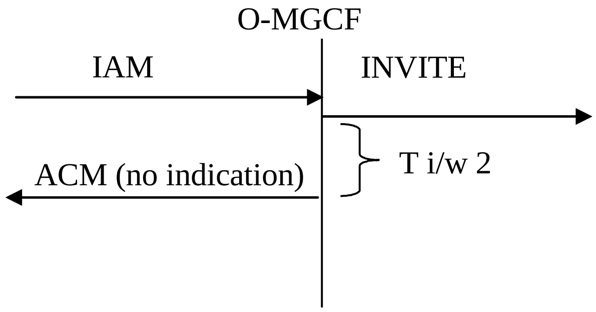


Figure 17: Sending of ACM (Ti/w$_2$ elapses)

The sending of an awaiting answer indication on speech or 3.1 kHz calls is described in clause 9.2.3.3.

- as a network option, at an O-MGCF supporting the P-Early-Media header, if the O-MGCF receives a 18x response with a P-Early-media header that changes the authorization of early media, the O-MGCF terminates the sending of the awaiting answer indication if the header authorizes early media, and initiates the sending of the awaiting answer indication if the header removes authorization of early media and if the O-MGCF has received the 180 Ringing response.

NOTE 2: Based on local knowledge that the call is transited to a PSTN network, the O-MGCF can make a decision not to generate the awaiting answer indication when receiving the 180 message without a P-Early-Media header.

Clause 7.2.3.2.5

Modify as follows.

7.2.3.2.5 Coding of the ACM

The description of the following ISDN user part parameters can be found in ITU-T Recommendation Q.763 [4].

7.2.3.2.5.1 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

00 no end-to-end method available

bit I Interworking indicator

1 interworking encountered

As a network operator option, the value I = 0 "no interworking encountered" is used for TMR = 64 kBit/s unrestricted

NOTE: This avoids sending of a progress indicator with Progress information 0 0 0 0 0 0 1 "Call is not end-to-end ISDN; further call progress information may be available in-band", so the call will not be released for that reason by an ISDN terminal.

bit J End-to-end information indicator

0 no end-to-end information available

bit K ISDN user part/~~BICC~~ indicator

0 ISDN user part not used all the way

As a network operator option, the value K = 1 "ISDN user part/~~BICC~~ used all the way" is used for TMR = 64 kBit/s unrestricted

NOTE: This avoids sending of a progress indicator with progress information 0 0 0 0 0 0 1 "Call is not end-to-end ISDN; further call progress information may be available in-band", so the call will not be released for that reason by an ISDN terminal.

bit L Holding indicator (national use)

0 holding not requested

bit M ISDN access indicator

0 terminating access non-ISDN

As a network operator option, the value M = 1 "terminating access ISDN" is used for TMR = 64 kBit/s unrestricted.

NOTE: This avoids sending of a progress indicator with progress information 0 0 0 0 0 1 0 " Destination access is non-ISDN", so the call will not be released for that reason by an ISDN terminal.

7.2.3.2.5.2 Optional Backward call indicators

Bit A 1 "in-band information or an appropriate pattern is now available" if 183 Session Progress response is received and a P-Early-Media header has been received authorizing early media.

Clause 7.2.3.2.6

Modify as follows.

7.2.3.2.6 Sending of the Call Progress message (CPG)

If the Address Complete Message (ACM) has already been sent, the O-MGCF shall send the Call Progress message (CPG) in the following cases: ~~when receiving the following message~~:

- Upon receipt of the SIP 180 Ringing provisional response without recent receipt of a P-Early-Media header authorizing early media; or

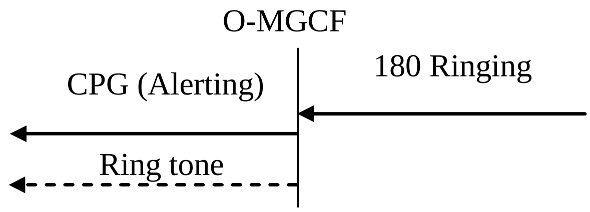


Figure 18: Sending of CPG(Alerting)
(Receipt of 180 Ringing response without authorization of early media)

- as a network option, upon receipt of the SIP 180 Ringing response and recent receipt of a P-Early-Media header authorizing early media at an O-MGCF supporting the P-Early-Media header; or

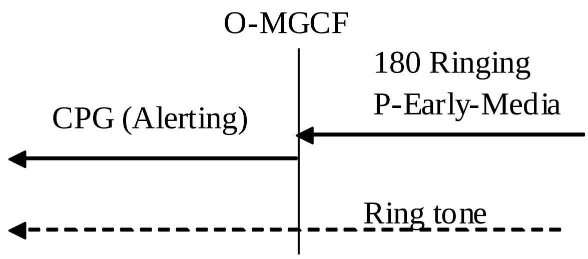


Figure 18a: Sending of CPG(Alerting)
(Receipt of 180 Ringing response and authorization of early media)

- as a network option, upon receipt the 183 Session Progress response and recent receipt of a P-Early-Media header authorizing early media at an O-MGCF supporting the P-Early-Media header.

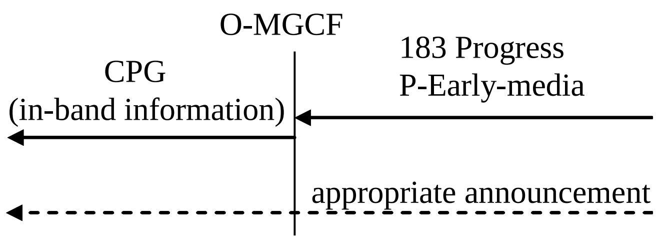


Figure 18b: Sending of CPG(in-band information available)

At an O-MGCF supporting the P-Early-Media header as a network option, if the O-MGCF receives a 18x response with P-Early-media header that changes the authorization of early media, the O-MGCF terminates the sending of the awaiting answer indication if the header authorizes early media and initiates the sending of the awaiting answer indication if the header removes authorization of early media and if the O-MGCF has received the 180 Ringing response.

NOTE: Based on local knowledge that the call is transited to a PSTN network, the O-MGCF can make a decision not to generate the awaiting answer indication when receiving the 180 message without a P-Early-Media header.

Clause 7.2.3.2.7.1

Modify as follows.

7.2.3.2.7.1 Event information

bits G-A Event indicator

0 0 0 0 0 0 1 *alerting if 180 Ringing response received*

0 0 0 0 0 1 1 in-band information or an appropriate pattern is now available", if 183 Session Progress response received and most recently received P-Early-Media header authorizes early media

Clause 7.2.3.3

Modify as follows.

7.2.3.3 Timers

Table 19: Timers for interworking

Symbol	Time-out value	Cause for initiation	Normal termination	At expiry	Reference
Ti/w1	4 s to 6 s (default of 4 s)	When last address message is received and the minimum number of digits required for routing the call have been received.	At the receipt of fresh address information.	Send INVITE, send the address complete message ~~and insert ring tone~~	7.2.3.2.1 7.2.3.2.4 (NOTE 1)
Ti/w2	4 s to 14 s (default of 4 s)	When INVITE is sent unless the ACM has already been sent.	On reception of 180 Ringing ,or 183 Session Progress and P-Early-Media header authorizing early media, or 404 Not Found or 484 Address Incomplete for an INVITE transaction for which Ti/w3 is running, or 200 OK (INVITE).	Send ACM (no indication) ~~and send the awaiting answer indication (e.g. ring tone) or appropriate progress announcement to the calling party.~~	7.2.3.2.4 7.2.3.2.1 (NOTE 2)
Ti/w3	4 to 6 seconds (default of 4 seconds)	On receipt of 404 Not Found or 484 Address Incomplete if there are no other pending INVITE transactions for the corresponding call.	At the receipt of SAM	Send REL with Cause Value 28 to the ~~BICC~~/ISUP side.	7.2.3.2.1A, 7.2.3.2.12.1 (NOTE 3)
NOTE 1: This timer is used when overlap signalling is received from ~~BICC~~/ISUP network and converted to en-block signalling at the MGCF. NOTE 2: This timer is used to send an early ACM if a delay is encountered in receiving a response from the subsequent SIP network. NOTE 3: This timer is known as the "SIP dialog protection timer". This timer is only used where the O-MGCF is configured to send INVITE before end of address signalling is determined.					

Clause 7.4.8

Modify as follows.

7.4.8 Explicit Call Transfer (ECT)

When the MGCF receives a FAC message with Generic notification indicator coded as "Call transfer active" and a CPG with Generic notification indicator coded as "Remote hold" was received previously for the current communication, the action described in table 24a applies. In all other cases the actions of the MGCF at the ISUP/~~BICC~~ side are described in ITU-T Recommendation Q.732.7 [42] under the clause "Interactions with other networks".

Table 24a: Mapping between ISUP and SIP for the Explicit Communication Transfer supplementary service

ISUP message	Mapping
FAC with a " call transfer, active " Generic notification indicator	As described for CPG message with a "remote retrieval" Generic notification indicator in Subclause 7.4.10.2

Clause 9.2.3.1.5

Modify as follows.

9.2.3.1.5 Called party alerting

The MGCF shall request the IM-MGW to provide an awaiting answer indication (ringing tone) to the calling party using the Send Tone procedure (signals 20 and 21 in figure 37) , when the ~~first of the~~following condition~~s~~ is satisfied:

- the MGCF receives the first 180 Ringing message
- ~~Timer T i/w₁ expires~~
- ~~Timer T i/w₂ expires~~

Clause 9.2.3.2.5

Modify as follows.

9.2.3.2.5 Called party alerting

The MGCF shall request the IM-MGW to provide an awaiting answer indication (ringing tone) to the calling party using the Send Tone procedure (signals 19 and 20 in figure 38) , when ~~the first of~~ the following condition~~s~~ is satisfied:

- the MGCF receives the first 180 Ringing message~~,~~

Clause 9.2.3.3.5

Modify as follows.

9.2.3.3.5 Called party alerting

The MGCF shall request the IM-MGW to provide an awaiting answer indication (ringing tone) to the calling party using the Send TDM Tone procedure (signals 19 and 20 in figure 39) , when the first of the following conditions is satisfied:

- the MGCF receives the first 180 Ringing message without authorization of early media.
- ~~Timer T i/w1 expires~~
- ~~Timer T i/w2 expires~~

Clause B.3.2.1.5

Modify as follows.

B.3.2.1.5 Called party alerting

The MGCF shall request the IM-MGW to provide an awaiting answer indication (ringing tone) to the calling party using the Send Tone procedure (signals 21 and 22 in figure B.2/1) , when the ~~first of the~~ following conditions~~s~~ is satisfied:

- the MGCF receives the first 180 Ringing message.

~~- Timer T i/w$_1$ expires~~

~~- Timer T i/w$_2$ expires~~

Annex ZA (normative): Interworking of Cpc parameter

ZA.1 Interworking SIP to ISUP

Table ZA.1.1 shows the mapping of a cpc parameter received in a P-Asserted-Identity header in the initial INVITE request to the Calling party's category parameter in the ISUP IAM.

Table ZA.1.1: Mapping of the cpc parameter to the ISUP Calling party's category parameter

SIP Parameter		ISUP Parameter
cpc received in a P-Asserted-Identity	**Accept-Contact "language"**	**Sent Calling party's category**
operator	French	operator, language French
operator	English	operator, language English
operator	German	operator, language German
operator	Russian	operator, language Russian
operator	Spanish	operator, language Spanish
ordinary		ordinary calling subscriber
priority		calling subscriber with priority
data		Data call (voice band data)
test		Test call
payphone		Payphone
cellular		mobile terminal located in the home PLMN
cellular-roaming		mobile terminal located in a visited PLMN
ieps		IEPS call marking for preferential call set up
NOTE: In case the cpc is absent or contains values that are not in this table then the ISUP shall contain the default cpc value "ordinary calling subscriber".		

ZA.2 Interworking ISUP to SIP

Table ZA.2.1 shows the mapping of a Calling party's category received in a ISUP with the cpc parameter within the regarding P-Asserted-Identity.

Table ZA.2.1: Mapping of the ISUP Calling party's category parameter to the cpc parameter

ISUP Parameter	SIP Parameter	
received calling party's category	**cpc sent in a P-Asserted-Identity**	**Accept-Contact "language"**
operator, language French	operator	French
operator, language English	operator	English
operator, language German	operator	German
operator, language Russian	operator	Russian
operator, language Spanish	operator	Spanish
ordinary calling subscriber	ordinary	
calling subscriber with priority	priority	
data call (voice band data)	data	
test call	test	
payphone	payphone	
mobile terminal located in the home PLMN	cellular	
mobile terminal located in a visited PLMN	cellular roaming	
IEPS call marking for preferential call set up	ieps	

Annex ZB (informative): Change history

TISPAN #	TISPAN Doc.	CR	Subject/Comment
11	11TD159	001	"CR to CPC parameters interworking"
11	11TD159	002	CR to Nature of connection indicators (outgoing Echo Device)
13bis	13bTD133	003r1	CR to ES 283 027 The interworking of the PSTN ECT service
void	void	004	Void
13	12tTD311r2	005	**WI 3087 authorization of early media discussion**
Void	Void	006	Void
13bis	13bTD471	007	Change of scope of ES 283 027.1.1.1
13bis	13bTD082r3	008	Mistake in the handling of sending ringing tone
13bis	13bTD486r1	009	WI 3087 correction of cpc interworking
13bis	13bTD233r1	010	Interworking of USI
13bis	13bTD273	011	Editorial corrections in the mapping between ACM/CPG and 18* message
13bis	13bTD271r2	012	Additional correction in the mapping between ACM/CPG and 18* message
13ter	13tTD236r3	013	Additional correction in the mapping between ACM and 183 message improving consensus reached in TISPAN 13bis

Annex ZZ (informative): Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2006-07					Publication as **ETSI ES 283 027**		1.1.1
2007-09					Publication as ETSI ES 283 027		1.4.0
2007-12					Conversion to **3GPP TS 29.427**		1.4.1
2008-05	CT#40	CP-080304			MCC updated to version 7.0.0 after approval at TSG CT#40	1.4.1	7.0.0
2011-09	CT#53	CP-110605	1		Updated of reference to RFC 5009	7.0.0	7.1.0