3GPP TSG-CN Meeting #26 **NP-040583**
8th ñ 10th December 2004. Athens, Greece.

Source: **TSG CN WG3**

Title: **CRs to Rel-6 on Work Item ì IMSî (Pack2)**

Agenda item: **9.16**

Document for: **APPROVAL**

Introduction:

This document contains 1 CRs to Rel-6 on Work Item ìIMSî(Pack2) that have been agreed by TSG CN WG3, and are forwarded to TSG CN Plenary for approval.

WG_tdoc	Spec	CR	R	Cat	Title	Rel	C_Ver	Work Item
N3-040874	29.163	058	2	F	Clarifications for Mn procedures for call hold	Rel-6	6.4.0	IMS-CCR-Mn

3GPP TSG-CN WG3 Meeting #34 **Tdoc ⌘N3-040874**

Seoul, Korea. 15th - 19th November 2004.

CR-Form-v7.1

CHANGE REQUEST

⌘	**29.163**	**CR 058**	⌘**rev**	**2**	⌘ Current version:	**6.4.0** ⌘

*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 [ ]	Radio Access Network [ ]	Core Network [X]

Title:	⌘ Clarifications for Mn procedures for call hold		
Source:	⌘ Siemens		
Work item code:	⌘ IMS-CCR_Mn	***Date:*** ⌘	08/11/2004
Category:	⌘ **F**	***Release:*** ⌘	Rel-6

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: ⌘ SA4 agreed a CR to TS 26.236 clarifying the RTCP handling for speech, which contains special provisions for putting media on hold and retrieving media:

ì7.4 RTP sender

The RTP sender implementation shall also include an RTCP implementation.

RTCP packets should be sent for all types of multimedia sessions except for point-to-point speech only sessions (i.e., using AMR and the AMR-WB codecs where synchronization with other RTP transported media or remote end-point aliveness information are not needed). For point-to-point speech only sessions, a UE should not send RTCP packets. Turning off RTCP can be done by setting to zero the SDP bandwidth modifiers (RR and RS) described in clause 7.1.

When RTCP is turned off (for point-to-point speech only sessions) and the media is put on hold, the terminal should re-negotiate the RTCP bandwidth with SDP bandwidth modifiers values greater than zero, and send RTCP packets to the other end, following the rules given below. This allows the remote end to detect link aliveness during hold. When media is resumed, the resuming terminal should turn off the RTCP sending again through a re-negotiation of the RTCP bandwidth with SDP bandwidth modifiers (as described in clause 7.1) equal to zero.

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This CR clarifies the resulting actions at the MGCF during the call establishment and when media are put on hold or resumed.

Summary of change:	⌘	When sending an INVITE for a CS network originated session, the O-MGCF should use the SDP RTCP bandwidth modifiers to disable RTCP. For a CS network originated hold/retrieve, the MGCF should only enable RTCP if link aliveness is required at the IM-MGW. In this case, IM-MGW interaction to enable RTCP is also required. For a IM CN subsystem originated hold/retrieve, the MGCF shall inform the IM-MGW if RTCP is temporarily enabled.
Consequences if not approved:	⌘	TS 29.163 is not in line with SA4¥s recommendation. If RTCP is used at the air interface as a consequence, this may have negative imapacts on the speech quality.

Clauses affected:	⌘	2, 7.2.3.2.2, 7.3.3.2.2, 7.4.10, 9.2.9, new Clause 9.2.10

		Y	N			
Other specs affected:	⌘		X	Other core specifications	⌘	
			X	Test specifications		
			X	O&M Specifications		

Other comments:	⌘	

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) ñ 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) ñ 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 ñ 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".

Next modified Clause

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

7.2.3.2.1 Sending of INVITE

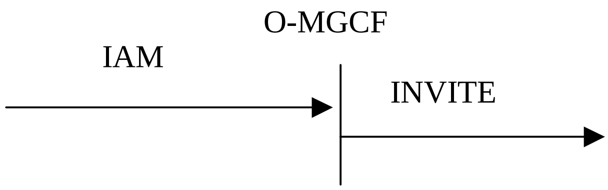

Figure 12: Receipt of an IAM (En bloc signalling in CS network)

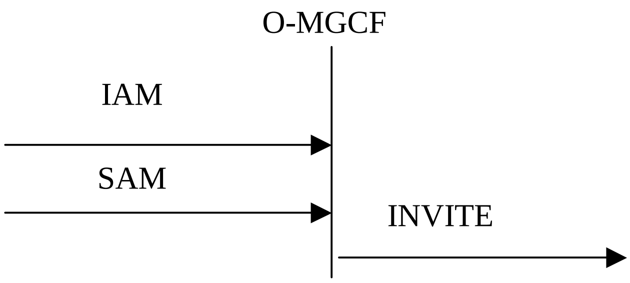

Figure 13: Receipt of an IAM (Overlap signalling in CS network)

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 with pre-conditions. 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.

7.2.3.2.2 Coding of the INVITE

7.2.3.2.2.1 REQUEST URI Header

The called party number parameter of the IAM message is used to derive Request URI of the INVITE Request. The Request URI is a tel URL and shall contain an International public telecommunication number prefixed by a "+" sign (e.g. tel:+4911231234567).

7.2.3.2.2.2 SDP Media Description

Depending on the coding of the continuity indicators different precondition information (RFC 3312 [37]) is included. If the continuity indicator indicates "continuity performed on a previous circuit" or "continuity required on this circuit", then the O-MGCF shall indicate that the precondition is not met. Otherwise the MGCF shall indicate whether the precondition is met, dependent on the possibly applied resource reservation within the IMS.

The SDP media description will contain precondition information as per RFC 3312 [37].

The O-MGCF shall include the AMR codec transported according to RFC 3267 [23] with the options listed in clause 5.1.1 of 3GPP TS 26.236 [32] in the SDP offer. Within the SDP offer, the O-MGCF should also provide SDP RR and RS bandwidth modifiers specified in IETF RFC 3556 [59] to disable RTCP, as detailed in Clause 7.4 of 3GPP TS 26.236 [32].

Table 11 provides a summary of how the header fields within the outgoing INVITE message are populated.

Table 11 ñ Interworked contents of the INVITE message

IAM→	INVITE→
Called Party Number	Request-URI
Calling Party Number	P-Asserted-Identity
	Privacy
	From
Generic Number (*"additional calling party number"*)	From
Hop Counter	Max-Forwards
TMR/USI	Message Body (application/SDP)

Next modified Clause

7.3.3.2 Outgoing Call Interworking from BICC to SIP at O-MGCF

7.3.3.2.1 Sending of INVITE

The following particularities applies for a BICC IAM received case, with regard to the already specified in clause 7.2.3.2.1.

An INVITE with precondition not yet satisfied on receipt of BICC IAM is sent.

7.3.3.2.2 Coding of the INVITE

7.3.3.2.2.1 REQUEST URI Header

See clause 7.2.3.2.2.1

7.3.3.2.2.2 SDP Media Description

The O-MGCF shall indicate that precondition is not met.

The SDP media description will contain precondition information as per RFC 3312 [37].

The O-MGCF shall include the AMR codec transported according to RFC 3267 [23] with the options listed in clause 5.1.1 of 3GPP TS 26.236 [32] in the SDP offer. Within the SDP offer, the O-MGCF should also provide SDP RR and RS bandwidth modifiers specified in IETF RFC 3556 [59] to disable RTCP, as detailed in Clause 7.4 of 3GPP TS 26.236 [32].

Next modified Clause

7.4.10 Call Hold

The service is interworked as indicated in 3GPP TS 23.228 [12].

7.4.10.1 Session hold initiated from the IM CN subsystem side

A SIP UE makes a hold request by sending an UPDATE (or re-INVITE) message with an "inactive" or a "sendonly" SDP attribute (refer to RFC 3264 [36]). Upon receipt of the hold/resume request from the IMS side, the MGCF shall send a CPG message to the CS side with a ë*remote holdí*/í*remote retrieval*í *Generic notification indicator*. The user plane interworking of the hold/resume request is described in the clause 9.2.9

7.4.10.2 Session hold initiated from the CS network side

When an MGCF receives a CPG message with a ë*remote holdí* Generic notification indicator, the MGCF shall forward the hold request by sending an UPDATE message containing SDP with ì*sendonly*î media.

When an MGCF receives a CPG message with a í*remote retrieval*í Generic notification indicator, the MGCF shall forward the resume request by sending an UPDATE message containing SDP with ì*sendrecv*î media.

If link aliveness information is required at the IM-MGW while the media are on hold, the O-MGCF should provide modified SDP RR and RS bandwidth modifiers specified in IETF RFC 3556 [59] within the UPDATE messages holding and retrieving the media to temporarily enable RTCP while the media are on hold, as detailed in Clause 7.4 of 3GPP TS 26.236 [32]. If no link aliveness information is required at the IM-MGW, the O-MGCF should provide the SDP RR and RS bandwidth modifiers previously used.

The interworking does not impact the user plane, unless the MGCF provides modified SDP RR and RS bandwidth modifiers within the UPDATE messages. If the MGCF provides modified SDP RR and RS bandwidth modifiers to the UE, the MGCF shall also provide modified SDP RR and RS bandwidths to the IM-MGW, as described in the clause 9.2.10.

Next modified Clause

9.2.9 Session hold initiated from IM CN subsystem

The network model in the clause 9.2.1 shall apply here.

Hold request

When a SIP UE makes a hold request by sending an UPDATE (or re-INVITE) message (signal 1 of figure 50), the MGCF shall request the IM-MGW to suspend sending media towards the SIP UE by changing the through-connection of the IM CN subsystem side termination to 'not through-connected' (signal 2 of figure 50). If the UE provides modified SDP RR or RS bandwidth modifiers, as specified in IETF RFC 3556 [59], within the hold request, the MGCF shall use the Configure IMS Resources Mn procedure to forward this information to the IM-MGW (not depicted in figure 50, but may be combined with signal 2). The MGCF shall send a CPG (Hold) message to the succeeding CS network node to indicate that the session is on hold (signal 4 of figure 50). Simultaneously a SIP message acknowledging the Hold request is sent to the SIP UE (signal 7 of figure 50, acknowledged by signal 7.a if the INVITE method is used). Announcements may be applied to the party on hold using the Play Announcement procedure (for BICC) or the Play TDM Announcement procedure (for ISUP, signal 5 in figure 50). The hold operation shall not block RTCP flows.

Resume request

When the SIP UE makes a request to retrieve the session on hold by sending an UPDATE (or re-INVITE) message (signal 8 of figure 50), the MGCF shall request the IM-MGW to re-establish communication towards the IMS network by changing the through-connection of the IM CN subsystem side termination to both-way through-connected (signal 11 of figure 50). If the UE provides modified SDP RR or RS bandwidth modifiers, as specified in IETF RFC 3556 [59], within the retrieve request, the MGCF shall use the Configure IMS Resources Mn procedure to forward this information to the IM-MGW (not depicted in figure 50, but may be combined with signal 11). Possible announcements to the party on hold shall be stopped using the Stop Announcement procedure (for BICC) or the Stop TDM Announcement procedure (for ISUP, signal 9 in figure 50). The MGCF shall send a CPG (Retrieve) message to the succeeding CS network node to indicate that the session is retrieved (signal 13 of figure 50).

Message sequence chart

Figure 50 shows the message sequence chart for the call hold and retrieval procedures.

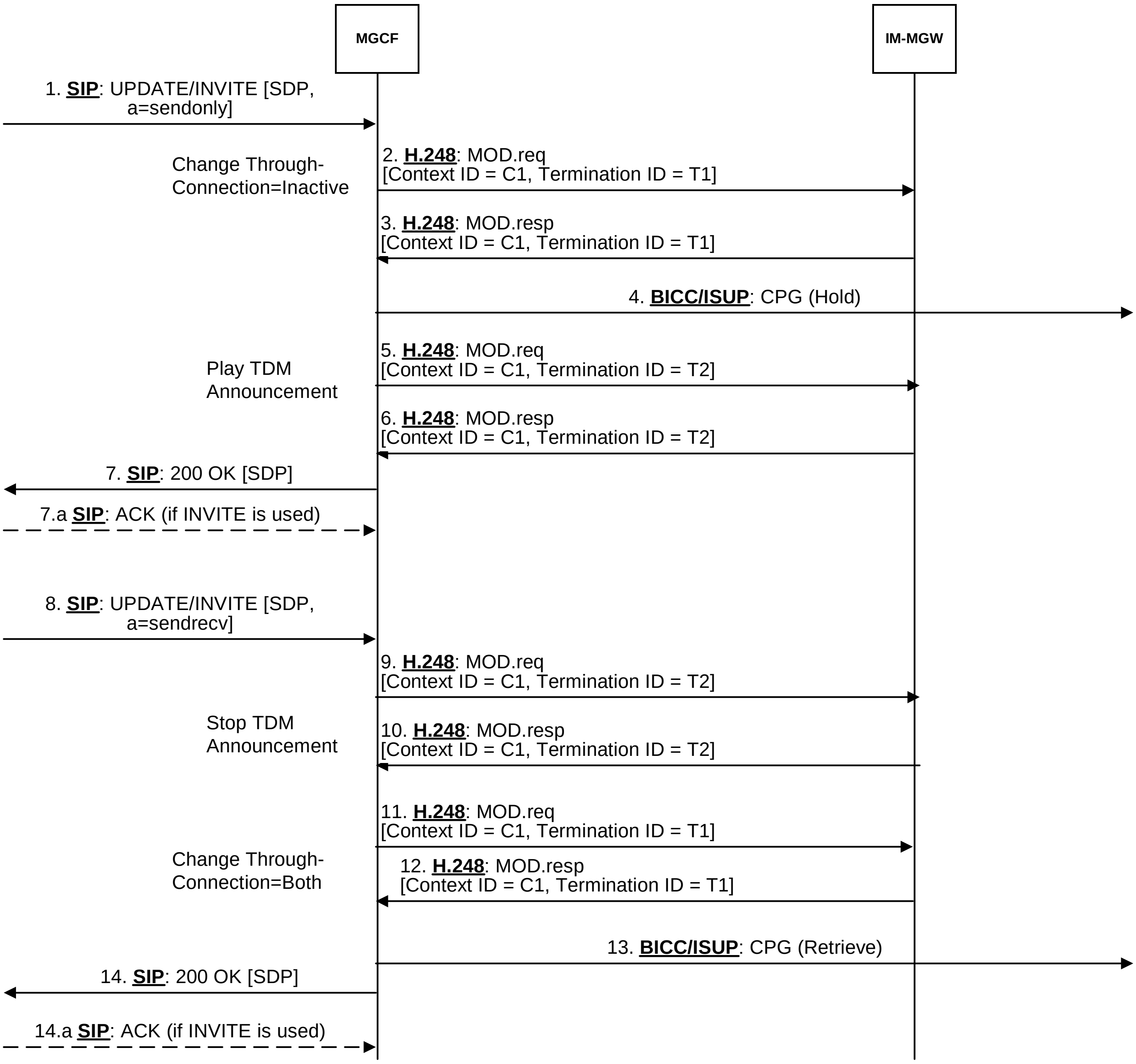

Figure 50 Session hold from IM CN subsystem

9.2.10 Session hold initiated from CS network

When an MGCF receives a CPG message with a ë*remote holdí* Generic notification indicator (signal 1 of figure 51), the MGCF forwards the hold request by sending an UPDATE message containing SDP with ì*sendonly*î media (signal 4 of figure 51).

When an MGCF receives a CPG message with a í*remote retrievalí* Generic notification indicator (signal 6 of figure 51), the MGCF forwards the resume request by sending an UPDATE message containing SDP with ì*sendrecv*î media (signal 9 of figure 51).

If link aliveness information is required at the IM-MGW while the media are on hold, the O-MGCF should provide to the modified SDP RR and RS bandwidth modifiers specified in IETF RFC 3556 [59] within the SDP offers in the UPDATE messages holding and retrieving the media to temporarily enable RTCP while the media are on hold, as detailed in Clause 7.4 of 3GPP TS 26.236 [32]. If no link aliveness information is required at the IM-MGW, the O-MGCF should provide the SDP RR and RS bandwidth modifiers previously used.

The interworking does not impact the user plane, unless the MGCF provides modified SDP RR and RS bandwidth modifiers in the UPDATE messages. If the MGCF provides modified SDP RR and RS bandwidth modifiers in the UPDATE messages, the MGCF shall also provide modified SDP RR and RS bandwidths to the IM-MGW using the Configure IMS Resoureces procedures (signals 2-3 and 7-8 of figure 51).

Message sequence chart

Figure 51 shows the message sequence chart for the call hold and retrieval procedures.

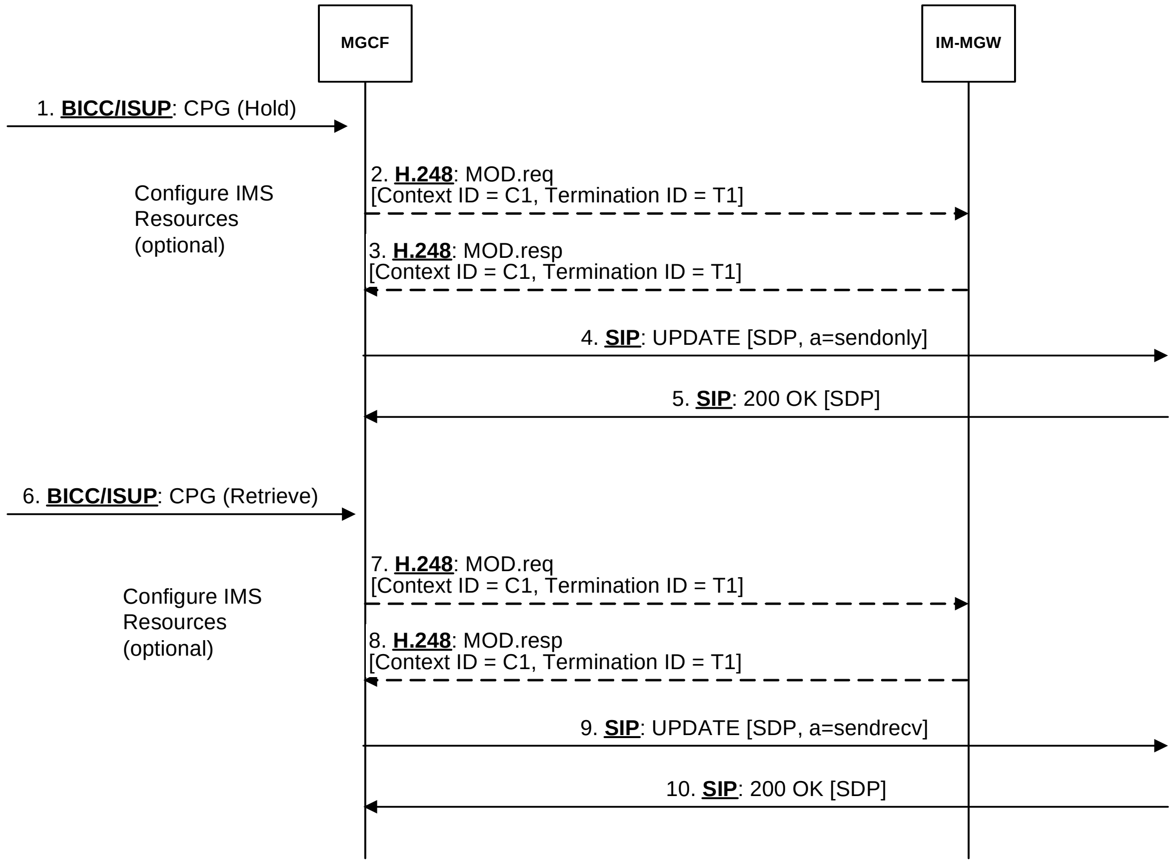

Figure 51 Session hold from CS network