3GPP TSG CT Meeting #28 **CP-0500031**
1st – 3rd June 2005. Quebec, CANADA.

Source: **MCC**

Title: **All LSs sent from CT3 since TSG CN#27 Meeting**

Agenda item: **6.2.1**

Document for: **APPROVAL**

Introduction:

This document contains all the LSs APPROVED and sent by CT3 since the last CT Plenary.

Tdoc	Title	LS To	LS Cc	Attachment
C3-050424	LS on use of the Auth-Application-Id AVP	CT4	SA5, CT	
C3-050435	Charging Implications of SCUDIF	SA1	SA5 SWGB	
C3-050436	Reply LS on tracing information for MBMS services	SA5	CT4	
C3-050438	LS on IMS support of TISPAN NGN supplementary services	ETSI TISPAN	CT1, CT4, SA2	C3-050421, C3-050372, C3-050302
C3-050440	Reply LS on network-initiated SCUDIF support	RAN3		

3GPP TSG-CT WG3 Meeting #36 **C3-050424**
Cancun, Mexico. 25th - 29th April 2005.

Title:	**LS on use of the Auth-Application-Id AVP**
Release:	**Release 6**
Work Item:	**CH-FBC, QoS1, MBMS**
Source:	**CT3**
To:	**CT4**
Cc:	**SA5, CT**

Contact Person:

Name: J. Javier Pastor
E-mail Address: **j.javier.pastor(AT)ericsson(DOT)com**

Name: Javier Gonzalez Gallego
E-mail Address: **ggfj(AT)nortel(DOT)com**

---

1. Overall Description:

CT3 wants to inform that it has detected some problems in several Diameter commands defined within the IETF. These changes affect to the following CT3 interfaces: Gx, Gmb, Rx and Gq. This problem may also affect other interfaces for what CT3 is not responsible.

The issue is related to the design of the ABNF of some Diameter commands. There are some commands that include a mandatory AVP called Auth-Application-Id. According to the rules stated in Diameter Base Protocol (RFC-3588), all the commands should only contain one of the application identifiers defined (Auth-Application-Id, Acc-Application-Id or Vendor-Specific-Application-Id). The presence of a mandatory Auth-Application-Id AVP avoids the possibility to add the needed Vendor-Specific-Application-Id AVP to carry the vendor application identifiers assigned by IANA to the 3GPP proprietary applications.

To solve this issue, there have been deep discussions within CT3 and with the IETF AAA delegates of the companies involved. Finally there was an agreement to on the solution. The agreement includes the following:

- For the Capabilities-Exchange commands (CER, CEA) there is no issue, and the normal procedure should be applied using the Vendor-Specific-Application-Id AVP.

- For other commands (CCR, CCA, RAR, STR, ASR, AAR, AAA) it has been agreed to use the mandatory Auth-Application-Id AVP to carry the vendor application identifiers assigned by IANA to 3GPP.
  This solution allows keeping the mandatory AVPs included in the IETF command ABNF definition unchanged and therefore conserving the assigned command code

The reasons for this agreement can be summarized as follows:

- Diameter Base Protocol (DBP) states that only one AVP including the application identifier can be used at the same time.
- Commands used in DBP, NASREQ application and Credit Control application have the "Auth-Application-Id" AVP as a mandatory AVP, and does not include the "Vendor-Specific-Application-Id" AVP. However, there is a contradictory recommendation that a vendor specific application identifier should use the "Vendor-Specific-Application-Id" AVP in DBP.
- According to IETF policies, the standard Diameter commands must have their mandatory parameters unmodified and always being included in the message. Therefore it is only feasible to add optional AVPs to the already defined AVPs to keep the command code unchanged. Optional AVPs can be added by all the applications that use a particular standard command.

- Changing command codes has a significant cost and requires IETF consensus.
- The numbering space used for the assignment of the application identifiers is shared amongst Auth-Application-Id AVP, Acct-Application-Id AVP and Vendor-Specific-Application-Id AVP and therefore a collision between them is not possible. Then, the reuse of Auth-Application-Id to contain the 3GPP vendor specific application identifiers is feasible.

2. Actions:

To CT4 group.

ACTION:

- CT3 kindly recommends CT4 group to consider the solution agreed in CT3.
- CT3 kindly asks CT4 group to contact IETF to formally state the contradictions and problems described above and for IETF to consider it in their future work with Diameter protocol and applications.

To SA5 group.

ACTION: CT3 kindly recommends SA5 group to consider whether the issues apply to SA5 and if so to take into account the solution agreed in CT3.

3. Date of Next CT3 Meetings:

CT3_37 29th August - 2th September 2005 London

3GPP TSG-CT WG3 Meeting #36 **C3-050438**
Cancun, Mexico. 25th - 29th April 2005.

Title:	**LS on IMS support of TISPAN NGN supplementary services**
Response to:	**CRs C3-050421, C3-050372 and C3-050302 from T-Mobile on TS 29.163 support of TISPAN Reason header, ACR and CDIV**
Release:	**Release 7**
Work Item:	**IMS services via fixed broadband**
Source:	**CT3**
To:	**ETSI TISPAN**
Cc:	**CT1, CT4, SA2**

Contact Person:
Name: Richard Ejzak
Tel. Number: +1 630 979 7036
E-mail Address: ejzak@lucent.com

Attachments: C3-050421, C3-050372 and C3-050302.

---

1. Overall Description:

CT3 would like to thank ETSI TISPAN and T-Mobile for providing information regarding some of the TISPAN NGN supplementary services including suggested changes to TS 29.163 in support of these services. CT3 would like to provide specific comments and to ask specific questions regarding the three T-Mobile CRs to TS 29.163 on support of certain TISPAN NGN supplementary services.

CT1 has primary responsibility for the work item on IMS services via fixed broadband, and has begun work on TR 24.819, *Protocol impact from providing IMS services via fixed broadband*. CT3 kindly requests that TISPAN and supporting companies present to CT3 future requested changes to IMS interworking procedures towards CS networks in support of TISPAN NGN supplementary services in the form of CRs to clause 8 of TR 24.819.

Regarding C3-050421 on the addition of interworking of TISPAN Reason header, CT3 has the following comments and questions:

- There seems to be a potential for fraudulent behaviour by a UE generating a Reason header towards an MGCF. Does TISPAN share this concern? Does TISPAN have any suggestions on how to ensure that an MGCF only interworks authorized Reason headers? Which entities in the IMS architecture have a role in providing this authorization/policing function? It seems likely that the potential for fraud will also apply to other SIP extensions developed by TISPAN for other supplementary services. Has TISPAN considered the potential need for a more general security architecture to assure that supplementary services data is properly secured?
- Why is it required to fill the component value with text?
- Is it meaningful to indicate a cause value for internal error in the network to a user? More generally, since it seems that some reason codes are not appropriate for a UE, which reason code values should be "passed" to a UE? Which reason code values, if any, should be trusted when coming from a UE?

Regarding C3-050372 on the addition of interworking of TISPAN ACR simulation service, CT3 has the following comments and questions:

- For various reasons, the originating side in an IMS session may not play out an audible ACR indication in early media from an I-MGCF. How does this impact the provision of the ACR service?
- For similar reasons, an O-MGCF may not be able to play out an audible ACR indication in early media from IMS. In addition, early media could not come directly from a terminating UE, so a special application server would be required to provide audible ACR indication. Has TISPAN considered the use of the SIP Alert-Info header as an alternative way of providing audible ACR indication without requiring full support of early media?

- ACM with ACR indication is not mapped to 183 response by TS 29.163, and neither is a 183 response of any kind mapped to an ACM. When mapping an ACM with ACR indication to a 183 response, is the intent to “piggyback” the ACR indication onto a 183 response in a typical IMS call flow or to add a new 183 response message to the flow?

Regarding C3-050302 on the addition of interworking of TISPAN CDIV simulation service, CT3 has the following comments and questions:

- Is there ever a case when information in a History header should not be passed to a UE during the normal course of establishing a SIP session with a UE?
- The proposed text for TS 29.163 does not include any reference to the sending or receiving of a SIP 302 redirection response message at an MGCF. What role, if any, does the 302 response have with regard to the CDIV service? Should the 302 response interoperate with the CDIV service and which IMS entities are involved?

C3-050372 on the TISPAN ACR simulation service was not seen as technically stable enough for TR 29.819. CT3 has agreed to CR C3-050421 against TR 29.819 on the TISPAN Reason header. C3-050302 was noted with limited discussion, pending receipt of a version agreed by TISPAN.

2. Actions:

To ETSI TISPAN.

ACTION: CT3 asks ETSI TISPAN to consider the comments expressed in this liaison and to also provide responses to the questions above in time for the next CT3 meeting CT3_37.

3. Date of Next CT3 Meetings:

CT3_37 29th August - 2nd September 2005 London

3GPP TSG-CT WG3 Meeting #36 **Tdoc ⌘DocNumber**
Cancun, Mexico. 25th - 29th April 2005.

CR-Form-v7.1

CHANGE REQUEST

⌘	TS29.163	CR	063	⌘ rev	-	⌘	Current version:	6.6.0	⌘

*For **HELP** on using this form, see bottom of this page or look at the pop-up text over the ⌘ symbols.*

Proposed change affects: UICC apps⌘ [ ] ME [X] Radio Access Network [ ] Core Network [X]

Title:	⌘	Addition of Interworking of TISPAN CDIV (Communication Diversion) simulation service
Source:	⌘	T-Mobile
Work item code:	⌘	
Date:	⌘	12.04.2005
Category:	⌘	**B**
Release:	⌘	Rel-7

Use one of the following categories:
F *(correction)*
A *(corresponds to a correction in an earlier release)*
B *(addition of feature),*
C *(functional modification of feature)*
D *(editorial modification)*
Detailed explanations of the above categories can be found in 3GPP TR 21.900.

Use one of the following releases:
Ph2 (GSM Phase 2)
R96 (Release 1996)
R97 (Release 1997)
R98 (Release 1998)
R99 (Release 1999)
Rel-4 (Release 4)
Rel-5 (Release 5)
Rel-6 (Release 6)
Rel-7 (Release 7)

Reason for change:	⌘	For supporting the TISPAN NGN simulation service “Communication Diversion ” in the TISPAN IMS the interworking between IMS and PSTN/ISDN is needed Due to the fac that this service will be discussed at a TISPAN simulation service meeting (20-22 April) there could be appear changes that will be brought into 3GPP as a Revised document.
Summary of change:	⌘	Subsection 7.4.6 and 7.47 modified
Consequences if not approved:	⌘	The interworking of CDIV can not be supported

Clauses affected:	⌘	7.4.6 and 7.4.7				
		Y	**N**			
Other specs affected:	⌘	X		Other core specifications	⌘	TS24.229 (History /Reason Header)
			X	Test specifications		
			X	O&M Specifications		
Other comments:	⌘					

How to create CRs using this form:

Comprehensive information and tips about how to create CRs can be found at http://www.3gpp.org/specs/CR.htm. Below is a brief summary:

1) Fill out the above form. The symbols above marked ⌘ contain pop-up help information about the field that they are closest to.

2) Obtain the latest version for the release of the specification to which the change is proposed. Use the MS Word "revision marks" feature (also known as "track changes") when making the changes. All 3GPP specifications can be downloaded from the 3GPP server under ftp://ftp.3gpp.org/specs/ For the latest version, look for the directory name with the latest date e.g. 2001-03 contains the specifications resulting from the March 2001 TSG meetings.

3) With "track changes" disabled, paste the entire CR form (use CTRL-A to select it) into the specification just in front of the clause containing the first piece of changed text. Delete those parts of the specification which are not relevant to the change request.

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*.

[1] ITU-T Recommendation G.711: "Pulse Code Modulation (PCM) of voice frequencies".

[2] ITU-T Recommendation H.248.1 (2002): "Gateway control protocol: Version 2".

[3] ITU-T Recommendation Q.701 to Q.709: "Functional description of the message transfer part (MTP) of Signalling System No. 7".

[4] ITU-T Recommendations Q.761to Q.764 (2000): "Specifications of Signalling System No.7 ISDN User Part (ISUP)".

[5] Void.

[6] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".

[7] Void.

[8] 3GPP TS 24.228: "Signalling flows for the IP multimedia call control based on SIP and SDP".

[9] 3GPP TS 24.229: "IP Multimedia Call Control Protocol based on SIP and SDP".

[10] 3GPP TS 23.002: "Network Architecture".

[11] 3GPP TS 22.228: "Service requirements for the IP Multimedia Core Network Subsystem".

[12] 3GPP TS 23.228: "IP Multimedia subsystem (IMS)".

[13] Void.

[14] 3GPP TS 29.205: "Application of Q.1900 series to Bearer Independent CS Network architecture; Stage 3".

[15] 3GPP TS 29.332: "Media Gateway Control Function (MGCF) – IM-Media Gateway (IM-MGW) interface, Stage 3".

[16] IETF RFC 791: "Internet Protocol".

[17] IETF RFC 768: "User Datagram Protocol".

[18] IETF RFC 2960: "Stream Control Transmission Protocol".

[19] IETF RFC 3261: "SIP: Session Initiation Protocol".

[20] 3GPP TS 29.202: "Signalling System No. 7 (SS7) signalling transport in core network; Stage 3".

[21] IETF RFC 2474: "Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers".

[22] IETF RFC 2475: "An Architecture for Differentiated Services".

[23] IETF RFC 3267: "Real-Time Transport Protocol (RTP) payload format and file storage format for the Adaptive Multi-Rate (AMR) Adaptive Multi-Rate Wideband (AMR-WB) audio codecs".

[24] IETF RFC 793: "Transmission Control Protocol".

[25] 3GPP TS 29.414: "Core network Nb data transport and transport signalling".

[26] 3GPP TS 29.415: "Core network Nb interface user plane protocols".

[27] 3GPP TS 23.205: "Bearer-independent circuit-switched core network; Stage 2".

[28] Void.

[29] ITU-T Recommendation Q.2150.1: "Signalling transport converter on MTP3 and MTP3b".

[30] ITU-T Recommendations Q.1902.1 to Q.1902.6 (07/2001): "Bearer Independent Call Control".

[31] ITU-T Recommendation Q.1950 (2002): "Bearer independent call bearer control protocol".

[32] 3GPP TS 26.236: "Packet switched conversational multimedia applications; Transport protocols".

[33] 3GPP TS 29.232: "Media Gateway Controller (MGC) – Media Gateway (MGW) interface; Stage 3".

[34] IETF RFC 2833: "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals".

[35] ITU-T Recommendation Q.765.5: "Signalling system No. 7 – Application transport mechanism: Bearer Independent Call Control (BICC)".

[36] IETF RFC 3264: "An Offer/Answer Model with the Session Description Protocol (SDP)".

[37] IETF RFC 3312: "Integration of Resource Management and Session Initiation Protocol (SIP)".

[38] ITU-T Recommendation Q.850 (1998): "Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part".

[39] IETF RFC 2460: "Internet Protocol, Version 6 (IPv6) Specification"

[40] IETF RFC 3323: "A Privacy Mechanism for the Session Initiation Protocol (SIP)".

[41] IETF RFC 3325: "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks".

[42] ITU-T Recommendation Q.730 to Q.737 (12/1999): "ISDN user part supplementary services".

[43] ITU-T Recommendation I.363.5 (1996): "B-ISDN ATM Adaptation Layer specification: Type 5 AAL".

[44] ITU-T Recommendation Q.2110 (1994): "B-ISDN ATM adaptation layer - Service Specific Connection Oriented Protocol (SSCOP)".

[45] ITU-T Recommendation Q.2140 (1995): "B-ISDN ATM adaptation layer - Service specific coordination function for signalling at the network node interface (SSCF AT NNI)".

[46] ITU-T Recommendation Q.2210 (1996): "Message transfer part level 3 functions and messages using the services of ITU-T Recommendation Q.2140".

[47] 3GPP TS 23.221: "Architectural requirements".

[48] ITU-T Recommendation E.164 (05/1997): "The international public telecommunication numbering plan".

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

[50] 3GPP TS 26.102: "Adaptive Multi-Rate (AMR) speech codec; Interface to Iu and Uu".

[51] IETF RFC 3550: "RTP: A Transport Protocol for Real-Time Applications".

[52] IETF RFC 3551: "RTP Profile for Audio and Video Conferences with Minimal Control".

[53] IETF RFC 3555: "MIME Type Registration of RTP Payload Formats".

[54] IETF RFC 3262: "Reliability of provisional responses".

[55] IETF RFC 3311: "SIP UPDATE method".

[56] IETF RFC 2327: "SDP: Session Description Protocol".

[57] 3GPP TS 26.103: " Speech Codec List for GSM and UMTS".

[58] 3GPP TS 28.062: " Inband Tandem Free Operation (TFO) of speech codecs".

[59] IETF RFC 3556: "Session Description Protocol (SDP) Bandwidth Modifiers for RTP Control Protocol (RTCP) bandwidth".

[61] Draft ETSI TS <3026> Telecommunications and Internet Converged Services and Protocols for Advanced Networking (TISPAN); NGN Signalling Control Protocol;Anonymous Communication Rejection (ACR) PSTN/ISDN simulation services

7.4.6 Call Forwarding Busy (CFB)/ Call Forwarding No Reply (CFNR) / Call Forwarding Unconditional (CFU)

~~The actions of the MGCF at the ISUP/BICC side are described in ITU-T Recommendation Q.732.2-4 [42] under the clause "Interactions with networks not providing any call diversion information".~~

In case of interaction with networks which do not provide any notification of the communication diversion or communication redirection information (e.g. redirection counter) in the signalling system, the communication continues according to the basic call procedures.

7.4.6.1 Interworking at the O-MGCF

7.4.6.1.1 Call forwarding within the SIP Network.

For the mapping of IAM to the INVITE Message no additional procedures beyond the basic call procedures are needed.

With regard to the backward messages the following mapping is valid:

Table 7.4.6.1.1-1 Mapping of SIP messages to ISUP messages

←Message sent to ISUP	**←Message Received from SIP**
ACM indicating call forwarding	181
CPG indicating Cal Diversion	181
ACM indicating ringing	180
CPG indicating Alerting	180
ANM	200 OK
CON	200 OK (Neither a 181 nor a 180 was sent)

Table 7.4.6.1.1-2 Mapping of History-Info Header to ISUP Original Called number

Source SIP header field	**Source**	**Original called number**	**Derived value of parameter field**

and component	**Component value**		
		Numbering Plan Indicator	*"ISDN (Telephony) numbering plan (Recommendation E.164)"*
Hi-target-to-uri of 1st History-Info entry appropriate global number portion of the URI, assumed to be in form "+" CC + NDC + SN	CC	Nature of address indicator	If CC is equal to the country code of the country where I-MGCF is located AND the next ISUP node is located in the same country, then set to "national (significant) number" else set to "international number"
	CC, NDC, SN	Address signals	If NOA is "national (significant) number" then set to NDC + SN. If NOA is "international number" then set to CC + NDC + SN

Table 7.4.6.1.1-3 Mapping of History Header to ISUP Redirection Information

Source SIP header field and component	**Source Component value**	**Redirection Information**	**Derived value of parameter field**
Privacy, priv-value component	*"history"* for the whole History header or for the last two index entries	Redirection indicator	*Call diverted, all redirection info presentation restricted*
	Privacy header field absent or "*none*"		*Call diverted*
		Original redirection reasons	*Unknown*
Reason header in History Index; Protocol=Redirection Cause value	**Protocol Cause and reason-text**	Call diversion information	***Redirecting Reason***
	1; Normal redirection		unconditional
	2 Forward unavailable		unknown
	3 Forward busy		User busy
	4 Forward no reply		no reply
	5 Forward immediate		unconditional
	6 Deflection		Deflection during alerting
	8 Mobile subscriber not reachable		Mobile subscriber not reachable

Table 7.4.6.2.1-4 Mapping of History-Info Header to ISUP Redirection number

Source SIP header field and component	**Source Component value**	**Redirection number**	**Derived value of parameter field**
Hi-target-to-uri appropriate global number portion of the URI, assumed to be in form "+" CC + NDC + SN	CC	Nature of address indicator	If CC is equal to the country code of the country where I-MGCF is located AND the next ISUP node is located in the same country, then set to "national (significant) number" else set to "international number"

	CC, NDC, SN	Address signals	If NOA is "national (significant) number" then set to NDC + SN. If NOA is "international number" then set to CC + NDC + SN

Table 7.4.6.1.1-5 Mapping of History-Info Header to ISUP Redirection number restriction indicator

Source SIP header field and component	**Source Component value**	**Redirection number restriction**	**Derived value of parameter field**
Privacy, priv-value component	history	Redirection number restriction indicator	presentation restricted
	Privacy header field absent or "none"		Presentation allowed or absent

Table 7.4.6.1.1-6 Mapping of History-Index to ISUP Call Diversion Information

Source SIP header field and component	**Source Component value**	**Call Diversion Information**	**Derived value of parameter field**
Privacy, priv-value component	history	Notification subscription options	If the priv-value history is set for the History-Info Header or to the hist-info element entries concerning the redirecting and diverted to uri then *presentation not allowed* shall be set If the priv-value history is set only to the hist-info element concerning the redirecting uri then *presentation allowed without redirection number shal be set.*
	Privacy header field absent or "none"		*presentation allowed with redirection number*
		Original redirection reasons	*Unknown*
Hi-index		Redirection Counter	Index entries which are caused by communication diversion shall be counted
Reason header in History Index; Protocol=Redirection Cause value	**Protocol Cause and reason-text**	Call diversion information	***Redirecting Reason***
	1; Normal redirection		unconditional
	2 Forward unavailable		unknown
	3 Forward busy		User busy
	4 Forward no reply		no reply
	5 Forward immediate		unconditional
	6 Deflection		Deflection during alerting
	8 Mobile subscriber not reachable		bile subscriber not reachable

Table 7.4.6.1.1-7 Mapping of History Index to ISUP Event Information

Source SIP header field and component	**Source Component value**	**Event Information**	**Derived value of parameter field**
		Event indication	Shall be set to ALERTING if mapped from a 180
			Shall be set to PROGRESS if mapped from a 181
Reason header in History Index; Protocol=Redirection **Protocol Cause and reason-text**	3 Forward busy		call forwarded on busy (national use)
	4 Forward no reply		call forwarded on no reply (national use)
	5 Forward immediate		call forwarded unconditional (national use)

Table 4.7.2.2.1 – 8 Mapping of 181 Forwarding → ACM

Source SIP header field and component	**Source Component value**	**ISUP Parameter or IE**	**Derived value of parameter field**
181		ACM	
		Optional backward call indicators	Bit B call diversion may occur
		Generic notification indicators	Call is diverting
History Header	see Table 7.4.6.1.1-4	Redirection number	see Table 7.4.6.1.1-4
Priv-value	See Table 7.4.6.1.1-5	Redirection number restriction indicator	See Table 7.4.6.1.1-5
Priv-value	See Table 7.4.6.1.1-6	Call diversion information *Notification subscription options*	See Table 7.4.6.1.1-6
History Index	**Reason Header: Reason = [4]** See Table 7.4.6.1.1-6	Call diversion information	***Redirecting Reason*** See Table 7.4.6.1.1-6

180 Ringing → ACM or CPG

Table 4.7.2.2.1 – 9 : Mapping of 181 Forwarding → CPG if ACM was already sent

Source SIP header field and component	**Source Component value**	**ISUP Parameter or IE**	**Derived value of parameter field**
181		CPG	
		Optional backward call indicators	call diversion may occur
		Generic notification indicators	Call is diverting
		Event indicator	PROGRESS)
Reason header in History Index; Protocol=Redirection **Protocol Cause and reason-text**	3 Forward busy		CFB(national use)
	4 Forward no reply		CFNR(national use)
	5 Forward immediate		CFU(national use
History Header	see Table 7.4.6.1.1-4	Redirection number	see Table 7.4.6.1.1-4
Priv-value	See Table 7.4.6.1.1-5	Redirection number restriction indicator	See Table 7.4.6.1.1-5
Priv-value	See Table 7.4.6.1.1-6	Call diversion information *Notification subscription options*	See Table 7.4.6.1.1-6
Reason Header Reason = "Q.732" Or History Index	See Table 7.4.6.1.1-6	Call diversion information *Redirecting Reason*	See Table 7.4.6.1.1-6

Table 4.7.2.2.1 – 10: Mapping of 180 → ACM if no 181 was received before

Source SIP header field and component	**Source Component value**	**ISUP Parameter or IE**	**Derived value of parameter field**
180		ACM	
History Header	If Index indicate that there is a call forwarding.	Optional backward call indicators	Bit B no indication call diversion may occur
History Header	If Index indicate that there is a call forwarding. More than two index	Generic notification indicators	Call is diverting

	entries with two digits Index X.X		
History Header	See Table 7.4.6.1.1-4	Redirection number	See Table 7.4.6.1.1-4
Priv-value	See Table 7.4.6.1.1-5	Redirection number restriction indicator	See Table 7.4.6.1.1-5
Priv-value	See Table 7.4.6.1.1-6	Call diversion information *Notification subscription options*	See Table 7.4.6.1.1-6
Reason Header in History Index Reason = "Redirection"	See Table 7.4.6.1.1-6	Call diversion information *Redirecting Reason*	See Table 7.4.6.1.1-6

The IWU can indicate the call diversion in the mapping of 180 to CPG in fact if the response before was a 181.

Table 4.7.2.2.1 – 11: (CFB) Mapping of 180 → CPG if a 181 was received before

Source SIP header field and component	**Source Component value**	**ISUP Parameter or IE**	**Derived value of parameter field**
180		CPG	
		Optional backward call indicators	call diversion may occur
		Generic notification indicators	Call is diverting
History-header		Event indicator	ALERTING
History Header	See Table 7.4.6.1.1-4	Redirection number	See Table 7.4.6.1.1-4
Priv-value	See Table 7.4.6.1.1-5	Redirection number restriction indicator	See Table 7.4.6.1.1-5
Priv-value	See Table 7.4.6.1.1-6	Call diversion information *Notification subscription options*	See Table 7.4.6.1.1-6
Reason Header in History Index Reason = "Q.732"	See Table 7.4.6.1.1-6	Call diversion information *Redirecting Reason*	See Table 7.4.6.1.1-6

Table 4.7.2.2.1 – 12:: (CFB) Mapping of 200OK

Source SIP header field and component	**Source Component value**	**ISUP Parameter or IE**	**Derived value of parameter field**
200 OK		ANM	
History Header	See Table 7.4.6.1.1-4	Redirection number	See Table 7.4.6.1.1-4
Priv-value	See Table 7.4.6.2.1-5	Redirection number restriction indicator	See Table 7.4.6.2.1-5

7.4.6.1.2 Call forwarding within the ISUP Network.

The following Scenario shows if a Call Forwarding appears in the ISUP/PSTN Network and the redirected Number is within the SIP Network. The following Figure should be seen as example.

For the mapping of 180 and 200 OK to the regarding ISUP messages and parameters no additional procedures beyond the basic call procedures are needed.

Table 4.7.2.1.2-1 mapping of IMS with SIP INVITE

ISUP Parameter or IE	**Derived value of parameter field**	**SIP component**	**Value**
IAM		INVITE	
Redirecting number		History Header	hi-targeted-to-uri
Nature of address indicator:	*"national (significant) number"*	hi-targeted-to-uri	Add CC (of the country where the IWU is located) to Generic Number Address Signals then map to user portion of URI scheme used. **Addr-spec** "+" CC NDC SN mapped to user portion of URI scheme used
	"international number"		Map complete Redirection number Address Signals to user portion of URI scheme used.

Address Signals	If NOA is "*national (significant) number*" then the format of the Address Signals is: NDC + SN If NOA is "*international number*" then the format of the Address Signals is: CC + NDC + SN	hi-targeted-to-uri	"+" CC NDC SN mapped to userinfo portion of URI scheme used
Redirecting number	APRI	Privacy Header	Priv-value
	"presentation restricted"		"*History*"-Index"
	"presentation allowed"		Privacy header field absent or "none"
Redirecting Information	Redirection indicator	Privacy Header	Priv-value
	Call diverted		"none"
	Call diverted, all redirection info presentation restricted		"*History*"-Index"
Redirecting Information	Redirection counter 1 …5	History Index	Index number for Redirecting number = value of redirecting counter Rapporteur's comment: Here a consideration is needed if only one Index is needed or the IWU has to add the amount of inexes regarding the counter
Redirecting Information	***Redirecting Reason***	Reason header in History Index; Protocol=Redirection Cause value	**Protocol Cause and reason-text**
	Unknown/not available		1; Normal redirection
	User busy		3 Forward busy
	No reply		4 Forward no reply
	Unconditional		5 Forward immediate
	Deflection during alerting		6 Deflection
	Deflection immediate response		6 Deflection
	Mobile subscriber not reachable		8 Mobile subscriber not reachable
Original Called Party Number	*See* Redirecting number	first Index entry of History Header	Redirecting number <sip:oCdPN@UA2?Reason:732; cause=XXX ;text="XXX"> ;index=1,

Original Called Party Number	APRI	Privacy Header	Priv-value
	“presentation restricted”		*“history”*
	“presentation allowed”		*“none”*

7.4.6.2 Interworking at the I-MGCF

7.4.6.2.1 Call forwarding within the ISUP Network.

Table 7.4.6.2.1 - 1 Mapping of ISUP to SIP Massages

←Message sent to SIP	←Message Received from BICC/ISUP
INVITE	IAM
181	ACM indicating call forwarding
180	ACM indicating ringing
181	CPG indicating call forwarding
180	CPG indicating ringing
200 OK	ANM
200 OK	CON

In case of a call forwarding a ACM with a Optional backward call indicator indicating that call diversion may occur and a Generic notification Call is diverting a 181 shall be send in backward direction.

Table 7.4.6.2.1 - 2 ACM → 181 Forwarding

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
ACM		181	
Optional backward call indicators	Bit B call diversion may occur		
Generic notification indicators	Call is diverting		Based on the ISUP Parameter the 181 message is created
Redirection number		History Header	

Nature of address indicator:	*“national (significant) number”* else set to	hi-targeted-to-uri	Add CC (of the country where the IWU is located) to Redirection Number Address Signals then map to user portion of URI scheme used.

	"international number"		Map complete Redirection number Address Signals to user portion of URI scheme used.
Address Signals	if NOA is *"national (significant) number"* then the format of the address signals is: NDC + SN If NOA is *"international number"* then the format of the address signals is: CC + NDC + SN	hi-targeted-to-uri	Addr-spec "+" CC NDC SN mapped to user portion of URI scheme used
Redirection number restriction indicator	entation restricted		"History"
	Presentation allowed or absent		Privacy header field absent or "none"
Call diversion information	*Notification subscription options* presentation not allowed	Priv-value	"History" for whole History-Index header
	presentation allowed without redirection number		"History" for History-Index entry of thr redirecting uri
	presentation allowed with redirection number		Privacy header field absent or "none"
Call diversion information	***Redirecting Reason***	Reason header in History Index; Protocol=Redirection Cause value	**Protocol Cause and reason-text**
	Unknown/not available		1; Normal redirection
	User busy		3 Forward busy
	No reply		4 Forward no reply
	Unconditional		5 Forward immediate
	Deflection during alerting		6 Deflection
	Deflection immediate response		6 Deflection
	Mobile subscriber not reachable		8 Mobile subscriber not reachable

A received CPG shall be mapped t a 180 if the CPC indicates a Alerting is due to the mapping ruled defined within the basic call.

Table 7.4.6.2.1 - 3: Mapping of CPG → 181

CPG		181	
Optional backward call indicators	call diversion may occur		
Event Indicator	PROGRESS CFB(national use) CFNR(national use) CFU(national use)		
Generic notification Indicator	Call is a diverting Call		Based on this Information the CPG shall be mapped to a 181
Redirection number	See Table 7.4.6.2.1-2	History Header	See Table 7.4.6.2.1-2
Call Diversion Information	See Table 7.4.6.2.1-2	History Header	See Table 7.4.6.2.1-2
Redirection number restriction indicator	See Table 7.4.6.2.1-2	Priv-value	See Table 7.4.6.2.1-2

Table 7.4.6.2.1 - 4: (CFB) Mapping of ANM/CON → 200OK

ANM		200	
Redirection number	*See Table* 7.4.6.2.1-2	History Header	*See Table* 7.4.6.2.1-2
Redirection number restriction indicator	*See Table* 7.4.6.2.1-2	Priv-value	*See Table* 7.4.6.2.1-2

7.4.6.2.2 Call forwarding within the SIP Network.

For the mapping of 180 and 200 OK to the regarding ISUP messages and parameters no additional procedures beyond the basic call procedures are needed.

Table 7.4.6.2.2- 1 Mapping of INVITE to IAM

INVITE		IAM	
History Header	<sip:e164-User2@UA2?Reason:SIP; cause=486 ;text="Busy >;index=2.1, appropriate global number portion of the URI, assumed to be in form "+" CC+NDC+SN	Redirecting number	*Nature of address indicator*: If CC is equal to the country code of the country where I-MGCF is located AND the next BICC/ISUP node is located in the same country then set to *"national (significant) number"* else set to *"international*

			number" **Address Signals** If NOA is "*national (significant) number*" then set to NDC + SN. If NOA is "*international number*" then set to CC+NDC+SN
Privacy Header	Priv-value	Redirecting number	APRI
	proposed Values for History "History-Index" or "History-header"		"presentation restricted"
	Privacy header field absent or "none"		"presentation allowed"
Privacy Header & Index History > 1 or 1.1	Priv-value	Redirecting Information	Redirection indicator
	"none"		Call diverted (based on Invormation given in the history-Index header this parameter is set)
	proposed Values for History "History-header" or "header" (if P-Asserted-ID is "id")		Call diverted, all redirection info presentation restricted
History Index	Index number for Redirecting number	Redirecting Information	Redirection counter = Value Index number for Redirecting number If Value >5 then release Call
Reason header in History Index; Protocol=Redirection Cause value	**Protocol Cause and reason-text**	Redirecting Information	***Redirecting Reason***
	1; Normal redirection		Unknown/not available
	3 Forward busy		User busy
	4 Forward no reply		No reply
	5 Forward immediate		Unconditional
	6 Deflection		Deflection during alerting
	6 Deflection		Deflection immediate response
	Mobile subscriber not reachable		Mobile subscriber not reachable

To header and first Index entry of History Header	Redirecting number <sip:oCdPN@UA2?Reason:732; cause=XXX ;text="XXX"> ;index=1,	Original Called Party Number	*See* Redirecting number
Privacy Header	Priv-value	Original Called Party Number	APRI
	proposed Values for History "History-Index" or "History-header" Note: depended on the privacy statement for the redirecting number		"presentation restricted"
	Privacy header field absent or "none"		"presentation allowed"

7.4.7 Call Deflection (CD)

~~The actions of the MGCF at the ISUP/BICC side are described in ITU-T Recommendation Q.732.5 [42] under the clause "Interactions with other networks".~~See 7.4.6

CR-Form-v7.1

CHANGE REQUEST

⌘	**TS29.163**	**CR 061**	⌘ **rev**	**1**	⌘ Current version:	**6.6.0** ⌘

*For **HELP** on using this form, see bottom of this page or look at the pop-up text over the ⌘ symbols.*

Proposed change affects:	UICC apps⌘		ME	X	Radio Access Network		Core Network X

Title: ⌘	Addition of Interworking of TISPAN ACR simulation service		
Source: ⌘	T-Mobile		
Work item code: ⌘	FBI	***Date:*** ⌘	12.04.2005
Category: ⌘	**B**	***Release:*** ⌘	Rel-7
	Use one of the following categories: ***F*** *(correction)* ***A*** *(corresponds to a correction in an earlier release)* ***B*** *(addition of feature),* ***C*** *(functional modification of feature)* ***D*** *(editorial modification)* Detailed explanations of the above categories can be found in 3GPP TR 21.900.		*Use one of the following releases:* *Ph2 (GSM Phase 2)* *R96 (Release 1996)* *R97 (Release 1997)* *R98 (Release 1998)* *R99 (Release 1999)* *Rel-4 (Release 4)* *Rel-5 (Release 5)* *Rel-6 (Release 6)* *Rel-7 (Release 7)*

Reason for change: ⌘	For supporting the TISPAN NGN simulation service “Anonymus communication rejection” in the TISPAN IMS the interworking between IMS and PSTN/ISDN is needed Within the TISPAN the yellow marked words are a Rapporteurs comments an dneeds further consideration.
Summary of change: ⌘	Subsection 7.4.23 added
Consequences if not approved: ⌘	The interworking of ACR can not be supported

Clauses affected: ⌘	7.3.23 (new clause)			
	Y	**N**		
Other specs affected: ⌘		X	Other core specifications	⌘
		X	Test specifications	
		X	O&M Specifications	
Other comments: ⌘				

How to create CRs using this form:

Comprehensive information and tips about how to create CRs can be found at http://www.3gpp.org/specs/CR.htm. Below is a brief summary:

1) Fill out the above form. The symbols above marked ⌘ contain pop-up help information about the field that they are closest to.

2) Obtain the latest version for the release of the specification to which the change is proposed. Use the MS Word "revision marks" feature (also known as "track changes") when making the changes. All 3GPP specifications can be downloaded from the 3GPP server under ftp://ftp.3gpp.org/specs/ For the latest version, look for the directory name with the latest date e.g. 2001-03 contains the specifications resulting from the March 2001 TSG meetings.

3) With "track changes" disabled, paste the entire CR form (use CTRL-A to select it) into the specification just in front of the clause containing the first piece of changed text. Delete those parts of the specification which are not relevant to the change request.

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*.

[1] ITU-T Recommendation G.711: "Pulse Code Modulation (PCM) of voice frequencies".

[2] ITU-T Recommendation H.248.1 (2002): "Gateway control protocol: Version 2".

[3] ITU-T Recommendation Q.701 to Q.709: "Functional description of the message transfer part (MTP) of Signalling System No. 7".

[4] ITU-T Recommendations Q.761to Q.764 (2000): "Specifications of Signalling System No.7 ISDN User Part (ISUP)".

[5] Void.

[6] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".

[7] Void.

[8] 3GPP TS 24.228: "Signalling flows for the IP multimedia call control based on SIP and SDP".

[9] 3GPP TS 24.229: "IP Multimedia Call Control Protocol based on SIP and SDP".

[10] 3GPP TS 23.002: "Network Architecture".

[11] 3GPP TS 22.228: "Service requirements for the IP Multimedia Core Network Subsystem".

[12] 3GPP TS 23.228: "IP Multimedia subsystem (IMS)".

[13] Void.

[14] 3GPP TS 29.205: "Application of Q.1900 series to Bearer Independent CS Network architecture; Stage 3".

[15] 3GPP TS 29.332: "Media Gateway Control Function (MGCF) – IM-Media Gateway (IM-MGW) interface, Stage 3".

[16] IETF RFC 791: "Internet Protocol".

[17] IETF RFC 768: "User Datagram Protocol".

[18] IETF RFC 2960: "Stream Control Transmission Protocol".

[19] IETF RFC 3261: "SIP: Session Initiation Protocol".

[20] 3GPP TS 29.202: "Signalling System No. 7 (SS7) signalling transport in core network; Stage 3".

[21] IETF RFC 2474: "Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers".

[22] IETF RFC 2475: "An Architecture for Differentiated Services".

[23] IETF RFC 3267: "Real-Time Transport Protocol (RTP) payload format and file storage format for the Adaptive Multi-Rate (AMR) Adaptive Multi-Rate Wideband (AMR-WB) audio codecs".

[24] IETF RFC 793: "Transmission Control Protocol".

[25] 3GPP TS 29.414: "Core network Nb data transport and transport signalling".

[26] 3GPP TS 29.415: "Core network Nb interface user plane protocols".

[27] 3GPP TS 23.205: "Bearer-independent circuit-switched core network; Stage 2".

[28] Void.

[29] ITU-T Recommendation Q.2150.1: "Signalling transport converter on MTP3 and MTP3b".

[30] ITU-T Recommendations Q.1902.1 to Q.1902.6 (07/2001): "Bearer Independent Call Control".

[31] ITU-T Recommendation Q.1950 (2002): "Bearer independent call bearer control protocol".

[32] 3GPP TS 26.236: "Packet switched conversational multimedia applications; Transport protocols".

[33] 3GPP TS 29.232: "Media Gateway Controller (MGC) – Media Gateway (MGW) interface; Stage 3".

[34] IETF RFC 2833: "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals".

[35] ITU-T Recommendation Q.765.5: "Signalling system No. 7 – Application transport mechanism: Bearer Independent Call Control (BICC)".

[36] IETF RFC 3264: "An Offer/Answer Model with the Session Description Protocol (SDP)".

[37] IETF RFC 3312: "Integration of Resource Management and Session Initiation Protocol (SIP)".

[38] ITU-T Recommendation Q.850 (1998): "Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part".

[39] IETF RFC 2460: "Internet Protocol, Version 6 (IPv6) Specification"

[40] IETF RFC 3323: "A Privacy Mechanism for the Session Initiation Protocol (SIP)".

[41] IETF RFC 3325: "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks".

[42] ITU-T Recommendation Q.730 to Q.737 (12/1999): "ISDN user part supplementary services".

[43] ITU-T Recommendation I.363.5 (1996): "B-ISDN ATM Adaptation Layer specification: Type 5 AAL".

[44] ITU-T Recommendation Q.2110 (1994): "B-ISDN ATM adaptation layer - Service Specific Connection Oriented Protocol (SSCOP)".

[45] ITU-T Recommendation Q.2140 (1995): "B-ISDN ATM adaptation layer - Service specific coordination function for signalling at the network node interface (SSCF AT NNI)".

[46] ITU-T Recommendation Q.2210 (1996): "Message transfer part level 3 functions and messages using the services of ITU-T Recommendation Q.2140".

[47] 3GPP TS 23.221: "Architectural requirements".

[48] ITU-T Recommendation E.164 (05/1997): "The international public telecommunication numbering plan".

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

[50] 3GPP TS 26.102: "Adaptive Multi-Rate (AMR) speech codec; Interface to Iu and Uu".

[51] IETF RFC 3550: "RTP: A Transport Protocol for Real-Time Applications".

[52] IETF RFC 3551: "RTP Profile for Audio and Video Conferences with Minimal Control".

[53] IETF RFC 3555: "MIME Type Registration of RTP Payload Formats".

[54] IETF RFC 3262: "Reliability of provisional responses".

[55] IETF RFC 3311: "SIP UPDATE method".

[56] IETF RFC 2327: "SDP: Session Description Protocol".

[57] 3GPP TS 26.103: " Speech Codec List for GSM and UMTS".

[58] 3GPP TS 28.062: " Inband Tandem Free Operation (TFO) of speech codecs".

[59] IETF RFC 3556: "Session Description Protocol (SDP) Bandwidth Modifiers for RTP Control Protocol (RTCP) bandwidth".

7.4.23 Anonymus Call Rejection (ACR) with Anonymus Communication Rejection (ACR)

This section deals with the interworking of:

1. The interworking of the ACR Supplementary Service executed within the PSTN/ISDN with the TISPAN NGN and;

2. The interworking of the ACR Simulation Service executed within the TISPAN NGN with the PSTN/ISDN.

[60] Draft ETSI TS <3029> Telecommunications and Internet Converged Services and Protocols for Advanced Networking (TISPAN); NGN Signalling Control Protocol;Anonymous Communication Rejection (ACR) PSTN/ISDN simulation services

Editor's note: The above document cannot be formally referenced until it is published as an TISPAN TS

[61] ETSI EN 300 485: "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 (SS7) ISDN User Part (ISUP) [ITU-T Recommendation Q.850 (1998) with addendum modified]".

The Reason Header shall be mapped to the Cause Value. The following procedures shall apply

NOTE: When interworking with existing implementations, the cause value 24 *"call rejected due to ACR supplementary service"* indicating that the call was rejected due to the ACR service, may be lost.

7.4.23.1 SIP-ISUP protocol interworking at the I-MGCF

7.4.23.1.1 Coding of 183

If the service is processed within the PSTN/ISDN the following procedures shall apply.

Rapporteurs comment: The following text is taken from the ETSI ACR specification:

When a call is rejected due to the ACR supplementary service, the calling user shall receive an appropriate indication by either;

- sending a Release message as specified in the "Unsuccessful call set-up" clause of [61] with cause value 24 "call rejected due to ACR supplementary service" as specified in EN 300 485 [61**Error! Reference source not found.**]; or

- by sending an Address Complete or Call Progress message, that may include cause value 24 and applying an in-band announcement as specified in the "Tones and announcements" clause of [61]; or

- the call may be forwarded to a voice message service to provide an announcement.

→ This shows that sending a ACM is appropriate.

Rapporteur's comment: How do we know that a Announcement is used. Information Flow is needed. IETF early media draft could be used.

If the I-MGCF is able to detect the announcement then:

A ACM shall be mapped to a 183.

- The Cause Value 24 *"call rejected due to ACR simulation service"shall* be mapped to the Reason header field of the 183 Session Progress response.

- The Announcement shall be forwarded to the UE:A

7.4.23.1.2 Coding of the REL to 603

If ISUP Cause Value field in the ISUP REL includes Cause Value 24 *"call rejected due to ACR supplementary service"* the I-MGCF maps this to 603 including a Reason header with a Cause Value 24 "call rejected due to ACR supplementary service" and a warning header with the "warn-code" 399 (Miscellaneous warning) and "warn text" (that is due to the requirements of the operator e.G. "Your Call has been rejected because your ~~number~~ Identity was not present")

7.4.23.2 SIP-ISUP protocol interworking at the O-MGCF

7.4.23.2.1 Coding of ACM

Rapporteurs Comment: How do we know that a Announcement is used. Information Flow is needed. IETF early media draft could be used.

If the O-MGCF is able to detect the announcement then:

If a announcement is served by TISPAN NGN a 183 must be mapped to the ACM including:

- A Backward call indicators including the Called Party's Status Indicator = "*no indication*"

- A Optional Backward indicator where the In-band information indicator = *"in-band information or an appropriate pattern is now available"*

- The Cause Value shall be taken from the Reason header field.

Rapporteur's Comment: more consideration Is needed

7.4.23.2.2 Receipt of 603

If the Reason header field includes Cause Value 24 *"call rejected due to ACR supplementary service"* in a 603, then the Cause Value shall be mapped to the ISUP Cause Value field in the ISUP REL.

3GPP TSG-CT WG3 Meeting #36 **Tdoc C3-050421**
Cancun, Mexico. 25th - 29th April 2005.

CR-Form-v7.1

CHANGE REQUEST

⌘	**TR 24.819**	**CR**		⌘ **rev**	**-**	⌘	Current version:	**0.1.0**	⌘

*For **HELP** on using this form, see bottom of this page or look at the pop-up text over the ⌘ symbols.*

Proposed change affects:	UICC apps⌘		ME	X	Radio Access Network		Core Network	X

Title:	⌘	Addition of Interworking of TISPAN Reason header		
Source:	⌘	T-Mobile		
Work item code:	⌘	FBI	***Date:*** ⌘	12.04.2005
Category:	⌘	**B**	***Release:*** ⌘	Rel-7
		Use one of the following categories: ***F*** *(correction)* ***A*** *(corresponds to a correction in an earlier release)* ***B*** *(addition of feature),* ***C*** *(functional modification of feature)* ***D*** *(editorial modification)* Detailed explanations of the above categories can be found in 3GPP TR 21.900.		*Use one of the following releases:* *Ph2 (GSM Phase 2)* *R96 (Release 1996)* *R97 (Release 1997)* *R98 (Release 1998)* *R99 (Release 1999)* *Rel-4 (Release 4)* *Rel-5 (Release 5)* *Rel-6 (Release 6)* *Rel-7 (Release 7)*

Reason for change:	⌘	For supporting the TISPAN NGN simulation service the mapping of the Reason Header is needed. This CR proposes to map the reason header to the ISUP Cause Value and vice versa.
Summary of change:	⌘	Modification of sections 7.2.3.1.7-7.2.3.1.10 and 7.2.3.2.12-7.2.3.2.18 in TS 29.163 V6.6.0
Consequences if not approved:	⌘	The interworking of ACR can not be supported

Clauses affected:	⌘	8 (new clause 7.3.23 in 29.163 V6.6.0)				
		Y	**N**			
Other specs affected:	⌘		X	Other core specifications	⌘	
			X	Test specifications		
			X	O&M Specifications		
Other comments:	⌘					

How to create CRs using this form:

Comprehensive information and tips about how to create CRs can be found at http://www.3gpp.org/specs/CR.htm. Below is a brief summary:

1) Fill out the above form. The symbols above marked ⌘ contain pop-up help information about the field that they are closest to.

2) Obtain the latest version for the release of the specification to which the change is proposed. Use the MS Word "revision marks" feature (also known as "track changes") when making the changes. All 3GPP specifications can be

downloaded from the 3GPP server under ftp://ftp.3gpp.org/specs/ For the latest version, look for the directory name with the latest date e.g. 2001-03 contains the specifications resulting from the March 2001 TSG meetings.

3) With "track changes" disabled, paste the entire CR form (use CTRL-A to select it) into the specification just in front of the clause containing the first piece of changed text. Delete those parts of the specification which are not relevant to the change request.

8 Interworking towards CS networks

Editor's note: This clause is under the responsibility of CN3. Material for this clause must be submitted to CN3. CN1 will not review the content of this clause.

Editor's note: This clause will cover requirements from providing IMS via fixed broadband with regards to interworking of SIP with ISUP/BICC networks. It is intended that material from this clause will be added to TS 29.163.

Editor's note: The mapping of reason headers towards the ISDN may be misused due to possible user creation of the reason header since there is no screening in IMS.

8.1 Coding of the REL

Editor's note: Subclause 7.2.3.1.7 of TS 29.163 will be modified as follows:

If the Reason-header field with Q.850 Cause Value is included in the BYE or CANCEL, 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*".~~The SIP BYE and CANCEL requests are mapped into a REL message with cause value #16 and #31 respectively as indicated in table 8.~~

Table 8: Coding of REL

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

~~NOTE: If an optional Reason header field is included in the BYE or CANCEL, then the Cause Value can be mapped to the ISUP Cause Value field in the ISUP REL. The mapping between the Cause Indicators parameter and the Reason header is out of the scope of the present specification.~~

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.			

8.2 Receipt of the Release Message

Editor's note: Subclause 7.2.3.1.8 of TS 29.163 will be modified as follows:

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 has been received) on the incoming side of the I-MGCF then the I- MGCF does not send a 487 Request terminated and instead waits until the ACK is received before sending a BYE message.

A Reason header field containing the received (Q.850) Cause Value of the REL shall be added to the SIP final response or BYE 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	Should be filled with the definition text as stated in ITU-T Rec. Q.850 (Note 2)
NOTE 1 – "XX" is the Cause Value as defined in ITU-T Rec. Q.850. NOTE 2 – Due to the fact that the Cause Indicators parameter does not include the definition text as defined in Table 1/Q.850, this is based on provisioning in the I-MGCF.			

Editor note: Is it really required to fill the component value with text ?

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 Status-Code 4xx (Client Error) or 5xx (Server Error). The Status code to be sent is determined by examining the Cause code value received in the REL message. Table 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)
404 Not Found	Cause value No. 5 (Misdialled trunk prefix)
486 Busy Here	Cause value No. 17 (user busy)
480 Temporarily unavailable	Cause value No 18 (no user responding)

←SIP Message	← REL
Status code	**Cause parameter**
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)
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 1)
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)
Note 1: Class 1 and class 2 have the same default value.	

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

Editor's note: Subclause 7.2.3.1.9 of TS 29.163 will be modified as follows:

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

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

2) If the final response (i.e. 200 OK (INVITE)) has not already been sent, the I-MGCF shall send a SIP response with Status-Code 480 Temporarily Unavailable.

8.4 Autonomous Release at I-MGCF

Editor's note: Subclause 7.2.3.1.10 of TS 29.163 will be modified as follows:

Table 10 shows the trigger events at the MGCF and the release initiated by the MGCF when the call is traversing from SIP to ISUP/BICC.

A Reason header field containing the (Q.850) Cause Value of the REL message sent by the I-IWU shall be added to the SIP Message (BYE or final response) sent by the SIP side of the I-MGCF.

Editor's note: Is it meaningsful to indicate cause value for internal error in the network to the users.

Table 10: Autonomous Release at I-MGCF

← SIP Response	Trigger event	REL → cause parameter
484 Address Incomplete	Determination that insufficient digits received.	Not sent.
480 Temporarily Unavailable	Congestion at the MGCF/Call is not routable.	Not sent.
BYE	ISUP/BICC procedures result in release after answer	According to ISUP/BICC procedures.
BYE	SIP procedures result in release after answer.	127 (Interworking unspecified)
500 Server Internal error	Call release due to the ISUP/BICC compatibility procedure (note)	According to ISUP/BICC procedures.
484 Address Incomplete	Call release due to expiry of T7 within the ISUP/BICC procedures	According to ISUP/BICC procedures.
480 Temporarily Unavailable	Call release due to expiry of T9 within the BICC/ISUP procedures	According to BICC/ISUP procedures.
480 Temporarily Unavailable.	Other BICC/ISUP procedures result in release before answer.	According to BICC/ISUP procedures.
Note: MGCF receives unrecognized ISUP or BICC signalling information and determines that the call needs to be released based on the coding of the compatibility indicators, refer to ITU-T Recommendation Q.764 [4] and ITU-T Q.1902.4 [30].		

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

Editor's note: Subclause 7.2.3.1.12 of TS 29.163 will be modified as follows:

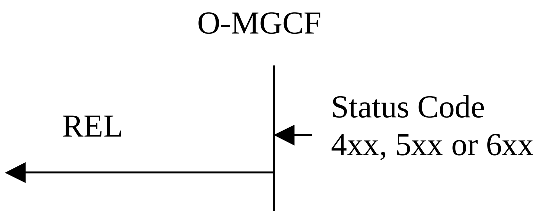

Figure 21: Receipt of Status codes 4xx, 5xx or 6xx

If a Reason header is included in a 4XX, 5XX, 6XX, 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

~~When receiving SIP response with status codes 4xx, 5xx or 6xx, the O-MGCF shall send a REL message. The~~ 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

←REL (cause code)	←4xx/5xx/6xx SIP Message
127 (interworking unspecified)	420 Bad Extension
127 (interworking unspecified)	421 Extension required
127 (interworking unspecified)	423 Interval Too Brief
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 1)	487 Request terminated
127 (interworking unspecified)	488 Not acceptable here
127 (interworking unspecified)	493 Undecipherable
127 (interworking unspecified)	500 Server Internal error
127 (interworking unspecified)	501 Not implemented
127 (interworking unspecified)	502 Bad Gateway
127 (interworking unspecified)	503 Service Unavailable
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 – No interworking if the O-MGCF previously issued a CANCEL request for the INVITE. Note 2 – The 4xx/5xx/6xx SIP Messages that are not covered in this table are not interworked.	

8.6 Receipt of a BYE

Editor's note: Subclause 7.2.3.1.13 of TS 29.163 will be modified as follows:

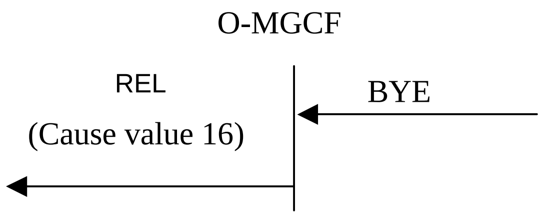

Figure 22: Receipt of BYE method

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

If a Reason header field with Q.850 Cause Value is included in the BYE, 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 method, the O-MGCF sends a REL message with Cause Code value 16 (Normal Call Clearing).

8.7 Receipt of the Release Message

Editor's note: Subclause 7.2.3.1.14 of TS 29.163 will be modified as follows:

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 method. 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).

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

Editor's note: Subclause 7.2.3.1.15 of TS 29.163 will be modified as follows:

Editor's note: Is it meaningsful to indicate cause value for internal error in the network to the users.

If a RSC, GRS or CGB (H/W oriented) message is received and a final response (i.e. 200 OK (INVITE) has already been received the O-MGCF shall send a BYE method. If a 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 (Q.850) Cause Value of the REL message sent by the O-MGCF shall be added to the SIP message (BYE or CANCEL) to be sent by the SIP side of the O-IWU.

8.9 Autonomous Release at O-MGCF

Editor's note: Subclause 7.2.3.1.16 of TS 29.163 will be modified as follows:

Editor's note: Is it meaningsful to indicate cause value for internal error in the network to the users.

If the O-MGCF determines due to internal procedures that the call shall be released then the MGCF shall send

- A BYE method if the ACK has been sent.
- A CANCEL method before 200 OK (INVITE) has been received.

NOTE: The MGCF shall send the ACK method before it sends the BYE, if 200 OK (INVITE) is received.

A Reason header field containing the (Q.850) Cause Value of the REL message sent by the O-IWU shall be added to the SIP Message (BYE or CANCEL) to be sent by the SIP side of the O-IWU.

Table 18a: Autonomous Release at O-MGCF

REL ← Cause parameter	Trigger event	→ SIP
As determined by BICC/ISUP procedure.	COT received with the Continuity Indicators parameter set to "*continuity check failed*" (ISUP only) or the BICC/ISUP timer T8 expires.	CANCEL or BYE according to the rules described in this subclause.
REL with cause value 47 (resource unavailable, unspecified).	Internal resource reservation unsuccessful	As determined by SIP procedure
As determined by BICC/ISUP procedure.	BICC/ISUP procedures result in generation of autonomous REL on BICC/ISUP side.	CANCEL or BYE according to the rules described in this subclause.
Depending on the SIP release reason.	SIP procedures result in a decision to release the call.	As determined by SIP procedure.

an3GPP TSG-CT WG3 Meeting #36 C3-050436
Cancun, Mexico. 25th - 29th April 2005.

Title:	**Reply LS on tracing information for MBMS services**
Response to:	**LS (C3-050264/ S5-058269) on Reply on tracing information for MBMS services**
Release:	**Rel-6**
Work Item:	**OAM-Trace**
Source:	**CT3**
To:	**SA5**
Cc:	**CT4**

Contact Person:
Name: Nico Gabriele
Tel. Number: +44 1635 673 956
E-mail Address: Nico.Gabriele@vodafone.com

---

1. Overall Description:

CT3 thanks SA5 for their reply LS on tracing information for MBMS services.

CT3 has updated the CR to TS 29.061 (N3-050077) submitted at CN3 #35 in Sydney. The CR discussed and approved at CT3 #36 in Cancun (C3-050411) and that modifies TS 29.061 to introduce the tracing capability in the BM-SC, is now fully aligned with the Trace concepts developed by SA5 and introduced in Rel-6.

CT3 would like to provide SA5 the answers to their questions.

Question 1: Are IMSI and IMEI(SV) available in the BM-SC?

Answer: Yes. The IMSI is already available according to the latest version of TS 29.061 and the IMEI(SV) availability to the BM-SC has been introduced at last meeting in Cancun (C3-050305).

Question 2: Which interfaces and what information should be traced in BM-SC?

Answer: The only interface to be traced is the Gmb. The information to be traced does not fall into CT3 competencies and it is part of the modifications that will be proposed in SA5 #42 in Montreal.

Question 3: What are BM-SC triggering events to start and to stop a Trace Recording Session?

Answer: Again, this does not fall into CT3 competencies and it will be part of the modifications proposed in SA5 #42 in Montreal.

Question 4: Does tracing on individual IMSI and IMEI(SV) cover the need from CT3? If not, is a trace on a service intended, and in that case what relation would it have to the Service Level Trace function in OMA?

Answer: Yes. Tracing for MBMS falls into the usual “Subscriber and equipment tracing” topic. Nothing new is introduced.

CT3 has also co-ordinated the work with CT4 and the necessary changes for the introduction of the tracing capability for the BM-SC have been approved at last CT4 #27 in Cancun (C4-050889 and C4-050737).

2. Actions:

To TSG SA5 group.

ACTION: No action is required.

3. Date of Next CT3 Meetings:

CT3_37	29th August – 2nd September 2005	London, UK
CT3_38	31st October - 4th November 2005	Berlin, DE

Title: **Charging Implications of SCUDIF**
Release: **REL-5**

Source: **CT3**
To: **SA1**
Cc: **SA5 SWGB**

Contact Person:
Name: **Thomas Belling**
Tel. Number: +49 89 636 75207
E-mail Address: Thomas.Belling AT siemens.com

---

1. Overall Description:

CT3 has become aware of LS S1-040719 on "Clarification of charging requirements for SCUDIF", which has been sent from SA1 to SA2. As CT3 has the stage 2 responsibility for SCUDIF, CT3 believes it is necessary to study this issue in CT3. SCUDIF including the user initiated service change is part of Rel-5.

SA1 stated in the LS: "SA1 expects that it will be possible in this case to collect CDRs from the party who modifies the call." (This seems to refer for instance to a user-initiated service change from speech to multimedia). However, CT3 noticed that Section 4.3.1.1 in TS 22.115 to which SA1 is referring in their LS is actually about the IP-multimedia charging requirements. The text defines the requirements with the assumption, that the speech connection and the multimedia are parallel but separate call components, which actually is not the case in CS video-voice service change. In SCUDIF call we have either the speech or the multimedia (CS video) service on.

Up to now, CT3s working assumption is that the party setting up the SCUDIF call will be charged, and that there is no requirement for the network to enable a charging of the party initiating a service change. CT3 would like to point out that the initiator of a SCUDIF call has the possibility to reject a service change from speech to multimedia during the call.

CT3 has concerns that a change of the charged party during the call might complicate the implementation of the charging system, in particular for prepaid charging. Furthermore, inter-operator accounting might also become more complicated.

In Rel-6, network-initiated service change from multimedia to speech at the loss of resources and the subsequent network-initiated service change from speech to multimedia at the recovery of resources has been added to SCUDIF. This may complicate charging even more, if charging is reversed during the call. For instance, assuming that the callee is initiating a video call, there may be a need to reverse the charging both at network-initiated service change from multimedia to speech at the loss of resources and the subsequent network-initiated service change from speech to multimedia at the recovery of resources.

2. Actions:

To SA1 group.

ACTION: CT3 kindly asks SA1 group to confirm CT3´s working assumption that the party setting up the SCUDIF call will be charged, and that there is no requirement for the network to enable a charging of the party initiating a service change.

3. Date of Next CT3 Meetings:

CT3_37 29th August - 2nd September 2005 London

Title: **Reply LS on network initiated SCUDIF support**
Response to: **LS (R3-050356/C3-050258) on network initiated SCUDIF support from RAN3**

Source: **CT3**
To: **RAN3**

Contact Person:
Name: **Thomas Belling**
Tel. Number: +49 89 636 75207
E-mail Address: Thomas.Belling AT siemens.com

---

1. Overall Description:

CT3 would like to thank RAN3 for their LS.

CT3 has now completed their SCUDIF stage 2 work on network-initiated service change for Rel-6.
The stage 2 solution agreed in CT3 does not require any further work from RAN3 for Rel-6.

CT3 may perform further work on the question raised by RAN3 in their LS in the Release 7 time scale and contact RAN 3 if necessary.

2. Actions:

None

3. Date of Next CT3 Meetings:

CT3_37 29th August - 2nd September 2005 London