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Session Initiation Protocol (SIP) to ISDN User Part (ISUP) Interworking

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Foreword

(This foreword is not part of this document).

This document was prepared by 3GPP2 TSG-X.

1 Scope

This document specifies the principles of interworking between the 3GPP2 IP Multimedia (IM) CN subsystem and ISUP based legacy CS networks, in order to support IM basic voice calls.

This document addresses the areas of control and user plane interworking between the IM CN subsystem and CS networks through the network functions, which include the MGCF and IM-MGW. For the specification of control plane interworking, areas such as the interworking between SIP and ISUP are detailed in terms of the processes and protocol mappings required for the support of both IM originated and terminated voice calls. This document concerns itself only with mappings at the upper protocol (i.e., signalling) layer and does not address lower layer (i.e., transport) interworking.

Other areas addressed encompass the transport protocol and signalling issues for negotiation and mapping of bearer capabilities and QoS information.

This document specifies the interworking between 3GPP2 profile of SIP (as detailed according to [9]) and ISUP, as specified in [73].

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of this 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 3GPP2 document, a non-specific reference implicitly refers to the latest version of that document.

[1] 3GPP TS 29.163 V7.2.0 (2006-03): *Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (Release 7)*

[2] ITU-T Recommendation H.248.1 (2002): *Gateway Control Protocol: Version 2*

[3] Void

[4] ITU-T Recommendations Q.761to Q.764 (2000): *Specifications of Signalling System Number 7 ISDN User Part (ISUP)*

[5] ITU-T Recommendation G.711: *Pulse Code Modulation (PCM) of Voice Frequencies*

[6] Void

[7] 3GPP2 X.S0013-002: *IP Multimedia Subsystem (IMS); Stage-2*

[8] 3GPP2 X.S0013-003: *IP Multimedia (IM) Session Handling; IM call model*

[9] 3GPP2 X.S0013-004: *IP Multimedia Call Control Protocol based on SIP and SDP; Stage 3*

[10] Void

[11] Void

[12] Void

[13] Void.

[14] Void

[15] Void

[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] IETF RFC 4788: *Enhancements to RTP Payload Formats for EVRC Family Codecs*

[21] Void

[22] Void

[23] Void

[24] IETF RFC 793: *Transmission Control Protocol*

[25] Void

[26] Void

[27] Void.

[28] Void.

[29] ITU-T Recommendation Q.2150.1: *Signalling Transport Converter on MTP3 and MTP3b*

[30] Void

[31] Void

[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] Void

[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] Void

[43] Void

[44] Void

[45] Void

[46] Void

[47] Void

[48] ITU-T Recommendation E.164 (May 1997): *The International Public Telecommunication Numbering Plan*

[49] Void

[50] Void

[51] Void

[52] Void

[53] Void

[54] Void

[55] Void

[56] Void

[57] Void

[58] Void

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

[60] Void

[61] Void

[62] Void

[63] Void

[64] Void

[65] Void

[67] Void

[68] Void

[69] IETF RFC 4040: *RTP Payload Format for a 64 kbit/s Transparent Call*

[70] Void

[71] Void

[72] IETF RFC 4694: *Number Portability Parameters for the "tel" URI*

[73] ANSI T1.113-2000: *Signalling System No. 7 (SS7) – Integrated Services Digital Network (ISDN) User Part*

[74] Void

[75] IETF RFC 4244: *An extension to the Session Initiation Protocol (SIP) for Request History Information*

[76] ANSI T1.611-1991 (R2003): *Signalling System Number 7 (SS7) – Supplementary Services for Non-ISDN Subscribers*

[77] ANSI T1.642-1995 (R2004): *ISDN Supplementary Service Call Deflection*

[78] ANSI T1.643-1998 (R2003): *Integrated Services Digital Network (ISDN) – Explicit Call Transfer Supplementary Service*

[79] ANSI T1.613-1991 (R2002): *Integrated Services Digital Network (ISDN) – Call Waiting Supplementary Service*

[80] ANSI T1.647-1995 (R2000): *Integrated Services Digital Network (ISDN) – Conference Calling Supplementary Service*

[81] ANSI T1.647a-1998 (R2005): *Integrated Services Digital Network (ISDN) – Conference Calling Supplementary Service – Operations Across Multiple Interfaces*

[82] ANSI T1.619-1992 (R2005): *Integrated Services Digital Network (ISDN) – Multi-Level Precedence and Preemption (MLPP) Service Capability*

[83] ANSI T1.619a-1994 (R1999): "Integrated Services Digital Network (ISDN) – Multi-Level Precedence and Preemption (MLPP) Service Capability (MLPP service domain and cause value changes)".

[84] ANSI T1.621-1992 (R2004): *Integrated Services Digital Network (ISDN) – User-to-User Signalling Supplementary Service*

[85] ANSI T1.607-2000 (R2004): *Integrated Services Digital Network (ISDN) – Layer 3 Signalling Specification for Circuit Switched Bearer Service for Digital Subscriber Signalling System Number 1 (DSS1)*

3 Definitions and Abbreviations

3.1 Definitions

For the purposes of this document, the terms and definitions given in [48] and the following apply:

SS7 signalling function: function in the CS network, which has the capabilities to transport the SS7 MTP-User part

SIP signalling function: function in the IM CN subsystem, which has the capabilities to transport SIP

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

Incoming MGCF (I-MGCF): entity that terminates incoming SIP calls from the IMS side and originates outgoing calls towards the CS side using the ISUP protocol.

Outgoing Interworking Unit (O-MGCF): entity that terminates incoming ISUP calls from the CS side and originates outgoing calls towards the IMS using SIP.

Signalling Transport Converter (STC): function that converts the services provided by a particular Signalling Transport to the services required by the Generic Signalling Transport Service.

STCmtp: Signalling Transport Converter on MTP. See [29].

3.2 Abbreviations

For the purposes of this document, the abbreviations given below apply:

ACM	Address Complete Message
ANM	ANswer Message
APRI	Address Presentation Restriction Indicator

BGCF	Breakout Gateway Control Function
CC	Country Code
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
CN	Core Network
CPG	Call ProGress message
CS	Circuit Switched
CSCF	Call Session Control Function
H/W	Hardware
IP	Internet Protocol
IM-MGW	IP Multimedia Media Gateway Function
ISDN	Integrated Services Data Network
ISUP	Integrated Services User Part
MGCF	Media Gateway Control Function
MGW	Media Gateway
MTP	Message Transfer Part
NDC	National Destination Code
NOA	Nature Of Address
PSTN	Public Switched Telephone Network
SCTP	Stream Control Transmission Protocol
SDP	Session Description Protocol
SGW	Signalling Gateway
SIP	Session Initiation Protocol
SN	Subscriber Number
SS7	Signalling System Number 7
UAC	User Agent Client
UE	User Equipment
URL	Uniform Resource Location

4 General

4.1 General Interworking Overview

The IM CN subsystem shall interwork with the ISUP based legacy CS networks (e.g., PSTN, ISDN) in order to provide the ability to support basic voice calls between a UE located in the IM CN subsystem and user equipment located in a CS network.

For the ability to support the delivery of basic voice calls between the IM CN subsystem and CS networks, basic protocol interworking between SIP [9] and ISUP (as specified in [73]) has to occur at a control plane level, in order that call setup, call maintenance and call release procedures can be supported. The MGCF shall provide this protocol mapping functionality within the IM CN subsystem.

User plane interworking between the IM CN subsystem and CS network bearers are supported by the functions within the IM-MGW. The IM-MGW resides in the IM CN subsystem and shall provide the bearer channel interconnection. The MGCF shall provide the call control to bearer setup association.

The IM CN subsystem shall interwork, at the control and user plane, with ISUP based legacy CS networks. The MGCF and IM-MGW shall support the interworking of the IM CN subsystem to an external ISUP based CS network.

5 Network Characteristics

5.1 Key Characteristics of ISUP-based CS Networks

This signalling interface to a PSTN is based on ISUP (see [73]).

5.2 Key Characteristics of IM CN Subsystem

The IM CN subsystem uses SIP to manage IP multimedia sessions in a 3GPP2 environment; it also uses IPv4 and IPv6, as defined in [16] and [39], respectively, as the transport mechanisms for both SIP session signalling and media transport. The 3GPP2 profile of SIP defining the usage of SIP within the IM CN subsystem is specified in [9]. Example call flows are provided in [8].

6 Interworking with CS Networks

6.1 Interworking Reference Model

Figure 1 details the reference model required to support interworking between the 3GPP2 IM CN subsystem and CS networks for IM basic voice calls.

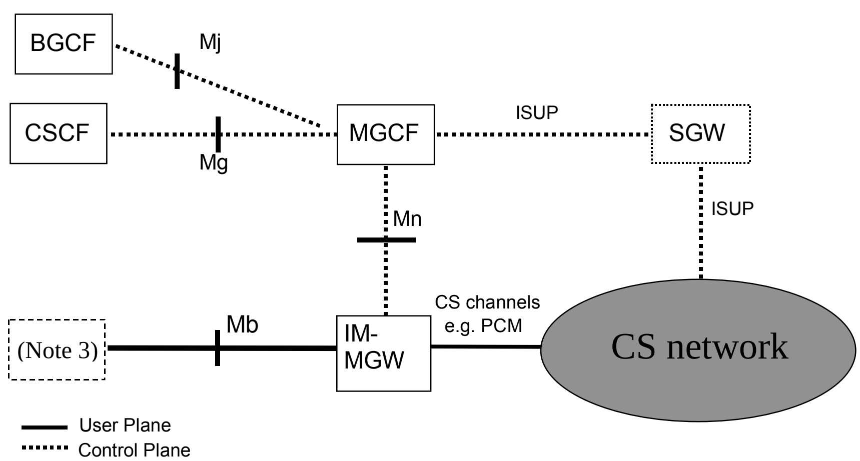


Figure 1 IM CN subsystem to CS network logical interworking reference model

NOTE 1 The logical split of the signalling and bearer path between the CS network and the IM CN subsystem is as shown, however the signalling and bearer may be logically directly connected to the IM-MGW.

NOTE 2 The SGW may be implemented as a stand-alone entity or it may be located in another entity either in the CS network or the IM-MGW. The implementation options are not discussed in this document.

NOTE 3 The IM-MGW may be connected via the Mb to various network entities, such as a UE, an MRFP, or an application server.

6.1.1 Interworking Reference Points and Interfaces

The reference points and network interfaces shown in Figure 1 are as described:

Protocol for Mg reference point: The single call control protocol applied across the Mg reference point (i.e., between CSCF and MGCF) will be based on the 3GPP2 profile of SIP as defined in accordance with [9].

Protocol for Mn reference point: The Mn reference point describes the interfaces between the MGCF and IM-MGW, and will be based on the profile of H.248 protocol defined in [33].

Protocol for Mj reference point: The single call control protocol applied across the Mj reference point (i.e., between BGCF and MGCF) will be based on the 3GPP2 profile of SIP as defined in accordance with [9].

Protocol for Mb reference point: The Mb reference point is an IP bearer facility (IPv4 or IPv6).

6.1.2 Interworking Functional Entities

6.1.2.1 Void

6.1.2.2 Media Gateway Control Function (MGCF)

This is the component within the IM CN subsystem, which controls the IM-MGW, and also performs the SIP to ISUP call related signalling interworking.

The functionality defined within MGCF shall be defined in accordance with [7].

6.1.2.3 IP Multimedia - Media Gateway Function (IM-MGW)

This is the component within the IM CN subsystem, that provides the interface between the PS domain and the CS domain, and it shall support the functions as defined in accordance with [7].

6.2 Control Plane Interworking Model

Within the IM CN subsystem, the 3GPP2 profile of SIP is used to originate and terminate IM sessions to and from the UE.

External CS networks use ISUP to originate and terminate voice calls to and from the IM CN subsystem.

Therefore, in order to provide the required interworking to enable inter network session control, the control plane protocols shall be interworked within the IM CN subsystem. This function is performed within the MGCF (see clause 6.1.2).

6.3 User Plane Interworking Model

Within the IM CN subsystem, IPv4/IPv6, and framing protocols such as RTP, are used to transport media packets to and from the IM CN subsystem entity like UE or MRFP.

External legacy CS networks use circuit switched bearer channels like TDM circuits (e.g., 64 kbit/s PCM) or IP bearers to carry encoded voice frames, to and from the IM CN subsystem.

Therefore, in order to provide the required interworking to enable media data exchange, the user plane protocols shall be translated within the IM CN subsystem. This function is performed within the IM-MGW (see clause 6.1.2).

7 Control Plane Interworking

Signalling between CS networks and the IM CN subsystem, where the associated supported signalling protocols are SS7 and IP, requires a level of interworking between the nodes across the Control Plane, i.e., the SS7 signalling function, MGCF and SIP signalling function. This interworking is required in order to provide a seamless support of a user part, i.e., SIP and ISUP.

7.1 General

The following sub-clauses define the signalling interworking between the ISDN User Part (ISUP) protocols and Session Initiation Protocol (SIP) with its associated Session Description Protocol (SDP) at a MGCF. The MGCF shall act as a Type A exchange ([73]) for the purposes of ISUP procedures. The services that can be supported through the use of the signalling interworking are limited to the services that are supported by ISUP and SIP based network domains.

The ISUP capabilities or signalling information defined for national use may be found in an annex to this document.

The capabilities of SIP and SDP that are interworked with ISUP are defined in [9]. Any interworking of ISUP messages or SIP methods not mentioned in this document is for further study.

Services that are common in SIP and ISUP network domains will seamlessly interwork by using the function of the MGCF. The MGCF will originate and/or terminate services or capabilities that do not interwork seamlessly across domains according to the relevant protocol recommendation or specification.

Table 1 lists the services seamlessly interworked and therefore are within the scope of this document.

Table 1 Interworking Capabilities between ISUP and SIP profile for 3GPP2

Service
Speech/3.1 kHz audio
En bloc address signalling
Inband transport of DTMF tones and information.
Calling Line Identification Presentation (CLIP)
Calling Line Identification Restriction (CLIR)

7.2 Interworking between CS Networks Supporting ISUP and the IM CN Subsystem

The control plane supports ISUP in the CS networks and SIP in the IM CN subsystem. One example of how this may be achieved is shown in Figure 2.

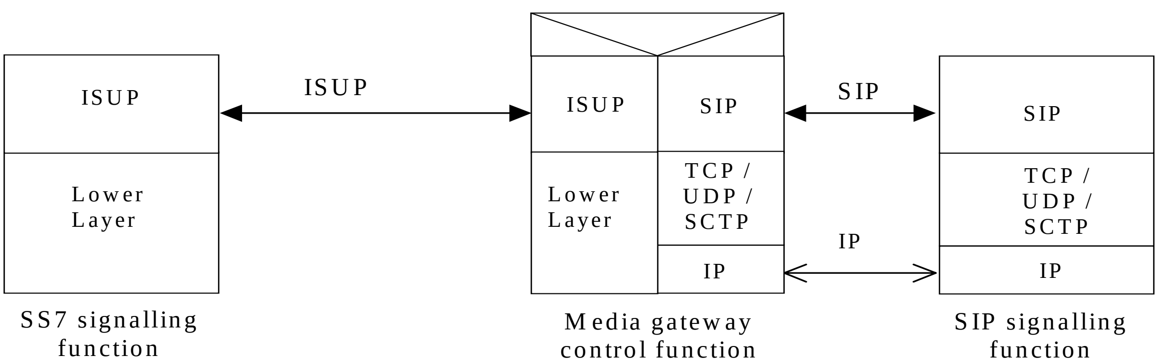


Figure 2 Control plane Interworking between CS Networks Supporting ISUP and the IM CN Subsystem

7.2.1 Services Performed by Network Entities in the Control Plane

7.2.1.1 Services Performed by the SS7 Signalling Function

The SS7 signalling function provides the capabilities to deliver or receive ISUP signalling messages across the SS7 signalling network.

7.2.1.2 Void

7.2.1.3 Services of the MGCF

The session handling and session control of the MGCF shall be as detailed in [9].

The MGCF interworking function shall provide the interaction and translation between the ISUP and SIP, where the interworking of SIP to ISUP is detailed below.

7.2.1.4 Services of the SIP Signalling Function

The SIP signalling function is a logical entity that provides the capabilities to deliver or receive multimedia session information across the IM CN subsystem signalling system.

7.2.2 Signalling Between Network Entities in the Control Plane

7.2.2.1 Signalling Between the SS7 Signalling Function and the MGCF

ISUP signalling messages are exchanged between the SS7 signalling function and the MGCF. The lower layer translation between the two entities for those signalling messages is outside of the scope of this document.

7.2.2.2 Signalling Between the MGCF and SIP Signalling Function

Signalling between the SIP signalling function and the MGCF uses the services of IP [39], and a transport protocol such as TCP [24], UDP [17], SCTP [18], plus SIP (see [9] and [19]).

The naming and addressing concepts between the MGCF and SIP signalling function shall be detailed in accordance with [7].

7.2.3 SIP-ISUP protocol interworking

When a coding of a parameter value is omitted it implies that it is not affected by the interworking, and the values are assigned by normal protocol procedures.

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

7.2.3.1.1 Sending of IAM

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

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

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

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

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

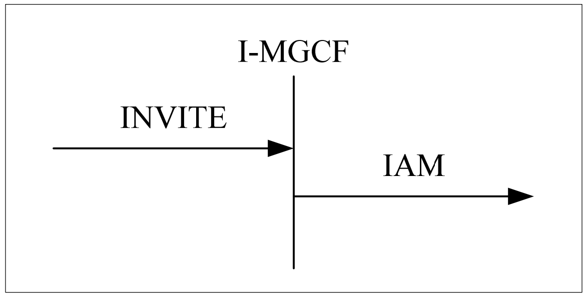


Figure 3 Receipt of an Invite Request (Continuity Procedure Supported in the ISUP Network)

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

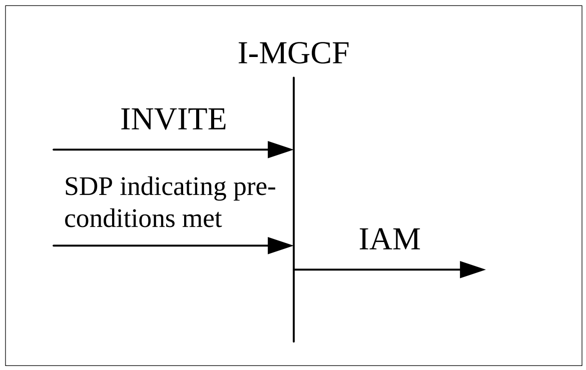


Figure 4 Receipt of an Invite request (continuity procedure not supported in the ISUP network)

The I-MGCF shall reject an INVITE request for a session only containing unsupported media or codec types by sending a status code 488 “Not Acceptable Here”. If several media streams are contained in a single INVITE request, the I-MGCF shall select one of the supported media streams, reserve the codec(s) for that media stream, and reject the other media streams and unselected codecs in the SDP answer, as detailed in [36]. If supported audio media stream(s) and supported non-audio media stream(s) are contained in a single INVITE request, an audio stream shall be selected.

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

If the INVITE message is received without an SDP (offer), then the I-MGCF shall send an SDP (offer) in the first reliable non-failure message as per [19] and [36].

7.2.3.1.2 Coding of the IAM

The following ISDN user part parameters description can be found in [73].

7.2.3.1.2.1 Called Party Number

The E.164 address encoded in the Request-URI shall be mapped to the called party number parameter of the IAM message.

Table 2 Coding of the Called Party Number

INVITE→	IAM→
Request-URI	**Called Party Number**
E.164 address (format +CC NDC SN) (This address is either: - the geographical number in the userinfo if routing number parameter does not exist in the userinfo - the routing number if the routing number parameter exists in the userinfo)	**Address Signal:** Analyse the information contained in received E.164 address. If CC is country code of the network in which the next hop terminates, then remove "+CC" and use the remaining digits to fill the Address signals. If CC is not the country code of the network in which the next hop terminates, then remove "+" and use the remaining digits to fill the Address signals.
	Odd/even indicator: set as required
	Nature of address indicator: Analyse the information contained in received E.164 address. If CC is country code of the network in which the next hop terminates, then set Nature of Address indicator to "National (significant) number. If CC is not the country code of the network in which the next hop terminates, then set Nature of Address indicator to "International number".
	Numbering plan Indicator: 001 ISDN (Telephony) numbering plan (Rec. E.164)

If the routing number parameter (following "rn=") (as defined in [72]) is not present in the userinfo component of the Request-URI, the geographic telephone number (e.g., as User info in SIP URI with user=phone, or as tel URL) shall be mapped to the Called Party Number parameter of the IAM as described in Table 2.

If the routing number parameter is present in the userinfo component of the Request-URI, the routing number parameter contained in the userinfo of the Request-URI shall be mapped to the Called Party Number parameter of the IAM. The geographic number in this case shall be mapped to the Generic Address (GAP) of the IAM, the GAP's Type of Address in this case is set to "ported number".

7.2.3.1.2.2 Nature of Connection Indicators

bits BA Satellite indicator

0 1 one satellite circuit in the connection

bits DC Continuity check indicator

0 0 continuity check not required, if the continuity check procedure is not supported in the succeeding network (Figure 4)

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

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

bit E Echo control device indicator

1 *outgoing echo control device included*

7.2.3.1.2.3 Forward Call Indicators

bits CB End-to-end method indicator

0 0 no end-to-end method available (only link-by-link method available)

bit D Interworking indicator

1 interworking encountered

bit F ISDN user part indicator

0 ISDN user part not used all the way

bits <u>HG</u> ISDN user part preference indicator

0 1 ISDN user part not required all the way

bit <u>I</u> ISDN access indicator

0 originating access non-ISDN

bits <u>KJ</u> SCCP method indicator

0 0 no indication

bit <u>M</u> Ported number translation indicator

0 number not translated

1 number translated

The value M = 1 “number translated” is used if an NP Database Dip Indicator (npdi) parameter (as defined in [72]) is present in the userinfo component of the Request-URI.

7.2.3.1.2.4 Calling Party's Category

0 0 0 0 1 0 1 0 ordinary calling subscriber

7.2.3.1.2.5 User Service Information

As a network option, either:

1) The USI parameter is set to 3.1 kHz audio and transcoding is applied when required (e.g., for 3GPP networks); or

2) If no SDP is received from the remote peer, the USI parameter fields are set as follows: Information Transfer Capability is set to 3.1 kHz audio; Information Transfer Rate is set to 64 kbit/s; and the User Information Layer 1 Protocol is set to G.711 µ-law. Transcoding is applied as required. If SDP is received from the remote peer before the IAM is sent and if transcoding is not supported at the I-MGCF, then the User Service Information (USI) parameter shall be derived from the SDP as described below and in Table 3. Otherwise they shall be set in accordance with local policy.

The I-MGCF may either transcode the selected codec(s) to the codec on the PSTN side or it may attempt to interwork the media without transcoding. If the I-MGCF does not transcode, it should map the USI and Access Transport parameters from the selected codec according to Table 3. The support of any of the media listed in Table 3 other than audio, is optional.

The SDP Media Description Part received by the I-MGCF should indicate only one media stream.

Only the “m=”, “b=”, and “a=” lines of the SDP Media Description Part are considered to interwork with the IAM USI and HLC parameters.

The first sub-field (i.e., <media> of the “m=” line will indicate one of the currently defined values “audio”, “video”, “application”, “data”, “image”, or “control”.

Further studies are needed if <media> of the “m=” line is “video”, “application”, “data” or “control”.

If the round-up bandwidth of <media> equal to audio is 64 kbps or the “b=” line is absent, then USI should be set to “3.1 kHz”, and the <transport> and <fmt-list> are evaluated to determine whether User information layer 1 protocol indicator of USI parameter should be set to “G.711 mu-law” or “G.711 A-law”.

Table 3 Coding of USI/HLC from SDP: SIP to ISUP

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

NOTE 1 In this table the codec G.711 is used only as an example. Other codecs are possible.
NOTE 2 CLEARMODE is specified in [69].
NOTE 3 HLC is normally absent in this case. It is possible for HLC to be present with the value “Telephony”, although [85], Clause 4.5.5, indicates that this would normally be accompanied by a value of “Speech” for the Information Transfer Capability element.
NOTE 4 If the b=line indicates a bandwidth greater than 64kbit/s then the call may use compression techniques or reject the call with a 415 response indicating that only one media stream of 64kbit/s is supported.
NOTE 5 <bandwidth value> for <modifier> of AS is in units of kbit/s.

7.2.3.1.2.6 Calling Party Number

The SIP "Privacy" header is defined within [40]. The SIP "P-Asserted-Identity" header is defined in [41].

Table 4 Mapping of SIP From/P-Asserted-Identity/Privacy headers to CLI parameters

Has a "P-Asserted-Identity" header field (NOTE 2, NOTE 4, NOTE 5) been received?	Has a "From" header field (NOTE 3) containing a URI that encodes an E.164 address been received (NOTE 5)?	Calling Party Number parameter Address signals	Calling Party Number parameter APRI	Generic Address (supplemental user provided calling address – not screened) address signals	Generic Address parameter APRI
No	No	Network option to either include a network provided E.164 number (See Table 5) or omit the Calling Party Number Parameter	Network option to set APRI to "presentation restricted" or "presentation allowed" (SeeTable 6)	Parameter not included	Not applicable
No	Yes	Network Option to either include a network provided E.164 number (See Table 5) or omit the Calling Party Number Parameter	Network option to set APRI to "presentation restricted" or "presentation allowed" (SeeTable 6)	Network Option to either omit the parameter (if CgPN has been omitted) or derive from the "From" header (NOTE 1) (See Table 7)	APRI = "presentation restricted" or "presentation allowed" depending on SIP Privacy header (See Table 7)
Yes	No	Derive from P-Asserted-Identity (See Table 6)	APRI = "presentation restricted" or "presentation allowed" depending on SIP Privacy header. (See Table 6)	Not included	Not applicable
Yes	Yes	Derived from P-Asserted-Identity (See Table 6)	APRI = "presentation restricted" or "presentation allowed" depending on SIP Privacy header. (See Table 6)	Network Option to either omit the parameter or derive from the "From" header (NOTE 1) (See Table 7)	APRI = "presentation restricted" or "presentation allowed" depending on SIP Privacy header (see Table 7)

NOTE 1 This mapping effectively gives the equivalent of Special Arrangement to all SIP UAC with access to the I-MGCF.

NOTE 2 It is possible that the P-Asserted-Identity header field includes both a tel URI and a sip or sips URI. In this case, the tel URI or SIP URI with user="phone". The content of the host portion is out of the scope of this document.

NOTE 3 The "From" header may contain an "Anonymous URI". An "Anonymous URI" includes information that does not point to the calling party. [19] recommends that the display-name component contain "Anonymous". [40] recommends that the Anonymous URI itself have the value "anonymous@anonymous.invalid".

NOTE 4 [7] guarantees that the received number is an E.164 number formatted as an international number, with a "+" sign as prefix.

NOTE 5 The E.164 numbers considered within this document are composed by a Country Code (CC), followed by a National Destination Code (NDC) , followed by a Subscriber Number (SN). On the IMS side, the numbers are international public telecommunication numbers ("CC"+"NDC"+"SN") and are prefixed by a "+" sign. On the CS side, it is a network option to omit the CC.

Table 5 Setting of Network-Provided ISUP Calling Party Number Parameter with a CLI (Network Option)

ISUP CgPN Parameter field	Value
Screening Indicator	*"network provided"*
Number Plan Indicator	*ISDN/Telephony (E.164)*
Address Presentation Restricted Indicator	*Presentation allowed/restricted*
Nature of Address Indicator	If next ISUP node is located in the same country set to "*National (Significant) numbe*r" else set to "International number"
Address signals	If NOA is *"national (significant) number"* no country code should be included. If NOA is "*international number",* then the country code of the network-provided number should be included.

Table 6 Mapping of P-Asserted-Identity and Privacy Headers to ISUP Calling Party Number Parameter

SIP Component	Value	ISUP Parameter / field	Value
P-Asserted-Identity header field (NOTE 1)	E.164 number	Calling Party Number	
		Numbering Plan Indicator	*"ISDN/Telephony (E.164)"*
		Nature of Address Indicator	If CC encoded in the URI is equal to the CC of the country where 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"*
		Address Presentation Restricted Indicator (APRI)	Depends on priv-value in Privacy header.
		Screening indicator	Network Provided
Addr-spec	"CC" "NDC" "SN" from the URI	Address signal	if NOA is "*national (significant) number"* then set to "NDC" + "SN" If NOA is "*international number"* Then set to "CC"+" NDC"+"SN"
Privacy header field is not present		APRI	Presentation allowed
Privacy header field	priv-value	APRI	*"Address Presentation Restricted Indicator"*
priv-value	"header"	APRI	Presentation restricted
	"user"	APRI	Presentation restricted
	"none"	APRI	Presentation allowed
	"id"	APRI	Presentation restricted
NOTE 1 It is possible that a P-Asserted –Identity header field includes both a TEL URI and a SIP or SIPS URI. In this case, either the TEL URI or SIP URI with user = "phone" and a specific host portion, as selected by operator policy, may be used.			

7.2.3.1.2.7 Generic Address

Table 7 Mapping of SIP from Header Field to ISUP Generic Address (Supplemental User Provided Calling Address – Not Screened) Parameter (Network Option)

SIP component	Value	ISUP parameter / field	Value
From header field	name-addr or addr-spec	Generic Address Type of Address	*"Supplemental user provided calling address – not screened"*
from-spec	(name-addr / addr-spec)	Nature of Address Indicator	If CC encoded in the URI is equal to the CC of the country where 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"*
		Numbering Plan Indicator	*"ISDN/Telephony (E.164)"*
		APRI	Depends on priv-value
Addr-spec	"CC" "NDC" + "SN" from the URI	Address signal	if NOA is "*national (significant) number"* then set to "NDC" + "SN" If NOA is "*international number"* Then set to "CC"+" NDC"+"SN"
Privacy header field	priv-value	APRI	*"Address Presentation Restricted Indicator"*
Use same APRI setting as for Calling Party Number.			

In the presence of the routing number and npdi parameters in the userinfo component of the Request-URI, the geographic telephone number field contained in the userinfo component of the Request-URI shall be mapped to the GAP of the IAM. The Number Qualifier Indicator (Address Type in [73]) shall be set to "ported number" (11000000). The coding of the GAP in this case is as specified in [73] and Table 8 below.

Table 8 Mapping of SIP Request-URI to ISUP generic address (ported number) parameter

SIP component	Value	ISUP parameter / field	Value
-	-	Generic Address Type of Address	*"ported number"*
-	-	Number Plan Indicator	*"ISDN/Telephony (E.164)"*
Geographical number in Userinfo	"+CC" "NDC" "SN"	Nature of Address Indicator	"national (significant) number"
		Address signal	set to "NDC" + "SN"

Note, the GAP parameter may be repeated within the IAM message as per [73]. If type of address is "ported number", the APRI field is not applicable.

7.2.3.1.2.8 Void

7.2.3.1.2.9 Original Called Number

Original Called Number parameter may be added to the IAM message if the History-Info header as defined in [75] is included in the INVITE message and it indicates that the call has been redirected at least once. Population of the Original Called Number (OCN) Parameter Field is done as shown in Table 9 below.

Table 9 Mapping of SIP History-Info Header Fields to Original Called Number (OCN)

SIP component	Value	ISUP parameter / field	Value
Hi_target_to_uri of 1st History-Info entry User portion of this URI (E.164)	+CC NDC SN	Nature of Address Indicator	If the CC is equal to the CC of the country where 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*"
		Address signal	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 component in History_info header field of the 1st History-Info entry, or as header itself (the whole History-Info header may be marked as restricted)	Privacy header field absent or "none"	APRI	"*presentation allowed*"
	"session", "header", or "history"		*"presentation restricted"*

7.2.3.1.2.10 Redirecting Number

Redirecting Number parameter may be added to the IAM message if the History-Info header as defined in [75] is included in the INVITE message and it indicates that the call has been redirected at least twice. Population of the Redirecting Number (RDN) Parameter Field is done as shown in Table 10 below.

Table 10 Mapping of SIP History-Info Header Fields to Redirecting Number

SIP component	Value	ISUP parameter / field	Value
Hi_target-to-uri of the second latest entry. User portion of this URI (E.164)	+CC NDC SN	Nature of Address Indicator	If the CC is equal to the CC of the country where 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*"
		Address signal	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
Privacy Header, priv_value component in History_info header field of the 2st latest History-Info entry, or as header itself (the whole History-Info header may be marked as restricted)	Other values or absent	APRI	"*presentation allowed*"
	"session", "header", or "history"		"*presentation restricted*"

7.2.3.1.2.11 Redirection Information

Redirection Information Parameter Field may be added to the IAM message if the OCN and/or RDN Parameter Fields have been added to this message.

Table 11 Mapping of SIP History-Info Header Fields to Redirection Information

SIP Component	ISUP parameter/field	Mapping
Reason parameter of first entry of the History-Info header (if OCN Parameter Field has been added to the IAM message)	Original Redirecting Reason	Mapping is done according to Table 12 below.
Reason parameter of second latest entry of the History-Info header (if RDN Parameter Field has been added to the IAM message)	Redirecting Reason	This field is set only if the RDN parameter has been added to the IAM message. The mapping is done according to Table 12 below.
History-Info header	Redirection counter	Index entries that are caused by call retargeting are counted and the redirection counter is set to that value.

Table 12 Mapping of Reason Parameter of the SIP History-Info Header to (Original) Redirecting Reason

Reason of History-Info header (SIP)	Redirecting Reason (ISUP)
Not Found (Cause 404)	Unknown/not available
Service Unavailable (Cause 503)	Unknown/not available
Busy Here (Cause 486)	User busy

Reason of History-Info header (SIP)	Redirecting Reason (ISUP)
Temporarily Unavailable (Cause 480)	deflection
Request Timeout (Cause 408)	No reply
Moved Temporarily (Cause 302)	unconditional
Request Terminated (487)	deflection

7.2.3.1.2.11a Jurisdiction Information

Jurisdiction Information Parameter (JIP) may be received in the P-Asserted-Identity header of the received INVITE SIP message. In that case, the JIP would be found in the "rn" parameter of the "tel" URI in the P-Asserted-Identity header as specified in [72]. If received in the INVITE message, the JIP parameter is mapped to the JIP ISUP optional parameter (as defined in [73]) in the outgoing IAM message. See Table 32 for mapping details.

Table 13 Mapping of JIP in P-Asserted-Identity Header into ISUP JIP Parameter

SIP component	Value	ISUP parameter / field	Value
P-Asserted-Identity header field Tel URI's routing number	";rn=NPANXX"	Jurisdiction Information Address signal	Set to NPA+NXX

7.2.3.1.2.12 Hop Counter (National Option)

The I-MGCF shall perform the following interworking procedure if the Hop Counter procedure is supported in the CS network.

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

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

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

Table 14 shows the principle of the mapping:

Table 14 Max Forwards -- Hop Counter

Max-Forwards	= X	Hop Counter	= INTEGER part of (X /Factor) =Y
NOTE: The Mapping of value X to Y should be done with the used (implemented) adaptation mechanism.			

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

7.2.3.1.2.12a Transit Network Selection

Based on network configuration option, if the Userinfo component of the INVITE Request URI contains "cic=" field as defined in [72], the I-MGCF may use the carrier identification code from the "cic=" field for routing the call. If the I-MGCF needs to send the ISUP Transit Network Selection (TNS) parameter in the outgoing IAM, based on network configuration option, the TNS may be populated using the carrier identification code from the "cic=" field, not including any country code present in the "cic=" field.

7.2.3.1.3 Sending of COT

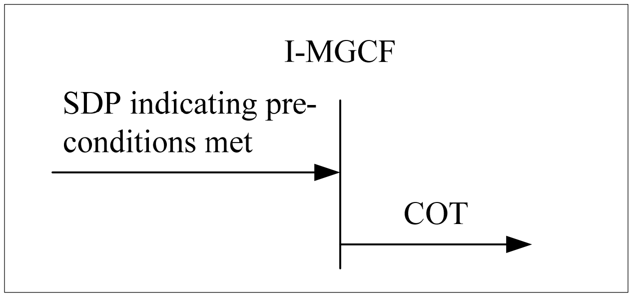


Figure 5 Sending of COT

If the IAM has already been sent, the Continuity message shall be sent indicating "continuity check successful", when all of the following conditions have been met:

- The requested preconditions (if any) in the IMS network have been met;
- A possible outstanding continuity check procedure is successfully performed on the outgoing circuit.

7.2.3.1.4 Receipt of ACM

On receipt of the ACM, the I-MGCF shall send the SIP 180 Ringing if the value of the Called Party's Status Indicator in the Backwards Call Indicator (BCI) field parameter of the ACM is set to "Subscriber Free". If the Called Party's Status Indicator is set to "no indication" or any value other than "subscriber free", the ACM is mapped to 183 Session Progress. Details are shown in Table 15.

Table 15 ACM Interworking

ACM Backward Call Indicators Field Parameter Called party's status indicator	SIP message
Subscriber free (01)	180 Ringing
Any value other than "Subscriber free"	183 Session Progress

Figure 6 shows the message flows for interworking the ACM with "subscriber free" BCI.

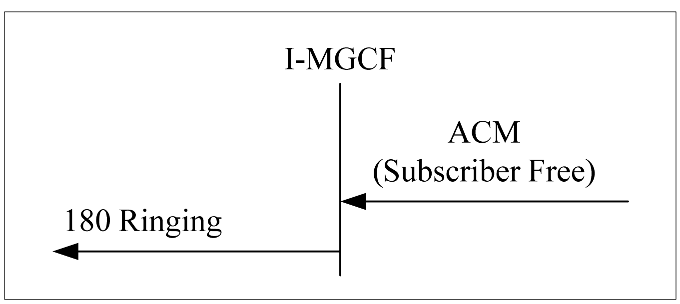


Figure 6 The Receipt of ACM ("Subscriber Free")

Figure 7 shows the message flows for interworking the ACM with a BCI other than "subscriber free".

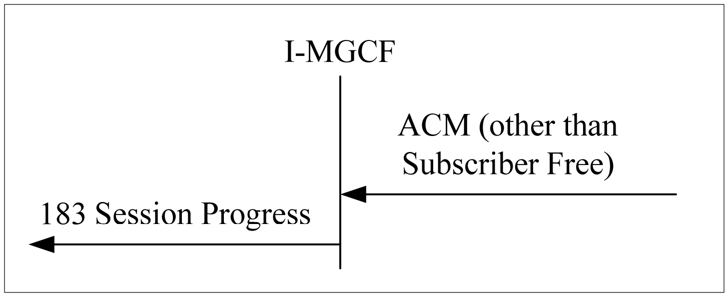


Figure 7 The Receipt of ACM (BCI other than "Subscriber Free")

7.2.3.1.5 Receipt of CPG

On receipt of the CPG, the I-MGCF shall send the 180 Ringing if the value of the event indicator is set to "alerting". If the event indicator in the CPG is set to a value other than "alerting", the CPG is not interworked. Details are shown in Table 16.

Table 16 CPG Interworking

CPG Event Indicator	SIP message
alerting (000 0001)	180 Ringing
Any value other than "alerting"	None (CPG not interworked)

Figure 8 shows the message flows for CPG interworking.

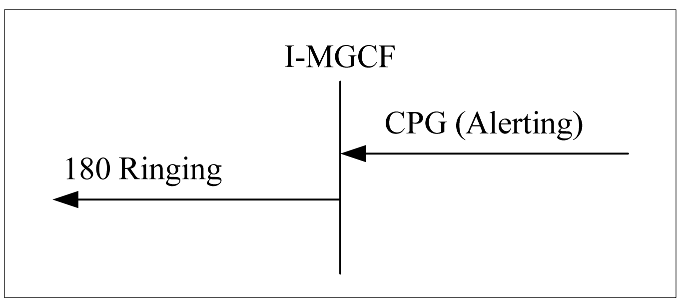


Figure 8 Receipt of CPG (Alerting)

7.2.3.1.5a Receipt of ANM

The I-MGCF shall send the 200 OK (INVITE) upon receipt of an ANM message.

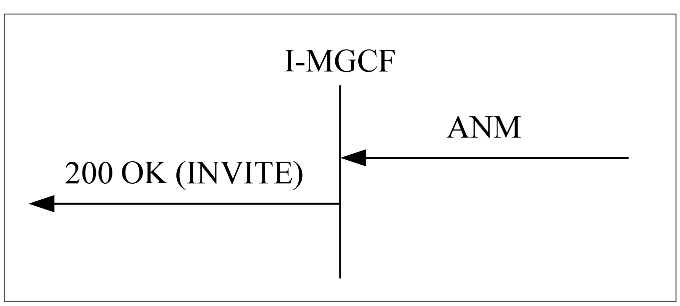


Figure 9 Receipt of ANM

7.2.3.1.6 Sending of the Release message (REL)

The following are possible triggers for sending the Release message:

- Receipt of the BYE method:

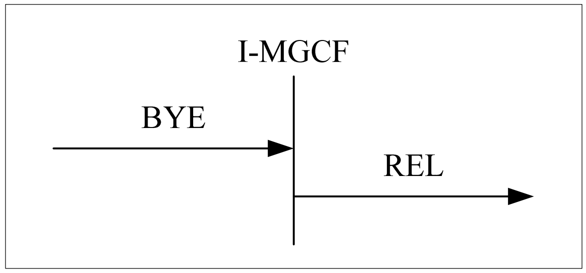


Figure 10 Receipt of the Bye method

- Receipt of the CANCEL method:

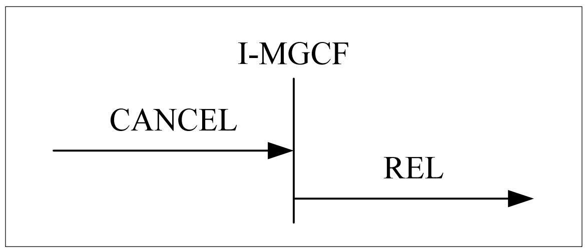


Figure 11 Receipt of Cancel method

Additional triggers are contained in Table 21.

7.2.3.1.7 Coding of the REL

If the Reason header field with Cause Value is included in the BYE or CANCEL request, then the Cause Value shall be mapped to the ISUP Cause Value field in the ISUP REL . The mapping of the Cause Indicators parameter to the Reason header is shown in Table 18. Table 17 shows the coding of the Cause Value in the REL if it is not available from the Reason header field. In both cases, the Location Field shall be set to “network beyond interworking point”.

Table 17 Coding of the REL

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

Table 18 Mapping of SIP Reason Header Fields into Cause Indicators Parameter

Component of SIP Reason header field	Component value	ISUP Parameter field	Value
Protocol	“*Q.850*”	Coding Standard	*ITU-T Standard*
Protocol	“*ANSI*”	Coding Standard	*ANSI Standard*
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 [38] or [73] (depending on value of Coding standard).			

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.

7.2.3.1.8 Receipt of the Release Message

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

NOTE According to SIP procedures, in the case that the REL message is received and a final response (e.g., 200 OK (INVITE)) has already been sent (but no ACK request has been received) on the incoming side of the I- MGCF then the I- MGCF does not send a 487 Request terminated response and instead waits until the ACK request is received before sending a BYE message.

If the REL message is received and the final response (i.e., 200 OK (INVITE)) has not already been sent, the I- MGCF shall send a Status-Code 4xx (Client Error) or 5xx (Server Error) response. The Status code to be sent is determined by examining the Cause code value received in the REL message. Table 19 specifies the mapping of the cause code values, as defined in [38] and [73], 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 [38] and [73].

Table 19 Receipt of the Release Message (REL)

← SIP Message	← REL
Status code	**Cause parameter**
Cause values with Coding Standard field set to 00 (ITUT-T standard) Note-2	
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)
No Mapping (No procedure specified for this value in US Networks)	Cause value No. 5 (Misdialled trunk prefix)
500 Server Internal Error	Cause value No. 8 (Preemption)
500 Server Internal Error	Cause value No. 9 (Preemption – circuit reserved for reuse)
486 Busy Here	Cause value No. 17 (user busy)
480 Temporarily unavailable	Cause value No 18 (no user responding)
480 Temporarily unavailable	Cause value No 19 (no answer from the user)
480 Temporarily unavailable	Cause value No. 20 (subscriber absent)
480Temporarily unavailable	Cause value No 21 (call rejected)
410 Gone	Cause value No 22 (number changed)
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)
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. 38-47) (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)

← SIP Message	← REL
Status code	**Cause parameter**
500 Server Internal error	Cause value in the Class 100 (service or option not implemented, Cause value No's. 65-79) 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. 103 (Parameter non-existent or not implemented, passed on)
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)
Cause values with Coding Standard field set to 10 (ANSI standard) (Note-2)	
404 Not Found	Cause value No. 23 (unallocated destination number)
500 Server Internal Error	Cause value No. 24 (unknown business group)
500 Server Internal Error	Cause value No. 25 (exchange routing error)
404 Not Found (Note 1)(	Cause value No. 26 (misrouted call to a ported number)
No mapping (No procedure specified for this cause value in U.S. networks)	Cause value No. 27 (Number Portability (NP) Query on Release (QoR) – number not found) (No procedures specified for this cause value in U.S. Networks)
500 Server Internal Error	Cause value in Class 010 (resource unavailable, Cause value Nos. 45 & 46)
500 Server Internal Error	Cause value in Class 011 (service or option not available, Cause value Nos. 51 & 54)
NOTE 1 Class 1 and Class 2 have the same default value. NOTE 2 The Coding Standard field in the Cause Indicators parameter in the received REL message may be set to either "ITU-T Standard" or "ANSI Standard." This table is separated into two sections pertaining to each of these values of the Coding Standard field.	

A Reason header field containing the received Cause Value of the REL shall be added to the SIP final response or BYE request sent as a result of this clause. The mapping of the Cause Indicators parameter to the Reason header is shown in Table 20.

Table 20 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
Coding Standard	*ITU-T Standard*	protocol	"*Q.850*"
Coding Standard	*ANSI Standard*	Protocol	"*ANSI*"
Cause Value	"XX" (NOTE 1)	protocol-cause	"*cause = XX*" (NOTE 1)
–	–	reason-text	Should be filled with definition text as stated in [38] or [73] (NOTE 2)

Cause indicators parameter field	Value of parameter field	component of SIP Reason header field	component value
NOTE 1 "XX" is the Cause Value as defined in [38] and [73]. NOTE 2 Due to the fact that the Cause Indicators parameter does not include the definition text as defined in [38] and [73], this is based on provisioning in the I-MGCF.			

7.2.3.1.9 Receipt of RSC, GRS or CGB (H/W Oriented)

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

1) If the final response (i.e., 200 OK (INVITE)) has already been sent, the I-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.

7.2.3.1.10 Autonomous Release at I-MGCF

Table 21 shows the trigger events at the MGCF and the release initiated by the MGCF when the call is traversing from SIP to ISUP.

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

Table 21 Autonomous Release at I MGCF

← SIP	Trigger event	REL →
Response		**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 procedures result in release after answer	According to ISUP procedures
BYE	SIP procedures result in release after answer	127 (Interworking unspecified)
484 Address Incomplete	Call release due to T7 expiry within ISUP procedures	According to ISUP procedures
480 Temporarily Unavailable	Call release due to T9 expiry within ISUP procedures	According to ISUP procedures
480 Temporarily Unavailable	Other ISUP procedures result in release before answer	According to ISUP procedures

7.2.3.1.11 Internal Through Connection of the Bearer Path

The through connection procedure is described in clause subclauses 9.2.3.1.7 and 9.2.3.2.7.

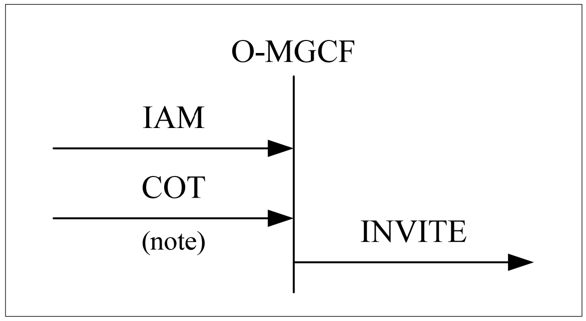
7.2.3.2 Outgoing Call Interworking from ISUP to SIP at O-MGCF

7.2.3.2.1 Sending of INVITE

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

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

If the Continuity Check indicator in the Nature of Connection Indicators parameter in the incoming IAM is set to indicate either "continuity check required on this circuit" or "continuity check performed on previous circuit", the O-MGCF should defer sending the INVITE request until receiving a COT message indicating continuity check successful.



NOTE Waiting for the COT is recommended if the Continuity Check indicator in the Nature of Connection Indicators parameter in the incoming IAM is set to indicate either "*continuity check required on this circuit*" or "*continuity check performed on previous circuit*"

Figure 12 Receipt of an IAM (En Bloc Signalling in CS network)

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

The 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 URI or SIP URI with "user=phone" and shall contain an International public telecommunication number prefixed by a "+" sign (e.g., tel:+4911231234567).

Table 22 Mapping Called Party Number and FCI Ported Number Translation Indicator (when GAP for the Ported Number is not Included) to SIP Request-URI

ISUP Parameter/field	Value	SIP Component	Value
Called party number	Digits	Request URI	Userinfo
Address signal	Either NCD + SN (national number) or CC + NCD + SN (international number)	Userinfo's geographical number	If national number, prepend +CC to Address signal digits, as in: "+CC" "NCD" "SN". If internation number, prepend "+".
Forward Call Indicators	Ported number translation indicator	Userinfo's npdi parameter	If Ported number translation indicator is equal to "1", append ";npdi" to Userinfo.
NOTE CC = Country Code of the network in which the O-MGCF is located.			

NOTE the usage of "Nature of address indicator" value "unknown" is allowed but the mapping is not specified in the present specification

If the IAM indicates that the dialled number has not been ported (GAP parameter) and the NP query has been performed (M bit of the FCI), the following procedure is applied:

- The Called Party Number parameter shall be mapped to the geographic telephone number field of the Request-URI and the To field.
- The NP Database Dip Indicator (npdi) parameter shall be appended to the userinfo of the Request-URI.

If the IAM indicates that the dialled number has been ported and has a routing number associated with it, the following procedure is applied:

- The Called Party Number parameter containing the location routing number (LRN) shall be mapped to the routing number parameter of the Request-URI.
- The Generic Address Parameter (GAP) containing the ported number shall be mapped to the geographic telephone number field of the Request-URI and the To field.

Table 23 Mapping of Generic Address (ported) and Called Party Number (When Both are Included), and FCI Ported Number to SIP Request-URI

ISUP Parameter/field	Value	SIP Component	Value
Generic Address Type of number	"ported number"	Request URI	Userinfo
Generic Address Address signal	Since NOA is "national (significant) number" then the format of the address signals is: NCD + SN	Userinfo's geographical number	Prepend +CC to Address signal digits, as in: "+CC" "NCD" "SN".
Forward Call Indicators	Ported number translation indicator	Userinfo's npdi parameter	";npdi" is added to Userinfo.
Called party number Address signal	Since NOA is "national (significant) number" then the format of the address signals is: NCD + SN	Userinfo's routing number	";rn=routing number" is added to Userinfo, with +CC being prefixed to Address signal's NCD+SN
NOTE CC = Country Code of the network in which the O-MGCF is located.			

The address signal that is used to build the geographical number in the Userinfo component of the Request-URI, is used to derive the addr-spec component of the To header field.

The NP Database Dip Indicator (npdi) parameter shall be appended to the userinfo of the Request-URI.

Based on network configuration option, the O-MGCF may follow the existing ISUP procedure for TNS to select the transit carrier. If the O-MGCF needs to send the transit network selection information to the SIP network, the Userinfo component of the SIP Request URI includes the "cic=" field (as defined in [72]). Based on network configuration option, the cic field may be populated with the carrier identification code from the TNS. Table 24 summarizes this mapping.

Table 24 Mapping of Transit Network Selection to SIP Request-URI

ISUP Parameter/field	Value	SIP Component	Value
Transit network selection (if available) Digits	4 digits, as in YYYY	Request URI Userinfo's carrier ID code	If TNS is available, cic is added to Userinfo as per [72]

Note that the "Transit Network Selection" parameter is used instead of the "Carrier identification" parameter for mapping to the Request-URI's Userinfo because the TNS, as per [73], is meant to be used for routing the call. In contrast, [73] states that the "Carrier identification" parameter is not used for routing the call.

7.2.3.2.2.2 SDP Media Description

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

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

If the O-MGCF determines that a speech call is incoming, the O-MGCF shall include the codec transported in the SDP offer. The O-MGCF may include other codecs according to operator policy.

To avoid transcoding or to support non-speech services, the O-MGCF may add media derived from the incoming ISUP information according to Table 25. The support of the media listed in Table 25 is optional.

Table 25 Coding of SDP Media Description Lines from USI: ISUP to SIP

USI parameter				HLC IE in ATP	m= line			b= line	a= line
Information Transfer Rate	Rate Multiplier	Information Transport Capability	User Information Layer 1 Protocol Indicator	High Layer Characteristics Identification	<media>	<transport>	<fmt-list>	<modifier>: <bandwidth-value>	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>
speech		“Speech”	“G.711 μ-law”	Ignore	audio	RTP/AVP	0 (and possibly 8) (NOTE 1)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000) (NOTE 1)
speech		“Speech”	“G.711 μ-law”	Ignore	audio	RTP/AVP	Dynamic PT (and possibly a second Dynamic PT) (NOTE 1)	AS:64	rtpmap:<dynamic-PT> PCMU/8000 (and possibly rtpmap:<dynamic-PT> PCMA/8000) (NOTE 1)
3.1 KHz audio		“3.1 KHz audio”	“G.711 μ-law”	(NOTE 3)	audio	RTP/AVP	0	AS:64	rtpmap:0 PCMU/8000
3.1 KHz audio		“3.1 KHz audio”		“Facsimile Group 2/3”	image	udptl	t38	AS:64	Based on T.38
3.1 KHz audio		“3.1 KHz audio”		“Facsimile Group 2/3”	image	tcptl	t38	AS:64	Based on T.38
64 kbit/s unrestricted		“Unrestricted digital inf. W/tone/ann.”	N/A	Ignore	audio	RTP/AVP	9	AS:64	rtpmap:9 G722/8000
64 kbit/s unrestricted		“Unrestricted digital information”	N/A	Ignore	audio	RTP/AVP	Dynamic PT	AS:64	rtpmap:<dynamic-PT> CLEARMODE/8000 (NOTE 2)

NOTE 1 Both PCMA and PCMU could be required.
NOTE 2 CLEARMODE is specified in [69].
NOTE 3 HLC is normally absent in this case. It is possible for HLC to be present with the value “Telephony”, although T1.706, Clause 4.5.5, indicates that this would normally be accompanied by a value of “Speech” for the Information Transfer Capability element.

Table 26 provides a summary of how the header fields within the outgoing INVITE message are populated.

Table 26 Interworked Contents of the INVITE Message

IAM→	INVITE→
Called Party Number	Request-URI
Calling Party Number	P-Asserted-Identity
	Privacy
	From
Generic Address (*"Supplemental User Provided calling address – not screened"*)	From
Original Called Number	History-Info Header
Redirecting Number	History-Info Header
Redirection Information	History-Info Header
Generic Address ("Ported Number")	Request-URI
Hop Counter	Max-Forwards
USI	Message Body (application/SDP)

7.2.3.2.2.3 P-Asserted-Identity – From and Privacy Header Fields

Table 27 Mapping ISUP CLI Parameters to SIP Header Fields

Has a Calling Party Number parameter with complete E.164 number, with Screening Indicator = UPVP or NP (See NOTE 1), and with APRI = "presentation allowed" or "presentation restricted" been received?	Has a Generic Address (supplemental user provided calling address – not screened) with a complete E.164 number with APRI = "presentation allowed" been received?	P-Asserted-Identity header field	From header field:	Privacy header field
N	N	Header field not included	SIP or SIPS URI with addr spec of "unavailable user identity" i.e., unavailable@anonymous.invalid (NOTE 4)	Header field not included
N (NOTE 3)	Y	Header field not included	addr-spec derived from Generic Address (supplemental calling address) address signals if available or network provided value (NOTE 4)	Header field not included
Y (NOTE 1)	N	Derived from Calling Party Number parameter address signals (See Table 29)	if APRI = "allowed", Tel URI or SIP URI derived from Calling Party Number parameter address signals (See Table 30) if APRI = "restricted", SIP or SIPS URI with addr spec of "anonymous user identity" i.e., anoymous@anonymous.invalid (NOTE 2) (NOTE 4)	If Calling Party Number parameter APRI = "restricted" then priv-value =: "id". For other APRI settings Privacy header is not included or if included, "id" is not included (See Table 31)

Has a Calling Party Number parameter with complete E.164 number, with Screening Indicator = UPVP or NP (See NOTE 1), and with APRI = “presentation allowed” or “presentation restricted” been received?	**Has a Generic Address (supplemental user provided calling address – not screened) with a complete E.164 number with APRI = “presentation allowed” been received?**	**P-Asserted-Identity header field**	**From header field:**	**Privacy header field**
Y	Y	Derived from Calling Party Number parameter address signals (See Table 29)	Derived from Generic Address (supplemental calling address) address signals (SeeTable 28) (NOTE 4)	If Calling Party Number parameter APRI = “restricted” then priv-value =: “id”. For other APRI settings Privacy header is not included or if included, “id” is not included (See Table 31)

NOTE 1 A Network Provided CLI in the CgPN parameter may occur on a call to IMS. Therefore in order to allow the “display” of this Network Provided CLI at a SIP UAS it shall be mapped into the SIP From header. It is also considered suitable to map into the P-Asserted-Identity header since in this context it is a fully authenticated CLI related exclusively to the calling line, and therefore as valid as a User Provided Verified and Passed CLI for this purpose.

NOTE 2 The “From” header may contain an “Anonymous URI”. An “Anonymous URI” includes information that does not point to the calling party. [19] recommends that the display-name component contains “Anonymous”. The Anonymous URI itself should have the value “anonymous@anonymous.invalid”.

NOTE 3 This combination of CgPN and supplemental calling address is an error case or will occur when the CgPN APRI is “presentation restricted by network and this is shown here to ensure consistent mapping across different implementations.

NOTE 4 In accordance with procedures in [19], a tag shall be added to the “From” header.

Table 28 Mapping of Generic Address (Supplemental User Provided Calling Address – Not Screened) to SIP From Header Fields

ISUP parameter / field	Value	SIP component	Value
Generic Address Type of Address	"*supplemental user provided calling address – not screened*"	From header field	display-name (optional) and addr-spec
Nature of Address Indicator	"*national (significant) number*"	Tel URI or SIP URI	Add CC (of the country where the MGCF is located) to GAP address signals to construct E.164 number in URI. Prefix number with "+".
	"*international number*"		Map complete GAP address signals to E.164 number in URI. Prefix number with "+".
Address signal	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	Tel URI or SIP URI	CC+NDC+SN as E.164 number in URI. Prefix number with "+".

Table 29 Mapping of Calling Party Number Parameter to SIP P-Asserted-Identity Header Fields

ISUP Parameter / field	Value	SIP component	Value
Calling Party Number		P-Asserted-Identity header field	
Nature of Address Indicator	"*national (significant) number*"	Tel URI or SIP URI	Add CC (of the country where the MGCF is located) to CgPN address signals to construct E.164 number in URI. Prefix number with "+".
	"*international number*"		Map complete CgPN address signals to E.164 number in URI. Prefix number with "+".
Address signal	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	Tel URI or SIP URI	CC+NDC+SN as E.164 number in URI. Prefix number with "+".

Table 30 Mapping of ISUP Calling Party Number Parameter to SIP From Header Fields

ISUP parameter / field	Value	SIP component	Value
Calling Party Number		From header field	
Nature of Address Indicator	*"national (significant) number"*	Tel URI or SIP URI (NOTE 1)	Add CC (of the country where the MGCF is located) to CgPN address signals then map to construct E.164 number in URI. Prefix number with "+".
	"international number"		Map complete CgPN address signals to construct E.164 number in URI. Prefix number with "+".
Address signal	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	Tel URI or SIP URI (NOTE 1)	CC+NDC+SN as E.164 number in URI. Prefix number with "+".
NOTE 1 A tel URI or a SIP URI with "user=phone" is used according to operator policy.			

Table 31 Mapping of ISUP APRIs into SIP Privacy Header Fields

ISUP parameter / field	Value	SIP component	Value
Calling Party Number		Privacy header field	priv-value
APRI (See to determine which APRI to use for this mapping)	*"presentation restricted"*	Priv-value	"id" ("id" included only if the P-Asserted-Identity header is included in the SIP INVITE)
	"presentation allowed"	Priv-value	omit Privacy header or Privacy header without "id" if other privacy service is needed
NOTE When Calling Party Number parameter exists, P-Asserted-Identity header is always derived from it as in Table 29.			

If the Jurisdiction Information Parameter (JIP) was received in the IAM message, it shall be mapped to the P-Asserted-Identity (PAI) header. The JIP is placed in the "rn" parameter of the "tel" URI in the P-Asserted-Identity header as specified in [72]. See Table 32 for mapping details.

Table 32 Mapping of ISUP JIP into SIP P-Asserted-Identity Header Fields

ISUP parameter / field	Value	SIP component	Value
Jurisdiction Information Address signal	NPA+NXX	P-Asserted-Identity header field Tel URI's routing number	";rn=NPANXX" is added to the Tel URI in the P-Asserted-Identity header.

If the JIP is received in the IAM message, the PAI header must be included in the INVITE message to carry the JIP parameter. If the Calling Party Number parameter is not received in the IAM message, then a dummy tel URI is constructed with "rn" parameter set to the JIP value (e.g., tel:000000000000000;rn=NPANXX) and then the dummy tel URI is included in the PAI header. In case a dummy tel URI is placed in the PAI, a privacy header with privacy value of "id" is added to the message so that the dummy tel URI is not rendered to the user.

7.2.3.2.2.4 History-Info Header

A History-Info header as defined in [75] may be added to the INVITE message if the OCN and/or the RDN Parameter Field is included in the received IAM message. Table 33, Table 34, and Table 35 show how the History-Info header is populated.

Table 33 Mapping of Original Called Number (OCN) to SIP History-Info Header Fields

ISUP parameter / field	Value	SIP component	Value
Nature of Address Indicator	"national (significant) number"	User portion of the URI (E.164) of the first entry in the History-Info header	Add CC (of the country where the MGCF is located) to the address signals. Prefix number with "+".
	"*international number"*		Prefix address signal with "+".
Address signal	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		CC+NDC+SN as E.164 number in URI. Prefix number with "+".
APRI	"*presentation allowed"*	Privacy Header of the first entry in the History-Info header Priv-value:	Header absent or "none"
	"presentation restricted"		"history"
		Index of the first entry in the History-Info header	1 (this is the first entry in the History-Info header)

Table 34 Mapping of Redirecting Number to SIP History-Info Header Fields

ISUP parameter / field	Value	SIP component	Value
Nature of Address Indicator	"national (significant) number"	User portion of the URI (E.164) that correspond of the second entry in the History-Info header	Add CC (of the country where the MGCF is located) to the address signals. Prefix number with "+".
	"*international number"*		Prefix address signal with "+".
Address signal	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		CC+NDC+SN as E.164 number in URI. Prefix number with "+".
APRI	"*presentation allowed"*	Privacy Header that corresponds to the second entry in the History-Info header Priv-value:	Header absent or "none"
	"presentation restricted"		"history"
		Index corresponds to the second entry in the History-Info header	See "Mapping of the Redirection Information"

Table 35 Mapping of Redirection Information to SIP History - Info Header Fields

ISUP parameter/field	SIP Component	Mapping
Original Redirecting Reason	Reason parameter of the first History-Info header entry	According to Table 36 below.
Redirecting Reason	Reason parameter of second History-Info header entry	According to Table 36 below.
Redirection counter	Index of the second History-Info header entry (in case Redirecting Number is provided in the IAM message)	Set the index of the second entry of the History-Info to reflect the redirection counter value (there may be a gap between the first entry's index and the second entry's index).

Table 36 Mapping of (Original) Redirecting Reason to Reason parameter of the SIP History-Info header

Redirecting Reason (ISUP)	Reason of History-Info header (SIP)
Unknown/not available	Service Unavailable (Cause 503)
User busy	Busy Here (Cause 486)
No reply	Request Timeout (Cause 408)
Unconditional	Moved Temporarily (Cause 302)
Deflection	Moved Temporarily (Cause 302)

7.2.3.2.2.5 Max Forwards Header

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

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

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

The Table 37 shows the principle of the mapping:

Table 37 Hop counter-Max Forwards

Hop Counter	= X	Max-Forwards	= Y = Integer part of (X * Factor)
NOTE The Mapping of value X to Y should be done with the used (implemented) adaptation mechanism.			

The factor used to map from Hop Counter to Max-Forwards for a given call will depend on call origin, and will be provisioned at the O-MGCF based on network topology, trust domain rules, and bilateral agreement.

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

7.2.3.2.3 Receipt of CONTINUITY

This clause only applies if the O-MGCF has sent the INVITE request without waiting for an outstanding COT message (see Clause 7.2.3.2.1).

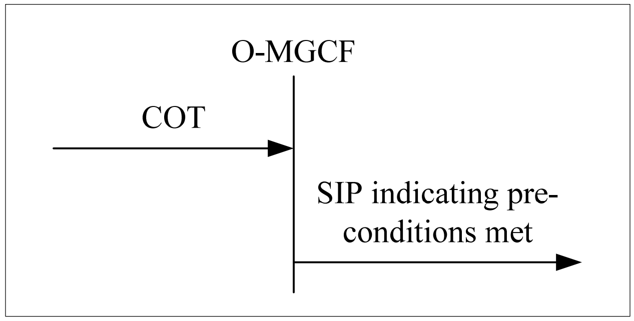


Figure 13 Receipt of COT (Success)

When the requested preconditions in the IMS (if any) have been met and if possible outstanding continuity procedures have successfully been completed (COT with the Continuity Indicators parameter set to "continuity check successful" is received), a SDP offer (e.g., a SIP UPDATE request) shall be sent for each early SIP dialogue confirming that all the required preconditions have been met.

7.2.3.2.4 Sending of ACM and Awaiting Answer Indication

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

- the reception of the first 180 Ringing or,

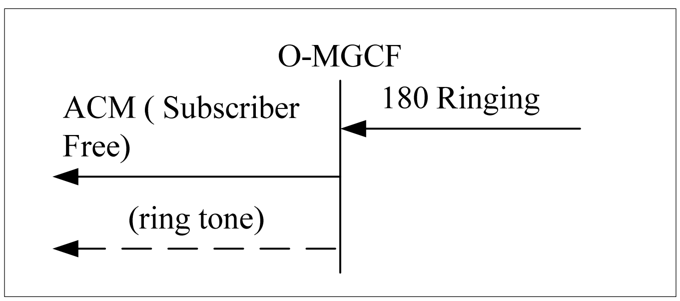


Figure 14 Sending of ACM (Receipt of first 180 ringing)

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

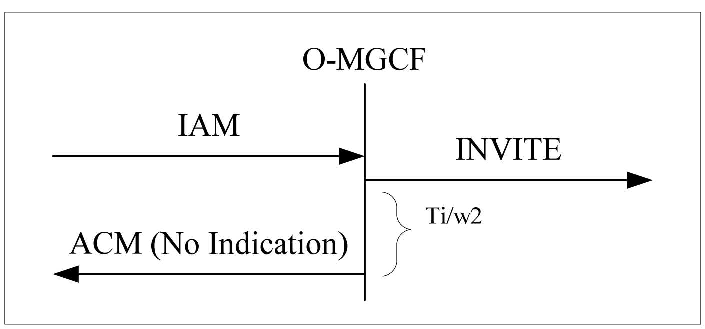


Figure 15 Sending of ACM (Ti/w_2 elapses)

- the reception of the first 183 Session Progress.

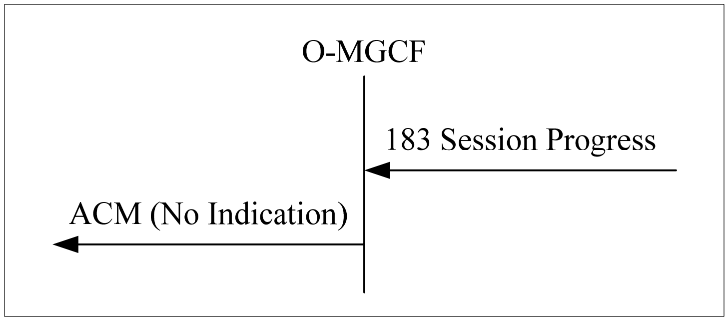


Figure 16 Sending of ACM (Receipt of 183 Session Progress)

The sending of an awaiting answer indication is described in clause 9.2.3.3

7.2.3.2.5 Coding of the ACM

The description of the following ISDN user part parameters can be found in [73].

7.2.3.2.5.1 Backward call indicators

bits	BA	Charge indicator
	0 0	no indication
bits	DC	Called party's status indicator
	0 1	*subscriber free* if the 180 Ringing has been received.
	0 0	*no indication* otherwise
bits	FE	Called party's category indicator
	0 0	no indication
bits	HG	End-to-end method indicator
	0 0	no end-to-end method available
bit	I	Interworking indicator
	1	interworking encountered
bit	J	IAM segmentation indicator
	0	no indication
bit	K	ISDN user part indicator
	0	ISDN user part not used all the way
bit	L	Holding indicator (national use)
	0	holding not requested
bit	M	ISDN access indicator
	0	terminating access non-ISDN

7.2.3.2.6 Sending of the Call Progress Message (CPG)

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

- the first SIP 180 Ringing provisional response.

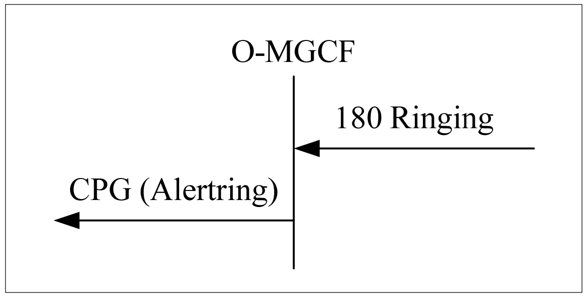


Figure 17 Sending of CPG (Alerting)

7.2.3.2.7 Coding of the CPG

The description of the following ISDN user part parameters can be found in [4].

7.2.3.2.7.1 Event Information

bits <u>G-A</u>	Event indicator
0 0 0 0 0 0 1	*alerting*

7.2.3.2.7a Receipt of 200 OK (INVITE)

Upon receipt of the first 200 OK (INVITE), the O-MGCF shall send an Answer Message (ANM) as described in clauses 7.2.3.2.8 and 7.2.3.2.9.

The O-MGCF shall not progress any further early dialogues to established dialogues. Therefore, upon the reception of a subsequent final 200 (OK) response for any further dialogue for an INVITE request (e.g., due to forking), the O-MGCF shall:

1) acknowledge the response with an ACK request; and

2) send a BYE request to this dialog in order to terminate it.

7.2.3.2.8 Sending of the Answer Message (ANM)

Upon receipt of the first 200 OK (INVITE), the O-MGCF shall send the Answer Message (ANM) to the preceding exchange.

NOTE Through connection and the stop of awaiting answer indication are described in clause 9.2.3.3

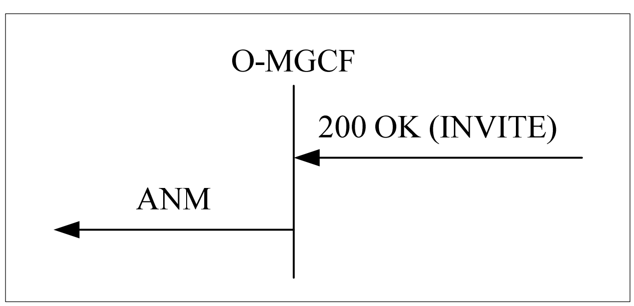


Figure 18 Sending of ANM

7.2.3.2.9 Coding of the ANM

7.2.3.2.9.1 Backwards Call Indicators

If Backwards Call Indicators are included in the ANM, then the coding of these parameters shall be as described in clause 7.2.3.2.5.1.

7.2.3.2.10 Void

7.2.3.2.11 Void

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

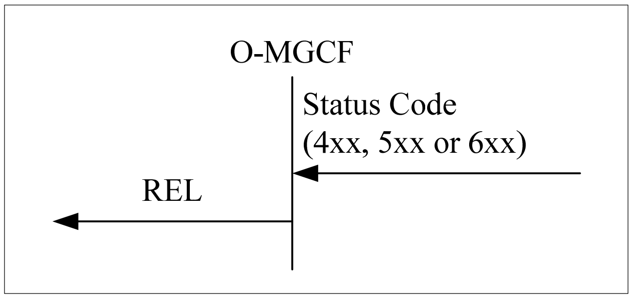


Figure 19 Receipt of Status Codes 4xx, 5xx or 6xx

If a Reason header is included in a 4XX, 5XX, 6XX response, then the Cause Value of the Reason header shall be mapped to the ISUP Cause Value field in the ISUP REL message. The mapping of the Reason header to the Cause Indicators parameter is shown in Table 18 (see 7.2.3.1.7). Otherwise coding of the Cause parameter value in the REL message is derived from the SIP Status code received according to Table 38. The Cause Parameter Values are defined in [38] and [73].

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 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 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 38 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

←REL (cause code)	←4xx/5xx/6xx SIP Message
127 (interworking unspecified)	408 Request Timeout
22 (Number changed)	410 Gone
127 (interworking unspecified)	413 Request Entity too long
127 (interworking unspecified)	414 Request-URI too long
127 (interworking unspecified)	415 Unsupported Media type
127 (interworking unspecified)	416 Unsupported URI scheme
127 (interworking unspecified)	420 Bad Extension
127 (interworking unspecified)	421 Extension required
127 (interworking unspecified)	423 Interval Too Brief
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
No mapping (NOTE 3)	491 Request Pending
127 (interworking unspecified)	493 Undecipherable
127 (interworking unspecified)	500 Server Internal error
127 (interworking unspecified)	501 Not implemented
127 (interworking unspecified)	502 Bad Gateway
127 (interworking unspecified)	503 Service Unavailable
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 responses that are not covered in this table are not interworked.
NOTE 3 This response does not terminate a SIP dialog, but only a specific transaction within it.

7.2.3.2.12.1 Void

7.2.3 2.13 Receipt of a BYE

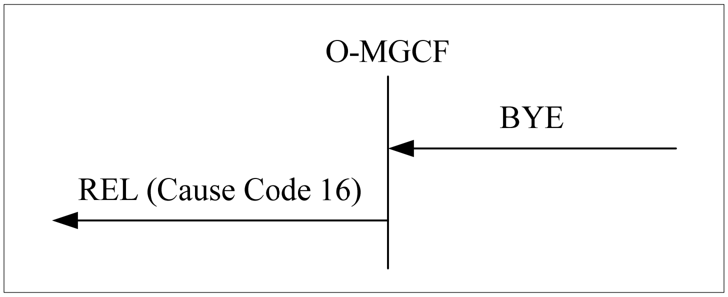


Figure 20 Receipt of BYE Method

If a Reason header field with Cause Value is included in the BYE request, then the Cause Value shall be mapped to the ISUP Cause Value field in the ISUP REL. The mapping of the Reason header to the Cause Indicators parameter is shown in Table 18 (see 7.2.3.1.7). On receipt of a BYE request, the O-MGCF sends a REL message with Cause Code value 16 (Normal Call Clearing).

7.2.3.2.14 Receipt of the Release Message

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

A Reason header field containing the received 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 20 (see 7.2.3.1.8).

7.2.3.2.15 Receipt of RSC, GRS or CGB (H/W Oriented)

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

Editor's Note: It is FFS whether to indicate the cause value for internal error in the network to the user.

7.2.3.2.16 Autonomous Release at O-MGCF

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 a 200 OK (INVITE) is received.

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

Editor's Note: It is FFS whether to indicate the cause value for internal error in the network to the user.

Table 39 Autonomous Release at O-MGCF

REL ← Cause parameter	Trigger event	→ SIP
As determined by ISUP procedure	COT received with the Continuity Indicators parameter set to "*continuity check failed*" or the 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 ISUP procedure	ISUP procedures result in generation of autonomous REL on 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

7.2.3.2.17 Special Handling of 580 Precondition Failure Received in Response to Either an INVITE or UPDATE

A 580 Precondition failure response may be received as a response either to an INVITE or to an UPDATE request.

7.2.3.2.17.1 580 Precondition Failure Response to an INVITE

Release with cause code as indicated in Table 38 is sent immediately to the ISUP network.

7.2.3.2.17.2 580 Precondition Failure Response to an UPDATE within an Early Dialog

Release with Cause Code '127 Interworking' is sent immediately to the ISUP network. A BYE request is sent for the INVITE transaction within which the UPDATE was sent.

7.2.3.2.18 Sending of CANCEL

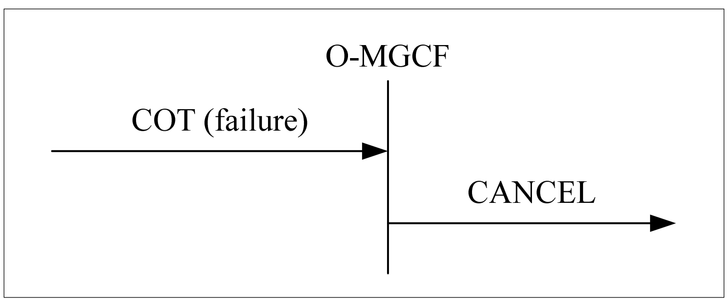


Figure 21 Receipt of COT (Failure).

CANCEL shall be sent if the Continuity message is received with the Continuity Indicators parameter set to "continuity check failed" or the ISUP timer T8 expires.

7.2.3.2.19 Receipt of SIP Redirect (3xx) Response

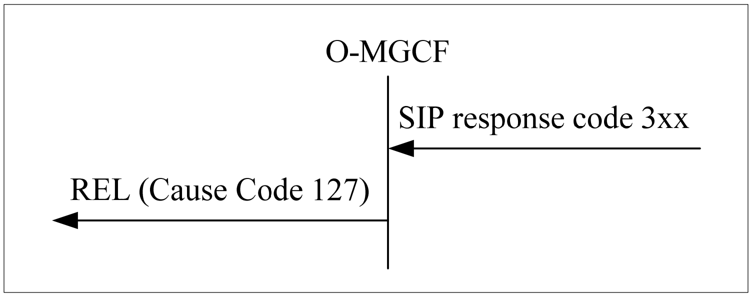


Figure 22 Receipt of SIP Response Code 3xx

When receiving a SIP response with a response code 3xx, the default behaviour of the O-MGCF is to release the call with a cause code value 127 (Interworking unspecified).

NOTE: The O-MGCF may also decide for example to redirect the call towards the URIs in the Contact header field of the response as an operator option, but such handling is outside of the scope of this document.

7.2.3.3 Timers

Table 40 Timers for Interworking

Symbol	Time-out value	Cause for initiation	Normal termination	At expiry	Reference
Ti/w2	15 s to 20 s (default of 15 s)	When INVITE is sent unless the ACM has already been sent.	On reception of 180 Ringing , or 404 Not Found or 484 Address Incomplete for an INVITE transaction, or 200 OK (INVITE).	Send ACM (no indication)	7.2.3.2.4 7.2.3.2.1 (NOTE 1)
NOTE 1	This timer is used to send an early ACM if a delay is encountered in receiving a response from the subsequent SIP network.				

7.3 Void

7.4 Supplementary Services

The following sub-clauses describe the MGCF behaviour related to supplementary services as defined in. The support of these supplementary services is optional. If the supplementary services are supported, the procedures described within this clause shall be applied.

7.4.1 Calling Line Identification Presentation/Restriction (CLIP/CLIR)

The inter working between the Calling Party Number parameter and the P-Asserted-ID header and vice versa used for the CLIP-CLIR service is defined in the clauses 7.2.3.1.2.6 and 7.2.3.2.2.3. This inter working is essentially the same as for basic call and differs only in that if the CLIR service is invoked the “Address Presentation Restriction Indicator (APRI)” (in the case of ISUP to SIP calls) or the “priv value” of the “calling” Privacy header field (in the case of SIP to ISUP calls) is set to the appropriate “restriction/privacy” value.

In the specific case of ISUP originated calls, use of the CLIP service additionally requires the ability to determine whether the number was network provided or provided by the access signalling system. Due to the possible SIP indication of the P-Asserted-Identity the Screening indicator is set to network provided as default. For the CLIP-CLIR service the mapping of the APRI from privacy header at the O-MGCF is described within Table 31 in Clause 7.2.3.2.2.3.

At the O-MGCF the presentation restricted indication shall be mapped to the privacy header = “id” and “header”. This is described in Table 6 in clause 7.2.3.1.2.3.

7.4.2 COLP/COLR

The COLP/COLR services are not supported by [73].

7.4.3 Void

7.4.4 Void

7.4.5 Void

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

The actions of the MGCF at the ISUP side are described in [76]. The service shall be terminated at the MGCF and the call shall continue according to the basic call procedures.

7.4.7 Call Deflection (CD)

The actions of the MGCF at the ISUP side are described in [77]. The service shall be terminated at the MGCF and the call shall continue according to the basic call procedures.

7.4.8 Explicit Call Transfer (ECT)

The actions of the MGCF at the ISUP side are described in [78]. The service shall be terminated at the MGCF and the call shall continue according to the basic call procedures.

7.4.9 Call Waiting

The actions of the MGCF at the ISUP side are described in [79]. The service shall be terminated at the MGCF and the call shall continue according to the basic call procedures.

7.4.10 Call Hold

The service is interworked as indicated in [7].

7.4.10.1 Session Hold Initiated from the IM CN Subsystem Side

The IMS network makes a hold request by sending an UPDATE or re-INVITE message with an “inactive” or a “sendonly” SDP attribute (refer to [36]) , depending on the current state of the session. Upon receipt of the hold request from the IMS side, the MGCF shall send a CPG message to the CS side with a ‘remote hold’ Notification Indicator. To resume the session, the IMS side sends an UPDATE or re-INVITE message with a “recvonly” or “sendrecv” SDP attribute, depending on the current state of the session. Upon receipt of the resume request from the IMS side, the MGCF shall send a CPG message to the CS side with a ‘remote hold released’ Notification Indicator. However, the I-MGCF shall not send a CPG message upon reception of SDP containing “inactive” media within an initial INVITE request establishing a new SIP dialogue and upon reception of the first subsequent SDP activating those media.

The user plane interworking of the hold/resume request is described in the clause 9.2.9.

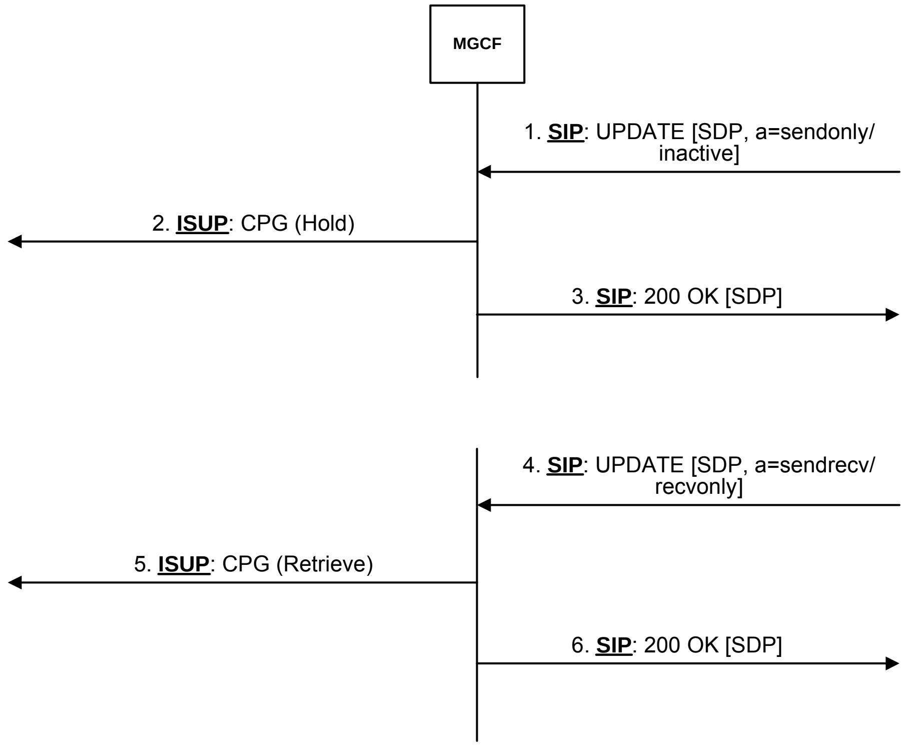


Figure 23 Session hold/resume initiated from the IM CN subsystem side

7.4.10.2 Session Hold Initiated from the CS Network Side

When an MGCF receives a CPG message with a ‘remote hold’ Notification Indicator and the media on the IMS side are not “sendonly” or “inactive”, the MGCF shall forward the hold request by sending an UPDATE or re-INVITE message containing SDP with “sendonly” or “inactive” media, as described in [36].

When an MGCF receives a CPG message with a ’remote hold released’ Generic Notification indicator and the media on the IMS side are not “sendrecv” or “recvonly”, the MGCF shall forward the resume request by sending an UPDATE or re-INVITE message containing SDP with “sendrecv” or “recvonly” media, as described in [36].

If the MGCF receives a CPG with ‘remote hold’ or ‘remote hold released’ before answer, it shall forward the request using an UPDATE message. If the MGCF receives a CPG with ‘remote hold’ or ‘remote hold released’ after answer, it should forward the request using re-INVITE but may use UPDATE.

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 [59] within the UPDATE or re-INVITE messages holding and retrieving the media to temporarily enable RTCP while the media are on hold, as detailed in Clause 7.4 of [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 or re-INVITE messages. If the MGCF provides modified SDP RR and RS bandwidth modifiers to the IMS side, the MGCF shall also provide modified SDP RR and RS bandwidths to the IM-MGW, as described in the clause 9.2.10.

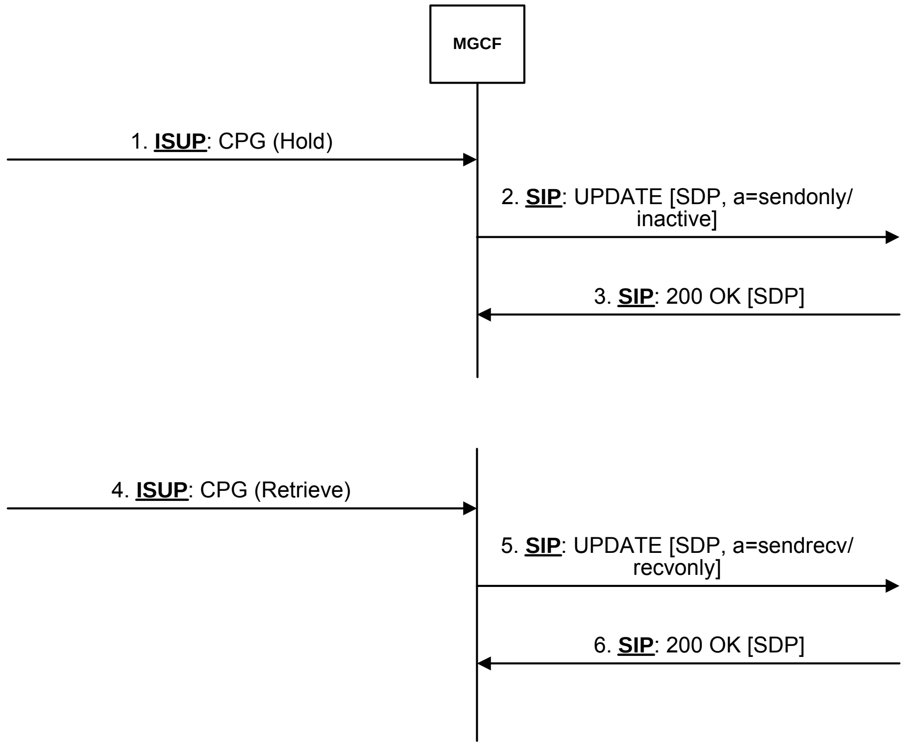


Figure 24 Session Hold/Resume Initiated from the CS Network Side

7.4.11 Void

7.4.12 Void

7.4.13 Void

7.4.14 Conference Calling (CONF) / Three-Party Service (3PTY)

The actions of the MGCF at the ISUP side are described in [80] and [81].

Table 41 Mapping between ISUP and SIP for the Conference Calling (CONF) and Three-Party Service (3PTY) Supplementary Service

ISUP message	Mapping
CPG with a “Conference established” Notification indicator	As described for CPG message with a ‘remote hold release’ Notification indicator in Subclause 7.4.10.2
CPG with a “Conference disconnected” Notification indicator	As described for CPG message with a ‘remote hold release’ Notification indicator in Subclause 7.4.10.2
CPG with an “isolated” Notification indicator	As described for CPG message with a ‘remote hold’ Notification indicator in Subclause 7.4.10.2
CPG with a “reattached” Notification indicator	As described for CPG message with a ‘remote hold release’ Notification indicator in Subclause 7.4.10.2

7.4.15 Void

7.4.16 Void

7.4.17 Multi-Level Precedence and Pre-emption (MLPP)

The actions of the MGCF at the ISUP side are described in [82] and [83]. The service shall be terminated at the MGCF and the call shall continue according to the basic call procedures.

7.4.18 Void

7.4.19 Void

7.4.20 Void

7.4.21 User-to-User Signalling (UUS)

The actions of the MGCF at the ISUP side are described in [84]. The service shall be terminated at the MGCF and the call shall continue according to the basic call procedures.

7.4.22 Void

7.4.23 Void

7.5 Void

8 User Plane Interworking

8.1 Void

8.2 Interworking between IM CN Subsystem and TDM-based CS Network

It shall be possible for the IM CN subsystem to interwork with the TDM based CS networks (e.g., PSTN, ISDN). Figure 25 describes the user plane protocol stack to provide the particular interworking.

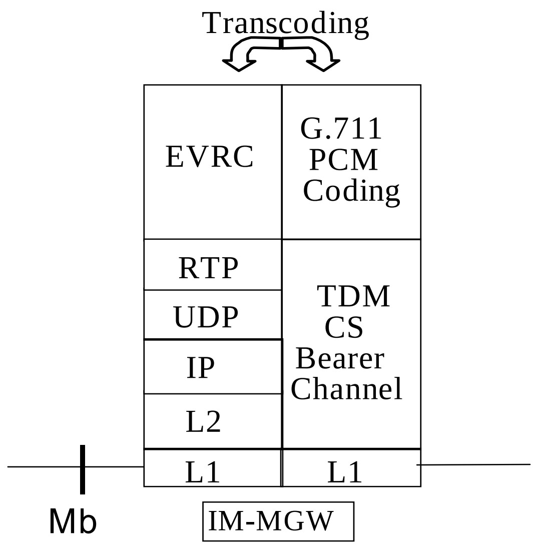


Figure 25 IM CN Subsystem to TDM-based CS network User Plane Protocol Stack

8.3 Transcoding Requirements

The IM CN subsystem supports the EVRC class codecs [20] as the native codec for basic voice services. For IM CN subsystem terminations, the IM MGW shall support the transport of EVRC over RTP according to [20].

It shall be possible for the IM CN subsystem to interwork with the CS networks (e.g., PSTN, ISDN) by supporting EVRC to G.711 transcoding (see [5]) in the IM-MGW. The IM-MGW may also perform transcoding between EVRC and other codec types supported by CS networks.

9 MGCF – IM-MGW Interaction

9.1 Overview

The MGCF shall control the functions of the IM-MGW, which are used to provide the connection between media streams of an IP based transport network and bearer channels from a CS network.

The MGCF shall interact with the IM-MGW across the Mn reference point. The MGCF shall terminate the signalling across the Mn interface towards the IM-MGW and the IM-MGW shall terminate the signalling from the MGCF.

The signalling interface across the Mn reference point shall be defined in accordance with [2].

The present specification describes Mn signalling procedures and their interaction with ISUP and SIP signalling in the control plane.

9.2 Mn Signalling Interactions

The following paragraphs describe the Mn interface procedures triggered by SIP signalling relayed in MGCF.

The SIP signalling occurring at the MGCF is described in [9].

All message sequence charts in this clause are examples.

9.2.1 Network Model

Figure 26 shows the network model, applicable to ISUP cases. The broken line represents the call control signalling. The dotted line represents the bearer control signalling (if applicable) and the user plane. The MGCF uses one context with two terminations in the IM-MGW. The termination T1 is used towards the IM CN subsystem entity and the bearer termination T2 is used for the bearer towards the succeeding CS network element.

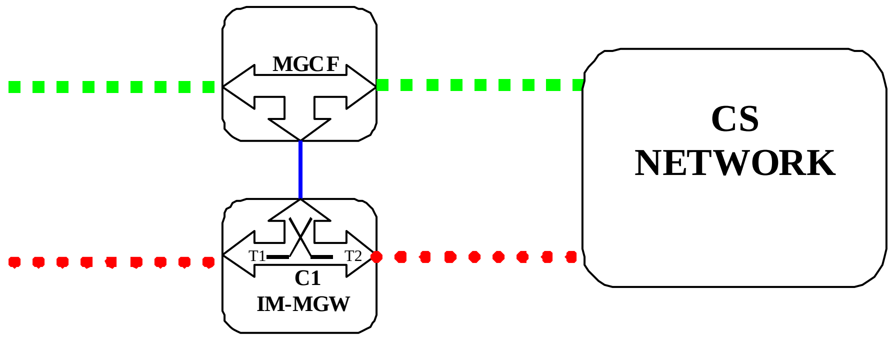


Figure 26 Network model

9.2.2 Basic IM CN Subsystem Originated Session

9.2.2.1 Void

9.2.2.2 Void

9.2.2.3 ISUP

9.2.2.3.1 IM-MGW Selection

The MGCF shall select an IM-MGW with circuits to the given destination in the CS domain before it performs the IM CN subsystem session establishment and before it sends the IAM (signal 8 in Figure 27).

9.2.2.3.2 IM CN Subsystem Side Termination Reservation

On receipt of an initial INVITE (signal 1 in Figure 27) the MGCF shall reserve the IMS connection (signal 3 and 4 in Figure 27). From the received SDP and local configuration data the MGCF shall:

- send the appropriate remote codec(s), the remote UDP port and the remote IP address to the IM-MGW. The remote UDP port and IP address refer to the destination of user plane data sent towards the IM CN subsystem. The remote codec(s) are the codec(s) the IM-MGW may select for user plane data sent towards the IM CN subsystem.
- indicate to the IM-MGW the appropriate local codec(s) and request a local IP address and UDP port. The local IP address and UDP port are used by the IM-MGW to receive user plane data from the IM CN subsystem. The local codec(s) are the codec(s) the IM-MGW may select to receive user plane data from the IM CN subsystem.
- If DTMF support together with speech support is required, the reserve value indicator shall be set to “true”.

The IM-MGW shall:

- reply to the MGCF with the selected local codec(s) and the selected remote codec(s) and the selected local UDP port and IP address;
- reserve resources for those codec(s).

The MCGF shall send selected local codec(s) and the selected remote codec and the selected local UDP port and IP address to the IMS in the Session Progress (signal 5 in Figure 27).

9.2.2.3.3 IM CN Subsystem Side Session Establishment

Dependent on what the MGCF receives in the PRACK message (signal 9 in Figure 27) the MGCF may change the IMS resources. If no SDP is received, or if the received SDP does not contain relevant changes compared to the previous SDP, the no action is taken. Otherwise the MGCF provides (c.f. signal 10) the IM-MGW:

- the appropriate remote codec(s), the remote UDP port and the remote IP address;
- optionally the appropriate local codec(s), UDP port and IP address;
- If DTMF support together with speech support is required, the reserve value indicator shall be set to “true”.

The IM-MGW shall:

- reply to the MGCF with the selected remote codec;
- reply to the MGCF with the selected local codec(s), if the MGCF supplied local codec(s);
- update the codec reservation and remote IP address and UDP port in accordance with the received information.

The MGCF shall include the selected codec(s) UDP port and IP address in 200 OK (PRACK) (signal 12 in Figure 27) sent back to the IMS.

9.2.2.3.4 CS Network Side Circuit Reservation

The MGCF shall request the IM-MGW to reserve a circuit. The MGCF sends the IAM to the succeeding node including the reserved circuit identity.

9.2.2.3.5 Through-Connection

During the reservation of the TDM Circuit and the IMS Connection Point procedures, the MGCF shall either request the IM-MGW to backward through-connect the termination, or the MGCF shall both-way through-connect the TDM termination already on this stage (signal 6 in Figure 27). During the reservation of the IMS connection, the MGCF shall request the IM-MGW to backward through-connect the IMS termination (signal 3 in Figure 27).

When the MGCF receives the ISUP:ANM answer indication, it shall request the IM-MGW to both-way through-connect the terminations (signal 21 in Figure 27), unless those terminations are already both-way through-connected.

9.2.2.3.6 Continuity Check

The MGCF may request a continuity check on the connection towards the CS network within the IAM message. In this case, the MGCF shall request the IM-MGW to generate a continuity check tone on the TDM termination. The IM-MGW shall then notify the MGCF of an incoming continuity check tone on the corresponding circuit. In addition to other conditions detailed in Section 7, the MGCF shall wait until receiving this notification before sending the COT. (Not depicted in Figure 27)

9.2.2.3.7 Codec Handling

The IM-MGW may include a speech transcoder based upon the speech coding information provided to each termination.

9.2.2.3.8 Voice Processing Function

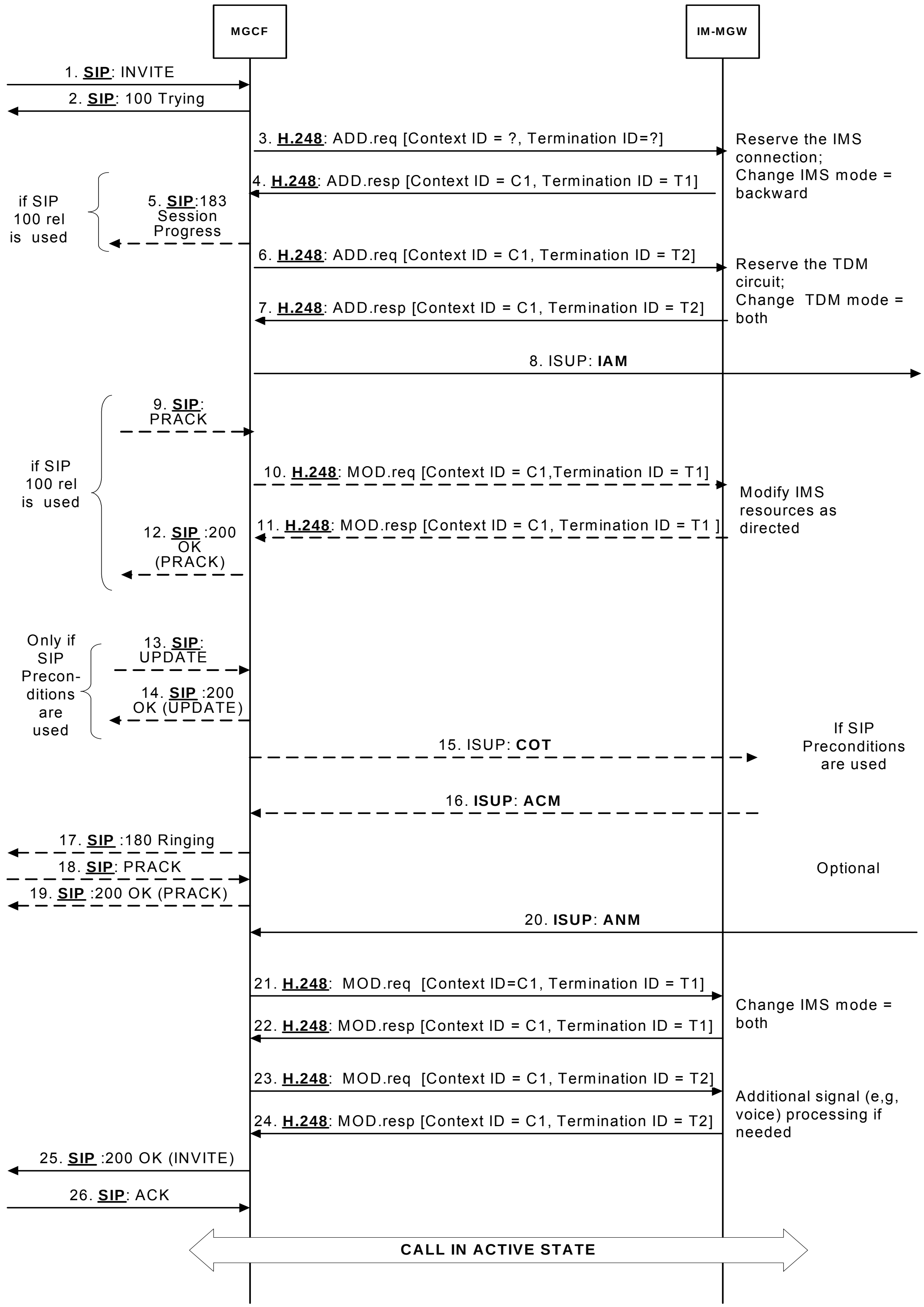
A voice processing function located on the IM-MGW may be used to achieve desired acoustic quality on the terminations. If the voice processing function is used, the MGCF shall request the activation of it in the termination towards the CS network (signal 23 in Figure 27).

9.2.2.3.9 Failure Handling in MGCF

If any procedure between the MGCF and the IM-MGW is not completed successfully session shall be released as described in clause 9.2.6.

9.2.2.3.10 Message Sequence Chart

Figure 27 shows the message sequence chart for the IM CN subsystem originating session. In the chart the MGCF requests the seizure of an IM CN subsystem side termination and a CS network side bearer termination. When the MGCF receives an answer indication, it requests the IM-MGW to both-way through-connect the terminations. The MGCF requests the possible activation of the voice processing functions for the bearer terminations. Dashed lines represent optional or conditional messages.



1
2 **Figure 27 Basic IM CN Subsystem Originating Session, ISUP (Message Sequence Chart)**

9.2.3 Basic CS Network Originated Session

9.2.3.1 Void

9.2.3.2 Void

9.2.3.3 ISUP

9.2.3.3.1 IM-MGW Selection

The MGCF selects the IM-MGW based on the received circuit identity in the IAM.

9.2.3.3.2 CS Network Side Circuit Reservation

The MGCF shall request the IM-MGW to reserve a circuit.

9.2.3.3.3 IM CN Subsystem Side Termination Reservation

The MGCF shall derive from configuration data one or several appropriate local codec(s) the IM-MGW may use to receive user plane data from the IM CN subsystem. At this time (c.f. signals 2 and 3 in Figure 28), the MGCF shall indicate the local codec(s) and request a local IP address and UDP port from the IM-MGW. The local IP address and UDP port are used by the IM-MGW to receive user plane data from the IM CN subsystem. If DTMF support together with speech support is required, or if the resources for multiple speech codecs shall be reserved at this stage, the reserve value indicator shall be set to "true".

The IM-MGW shall reply to the MGCF with the selected local codec(s) and the selected local IP address and UDP port.

The MGCF shall send this information in the INVITE (signal 6 in Figure 28) to the IM CN subsystem.

9.2.3.3.4 IM CN Subsystem Side Session Establishment

The MGCF shall provide configuration data (derived from SDP received in signal 8 in Figure 28 and local configuration data) using signals 9 and 10 or 22a and 22b in Figure 28, as detailed below:

- The MGCF shall indicate the remote IP address and UDP port, i.e., the destination IP address and UDP port for data sent in the user plane towards the IM CN subsystem.
- The MGCF shall indicate the remote codec(s), i.e., the speech codec(s) for data sent in the user plane towards the IM CN subsystem.
  - The MGCF may indicate the local codec(s) and the local IP address and UDP port. The MGCF shall indicate the local codec(s) if a change is required.
  - If DTMF support together with speech support is required, the reserve value indicator shall be set to "true".

The IM-MGW shall reply with the selected remote codec(s) and reserve resources for these codec(s). If local codec(s) were received, the IM-MGW shall also reply with the selected local codec(s) and reserve the corresponding resources.

If the selected local codec(s) differ from the codec(s) received in the SDP of signal 8 in Figure 28 (if any), the MGCF shall send the reserved speech codec(s), and the local IP address and UDP port in the PRACK (signal 11 in Figure 28) to the IMS.

If the selected local codec(s) differ from the codec(s) received in the SDP of signal 22 in Figure 28 (if any), the MGCF shall send the local reserved codec(s), and the local IP address and UDP port in an re-INVITE or UPDATE (not depicted in Figure 28) to the IMS.

9.2.3.3.5 Called Party Alerting

The MGCF shall request the IM-MGW to provide an awaiting answer indication (ringing tone) to the calling party (signals 19 and 20 in Figure 28) , when the first of the following conditions is satisfied:

- the MGCF receives the first 180 Ringing message;

- Timer T i/w_2 expires.

9.2.3.3.6 Called Party Answer

When the MGCF receives a 200 OK message (signal 22 in Figure 28), it shall request the IM-MGW to stop providing the ringing tone to the calling party (signals 23 and 24 in Figure 28).

9.2.3.3.7 Through-Connection

The MGCF shall either request the IM-MGW to backward through-connect the TDM termination, or the MGCF shall both-way through-connect the TDM termination already on this stage (signals 2 and 3 in Figure 28). The MGCF shall request the IM-MGW to backward through-connect the IMS termination (signals 4 and 5 in Figure 28).

When the MGCF receives the SIP 200 OK(INVITE) message, it shall request the IM-MGW to both-way through-connect the terminations (signals 25 and 26 in Figure 28), unless those terminations are already both-way through-connected.

9.2.3.3.8 Continuity Check

If a continuity check on the connection towards the CS network is requested in the IAM message, the MGCF shall request loop-back of a received continuity check tone on the TDM circuit. Upon reception of the COT message, the MGCF shall request the removal of the loop-back. (Not depicted in Figure 28)

9.2.3.3.9 Codec Handling

The IM-MGW may include a speech transcoder based upon the speech coding information provided to each termination.

9.2.3.3.10 Voice Processing Function

A voice processing function located on the IM-MGW may be used to achieve desired acoustic quality on the terminations. If the voice processing function is used, the MGCF shall request the activation of it in the termination towards the CS network (signal 23 in Figure 28).

9.2.3.3.11 Failure Handling in MGCF

If any procedure between the MGCF and the IM-MGW is not completed successfully, the session shall be released as described in clause 9.2.6.

9.2.3.3.12 Message Sequence Chart

Figure 28 shows the message sequence chart for the CS network originating Session with ISUP. In the chart the MGCF requests seizure of the IM CN subsystem side termination and CS network side bearer termination. When the MGCF receives an answer indication, it requests the IM-MGW to both-way through-connect the terminations. The MGCF may request the possible activation of the voice processing functions for the terminations. Dashed lines represent optional or conditional messages.

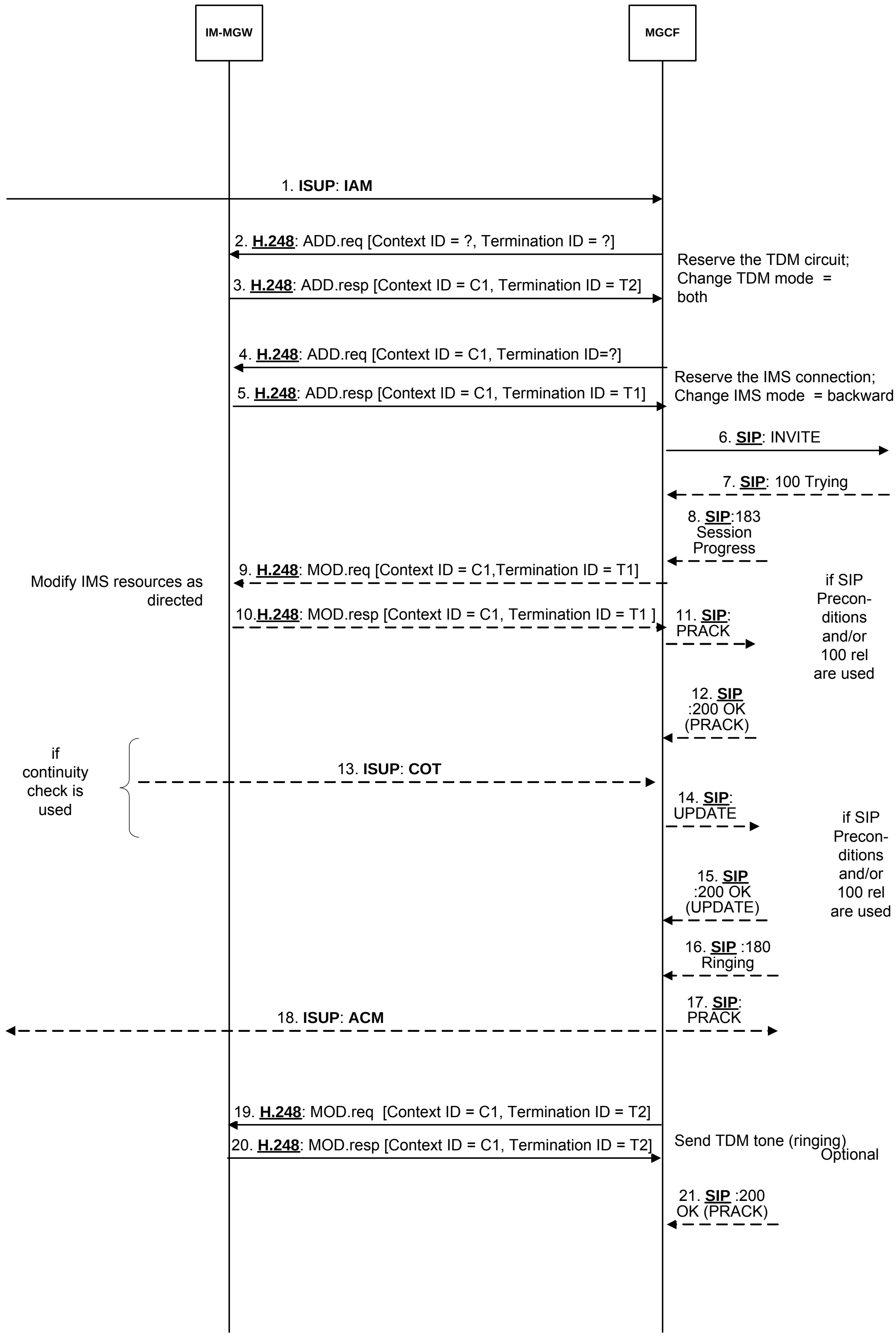


Figure 28/1 Basic CS Network Originating Session – ISUP (Message Sequence Chart)

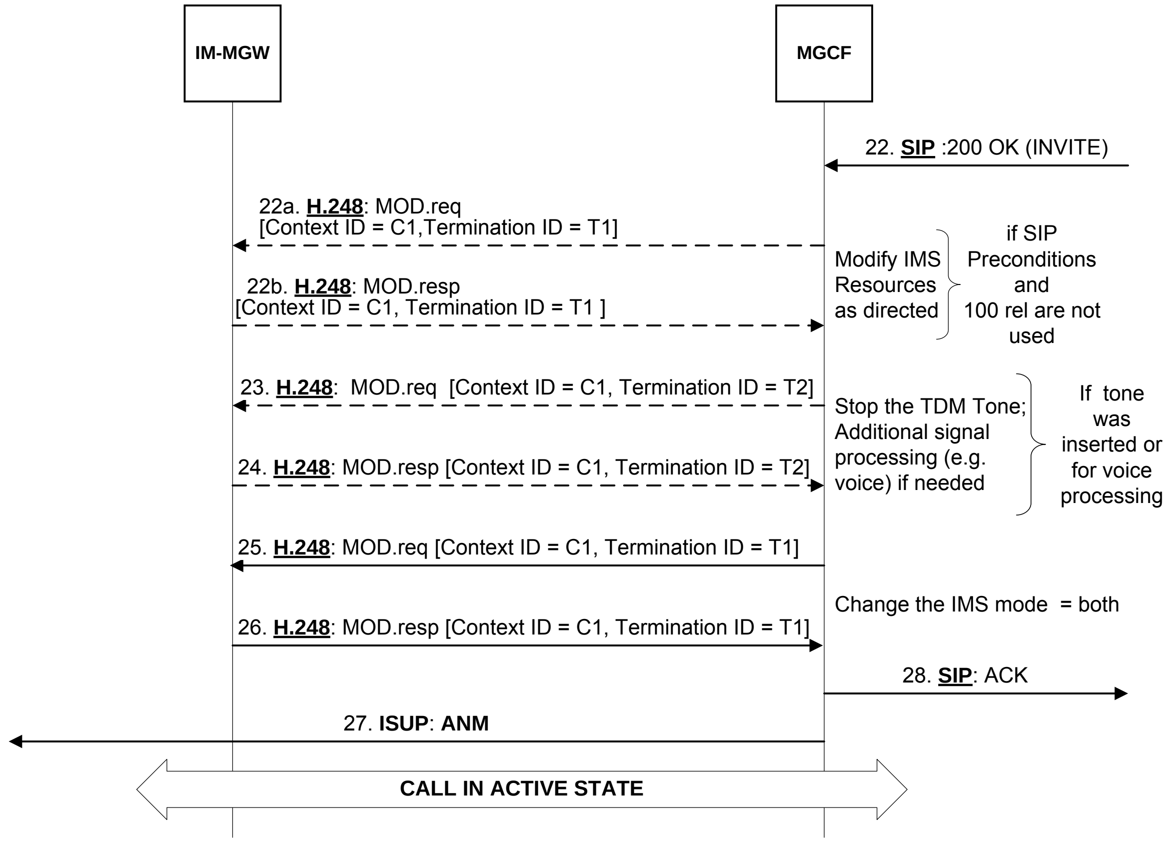


Figure 28/2 Basic CS Network Originating Session – ISUP (Message Sequence Chart)

9.2.3.4 Handling of Forking

The procedures described in clauses 9.2.3.1 to 9.2.3.3 shall be applied with the following additions.

9.2.3.4.1 Detection of Forking

According to SIP procedures, the O-MGCF inspects the tags in the “to” SIP header fields of provisional and final responses to identify the SIP dialogue the response belongs to. If responses belonging to different dialogues are received (signals 8 and 13 in Figure 29) , the INVITE request (signal 6 in Figure 29) has been forked.

9.2.3.4.2 IM CN Subsystem Side Session Establishment

If SDP is received in a provisional response and more than one SIP dialogue exists (signal 13 in Figure 29), the MGCF may either refrain from reconfiguring the IM-MGW, or it may respond (signals 14 and 15 in Figure 29) as detailed below:

- The MGCF may compare the selected local codecs of the different dialogues (which the MGCF selects due to the received SDP answer and local configuration data). If different local codecs are selected for the different dialogues, the MGCF may include all these codecs in the “local IMS resources”, and set the “reserve value” to indicate that resources for all these codecs shall be reserved. Alternatively, the MGCF may only include the codecs received in the last SDP in the “local IMS resources”.

- The MGCF may update the “remote IMS resources” with the information received in the latest SDP. The MGCF should provide the remote IP address and UDP port, and the remote codec selected from the received SDP and local configuration data.

NOTE The behaviour in the second bullet is beneficial if forking is applied in a sequential manner.

9.2.3.4.3 IM CN Subsystem Side Session Establishment Completion

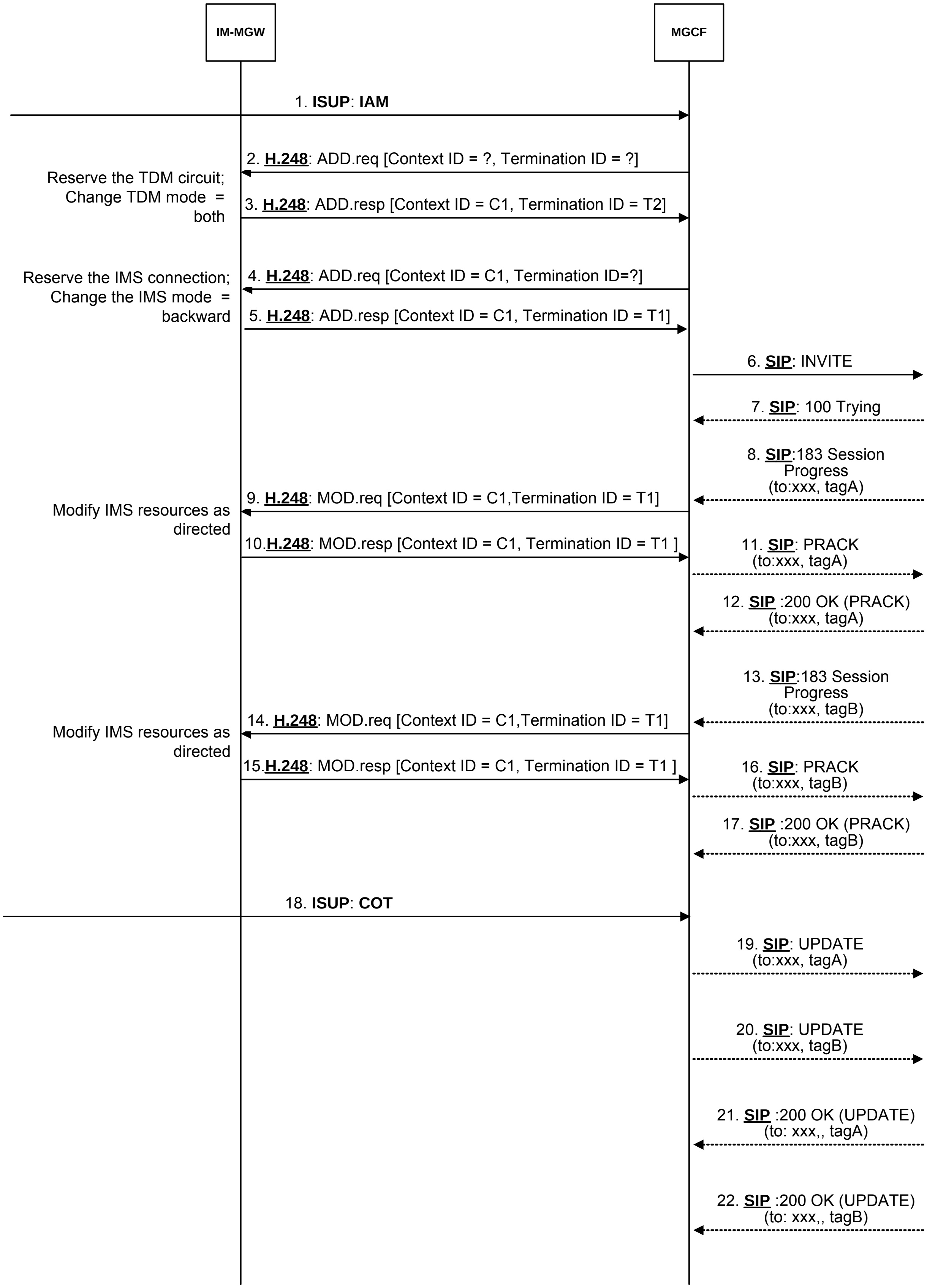
Upon reception of the first final 2xx response (signal 32 in Figure 29), the MGCF shall respond (signals 35 and 36 in Figure 29) as detailed below unless the IM-MGW is already configured accordingly:

- If the remote IMS resources configured at the IM-MGW do not match the remote resources selected for the established dialogue of the final response, the MGCF shall provide the remote IP address and UDP port from the latest received SDP of this established dialogue, and the remote codec selected from the latest received SDP of this established dialogue and local configuration data within the “remote IMS resources”.

- If the local IMS resources configured at the IM-MGW contain more codecs than selected for the established dialogue of the final response, the MGCF should update the “local IMS resources” with the selected local codec derived from the latest SDP of this established dialogue and local configuration data. The “reserve value” may be cleared unless it is required for DTMF.

9.2.3.4.4 Message Sequence Chart

Figure 29 shows an example message sequence chart for an CS network originating Session Setup with ISUP, where forking occurs.



1

2 **Figure 29/1 CS Network Originating Session with Forking – ISUP (Message Sequence Chart)**

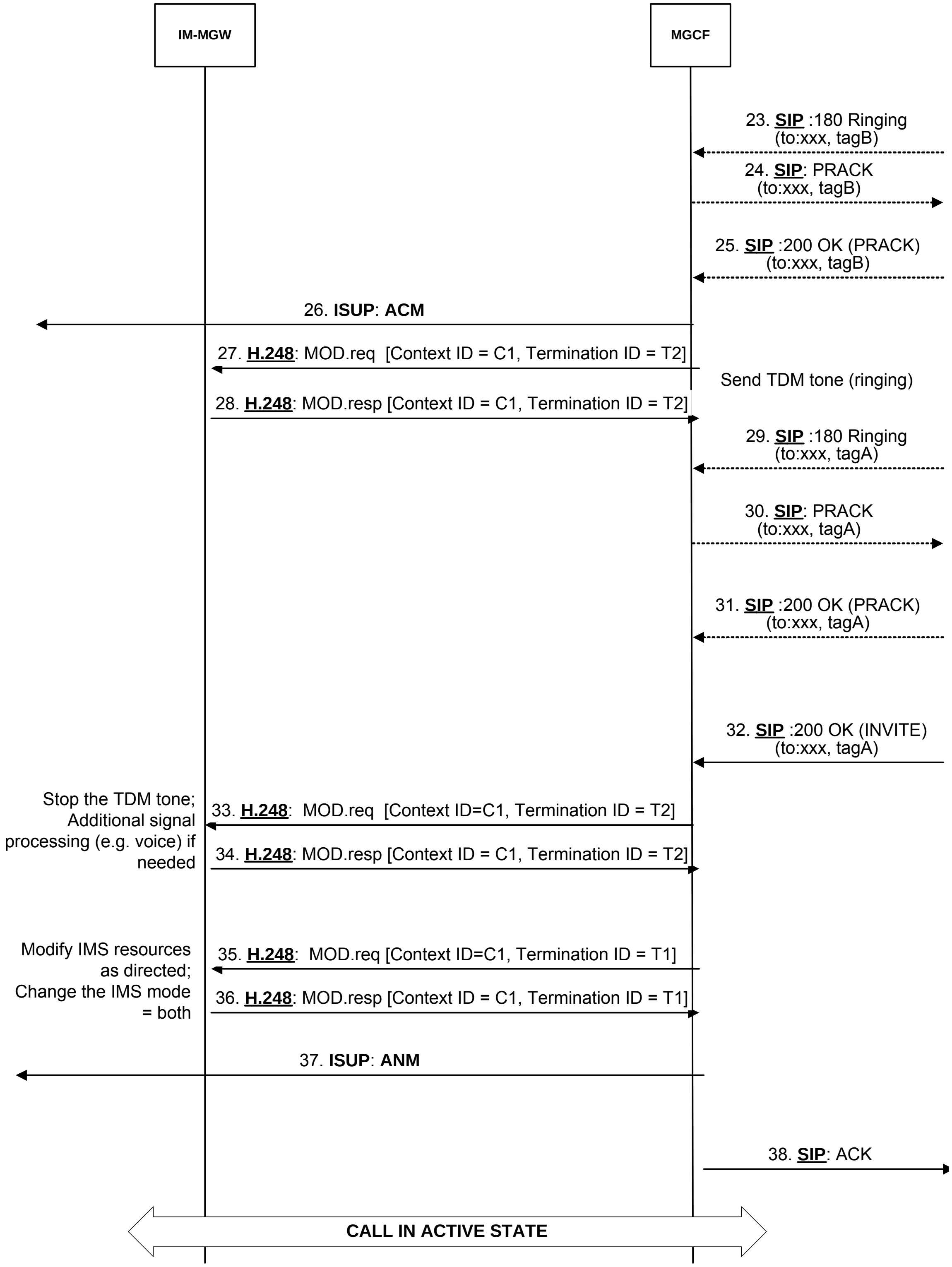


Figure 29/2 CS Network Originating Session with Forking, ISUP (Message Sequence Chart continued)

9.2.4 Session Release Initiated from IM CN Subsystem Side

9.2.4.1 Void

9.2.4.2 ISUP

9.2.4.2.1 Session Release in the IM CN Subsystem Side

When the MGCF has received a BYE message from the IM CN subsystem side, the MGCF shall release resources in the IM-MGW serving the relevant Mb interface connection (signals 4 and 5 in Figure 30). After receiving the BYE message, the MGCF shall also send a 200 OK [BYE] message towards the IM CN subsystem (signal 2 in Figure 30).

9.2.4.2.2 Session Release in the CS Network Side

When the MGCF has received a BYE message from the IM CN subsystem side, the MGCF shall send a REL message to the succeeding node (signal 3 in Figure 30). After sending the REL message, the MGCF shall expect a RLC message (signal 8 in Figure 30) from the succeeding node. The MGCF shall also release the resources for the CS network side in the IM-MGW. If any resources were seized in the IM-MGW, the MGCF shall indicate to the IM-MGW (signals 6 to 7 in Figure 30) that the CS network side bearer termination can be released.

9.2.4.2.3 Message Sequence Chart

Figure 30 shows the message sequence chart for the session release initiated from the IM CN subsystem side.

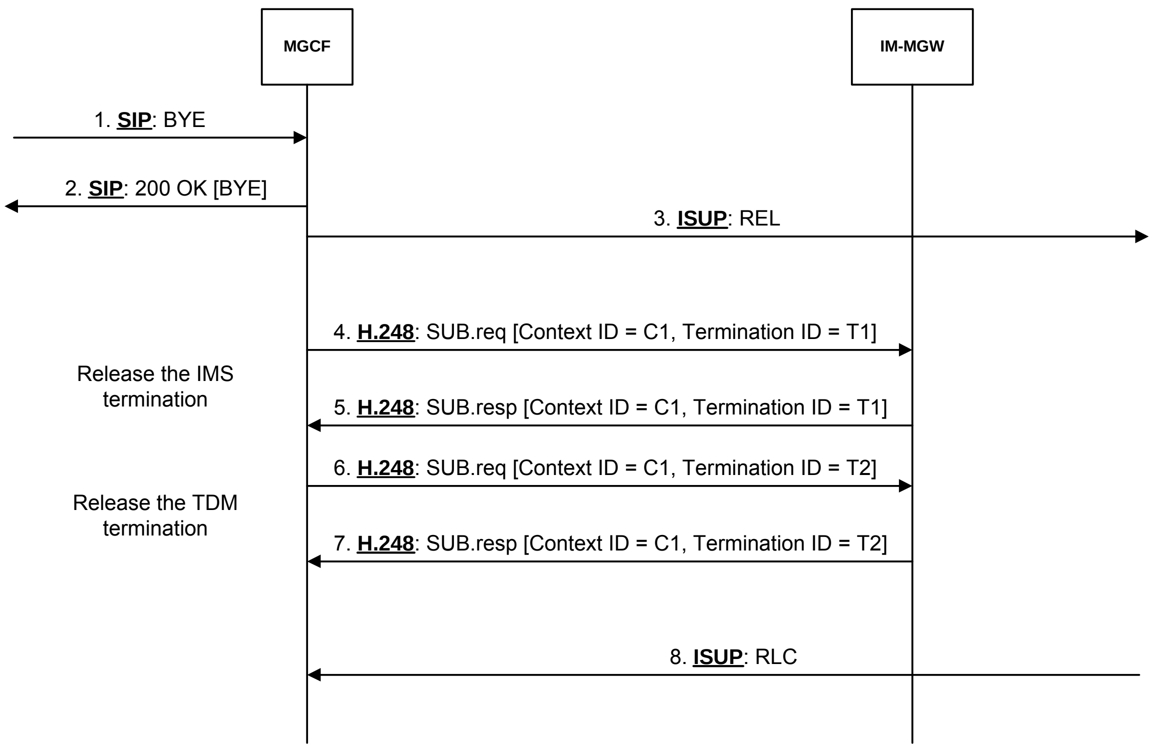


Figure 30 Session Release from IM CN Subsystem Side for ISUP (Message Sequence Chart)

9.2.5 Session Release Initiated from CS Network Side

9.2.5.1 Void

9.2.5.2 ISUP

9.2.5.2.1 Session Release in the CS Network Side

When the MGCF receives a REL message from the preceding node (signal 1 in Figure 31), the MGCF shall release resources for the CS network side in the IM-MGW. If any resources were seized in the IM-MGW, the MGCF shall indicate to the IM-MGW that the CS network side bearer termination can be released (signal 3 to 4 in Figure 31). After completion of resource release, the MGCF shall send a RLC message towards the preceding node.

9.2.5.2.2 Session Release in the IM CN Subsystem Side

When the MGCF receives a REL message from the preceding node (signal 1 in Figure 31), the MGCF shall send a BYE message to the IM CN subsystem (signal 2 in Figure 31) and the MGCF shall release the resources in the IM-MGW serving the relevant Mb interface connection (signal 5 to 6 in Figure 31). The MGCF shall also expect to receive a 200 OK [BYE] message from the IM CN subsystem side (signal 8 in Figure 31).

9.2.5.2.3 Message Sequence Chart

Figure 31 shows the message sequence chart for the session release initiated from the CS network side.

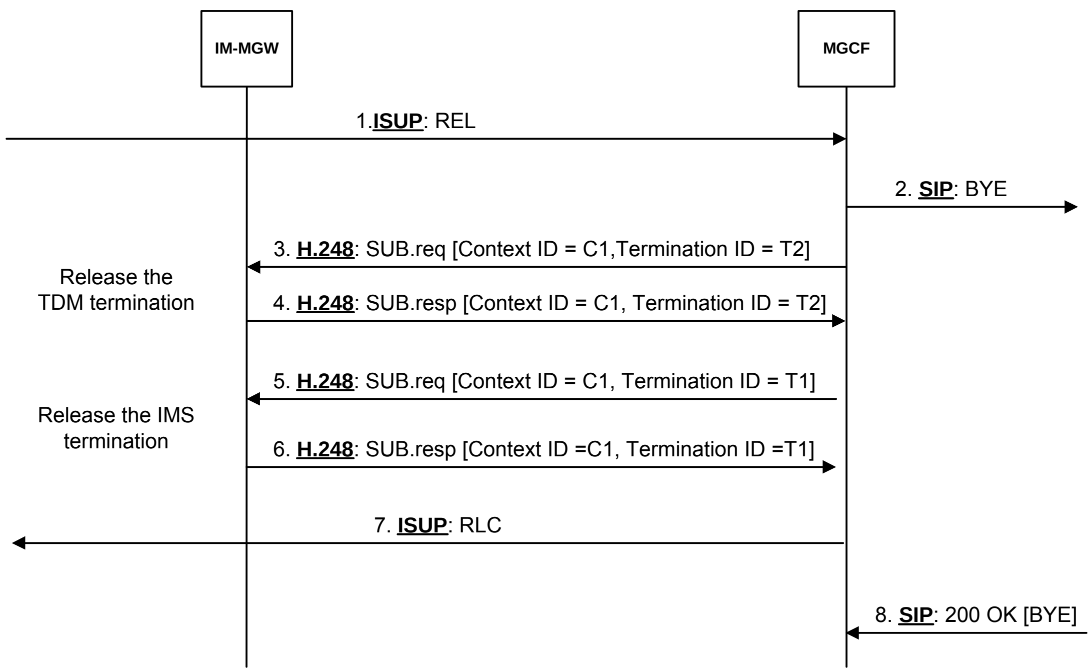


Figure 31 Session Release from CS Network Side for ISUP (Message Sequence Chart)

9.2.6 Session Release Initiated by MGCF

9.2.6.1 Void

9.2.6.2 ISUP

9.2.6.2.1 Session Release in the CS Network Side

The MGCF shall send a REL message to the succeeding node on the CS network side (signal 2 in Figure 32) and the MGCF shall release the resources for the CS network side in the IM-MGW. If any resources were seized in the IM-MGW, the MGCF shall indicate to the IM-MGW that the CS network side termination shall be released (signal 5 to 6 in Figure 32). The MGCF shall also expect to receive a RLC message from the succeeding node on the CS network side (signal 7 in Figure 32).

9.2.6.2.2 Session Release in the IM CN Subsystem Side

The MGCF shall send a BYE message to the IM CN subsystem side (signal 1 in Figure 32) and the MGCF shall release the resources in the IM-MGW serving the relevant Mb interface connection (signal 5 to 6 in Figure 32). The MGCF shall also expect to receive a 200 OK [BYE] message from the IM CN subsystem side (signal 8 in Figure 32).

9.2.6.2.3 Message Sequence Chart

Figure 32 shows the message sequence chart for the session release initiated by the MGCF.

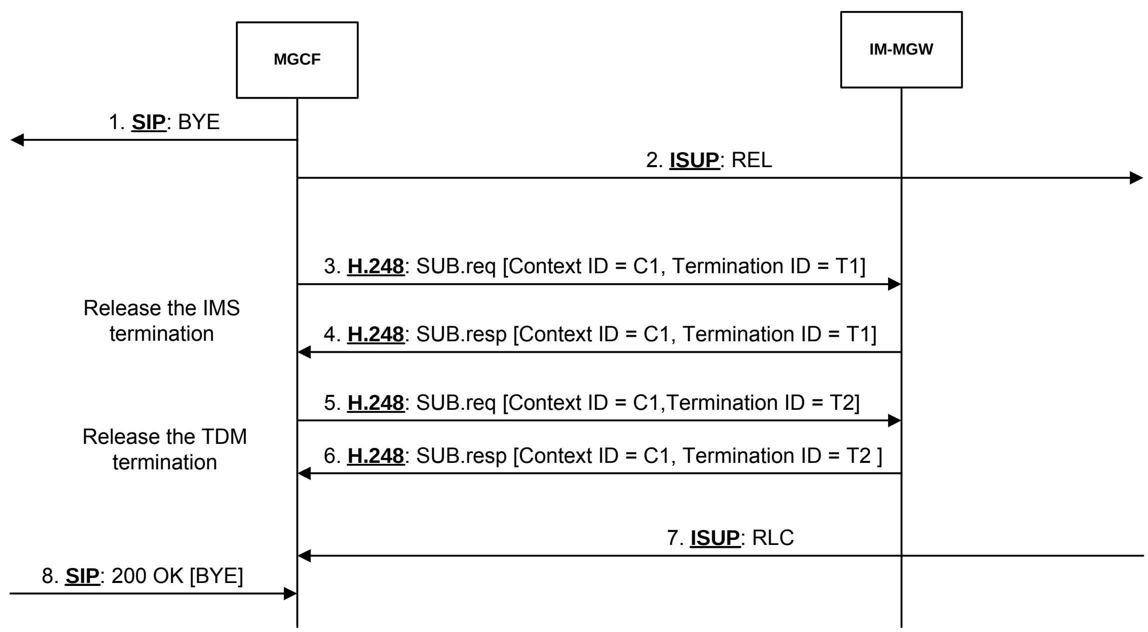


Figure 32 Session Release Initiated by MGCF for ISUP (Message Sequence Chart)

9.2.7 Session Release Initiated by IM-MGW

9.2.7.1 Void

9.2.7.2 ISUP

9.2.7.2.1 Session Release in the CS Network Side

Upon receiving from the IM-MGW an indication of an immediate release, the MGCF shall send a REL message to the succeeding node (signal 3 in Figure 33). The indication of immediate release includes a:

a. a“termination out of service” (signals 1a and 2a in Figure 33);

b. a “bearer released” (signals 1b and 2b in Figure 33); or

c. a “MGW out of service” (not shown in Figure 33) consisting of a H248 ServiceChangeMethod=“Forced”.

Upon receiving from the IM-MGW an “immediate release” (see “a” or “b” immediately above), the MGCF shall also release the resources for the corresponding CS network side termination(s) in the IM-MGW. If any resources were seized in the IM-MGW, the MGCF shall indicate to the IM-MGW that the CS network side bearer termination can be removed (signals 7 and 8 in Figure 33). The MGCF also expects to receive a RLC message on the CS network side (signal 9 in Figure 33) before the circuit is reselectable.

9.2.7.2.2 Session Release in the IM CN Subsystem Side

Upon receiving from the IM-MGW a “termination out of service” indication (see “a” above) or a “MGW out of service” indication (see “b” above), an immediate release on the CS termination is requested and the MGCF shall send a BYE/CANCEL message to the IM CN subsystem side (signal 4 in Figure 33). Upon receiving from the IM-MGW a “termination out of service” (see “a” above) the MGCF shall also release the resources in the IM-MGW for the corresponding terminations towards the IM CN subsystem (signals 5 and 6 in Figure 33). The MGCF also expects to receive a 200 OK [BYE] message from the IM CN subsystem side (signal 10 in Figure 33).

9.2.7.2.3 Message Sequence Chart

Figure 33 shows the message sequence chart for the session release initiated by the IM-MGW.

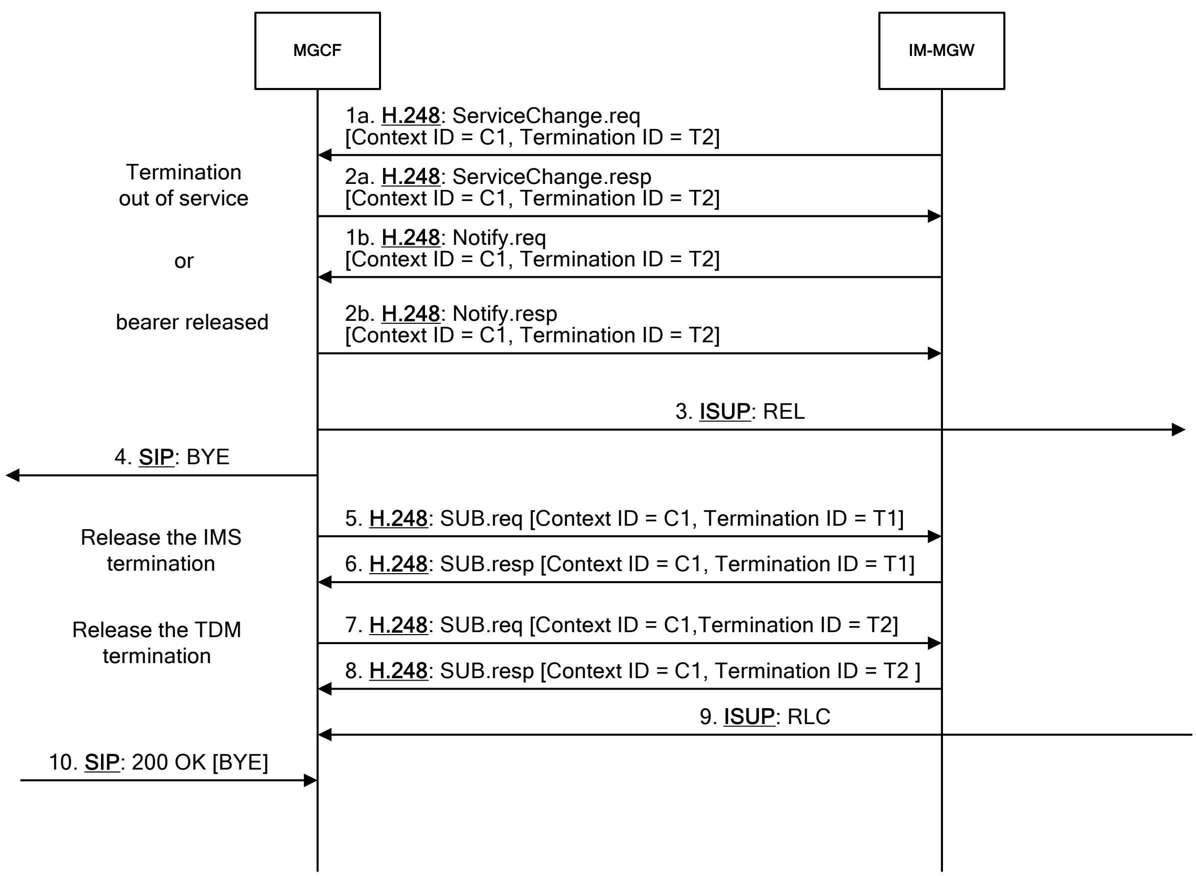


Figure 33 Session Release Initiated by the IM-MGW for ISUP (Message Sequence Chart)

9.2.8 Handling of RTP Telephone Events

DTMF digits, telephony tones and signals (telephone events) can be transferred using different mechanisms. For the IM CN Subsystem, [9] defines the usage of the RTP payload format which is defined for DTMF Digits, Telephony Tones and Telephony Signals in [34]. If ISUP signalling is used the DTMF tones are sent inband. The following paragraphs describe the Mn interface procedures to transfer DTMF between RTP format defined in [34] and the CS CN.

Before the actual usage of the telephony signals can occur the sending/receiving of telephone events need to be agreed with the SDP offer-answer mechanism defined in [36]. The outcome of the negotiation can be e.g., that no telephone events are sent in RTP payload, telephone events are sent only in one direction or in both directions. If the outcome of the negotiation is that RTP payload telephone-events are sent in both directions, the IM-MGW may nevertheless be configured to interwork only mobile originated telephone-events.

When the offer-answer mechanism based session parameters negotiation results in an agreement that telephone events are sent in the RTP payload and the needed preconditions are fulfilled, telephone events can be sent in RTP payload. This negotiation can be done at call control signalling phase or during an ongoing call.

If the MGCF and IM-MGW support the reception and/or transmission of the RTP MIME type “telephone event” (as defined in [34]) with the IMS, the following applies:

- For CS Network Originating Sessions, the MGCF shall include the MIME type “telephone events” with default events in the first SDP offer. After the usage of telephone events is agreed in the subsequent offer-answer parameter exchanges and the needed preconditions defined in [37] are fulfilled, telephone events can be sent as RTP payload.

- In case of IM CN Subsystem Originating Sessions, the MGCF shall accept the MIME type “telephone events” with default events in any SDP answer when it received such an offer.

9.2.8.1 Void

9.2.8.2 Sending and Receiving DTMF Digits Inband to/from CS CN (ISUP)

For the IM CN subsystem terminated session, the MGCF shall configure the IMS resources as described in Clause 9.2.3. For the IM CN subsystem originating session , the MGCF shall reserve the IMS connections and configure remote resources as described in Clause 9.2.2. If DTMF is supported, the MGCF shall include “telephone event” along with the selected speech codecs when requesting the MGW to detect incoming telephone events and transform them into speech signals on the CS side. When receiving this configuration, the MGW may in addition optionally detect incoming telephone events received inband from the CS CN network and transform them into telephone events on the IMS side. The same termination shall be used to receive and transmit DTMF and speech of the same call.

Figure 34 shows the message sequence chart to configure the IM-MGW to receive DTMF detection on the IMS side and transfer the DTMF inband on the CS side. When receiving this configuration, the IM-MGW may in addition optionally detect DTMF inband on the CS side and transmit DTMF on the IMS side.

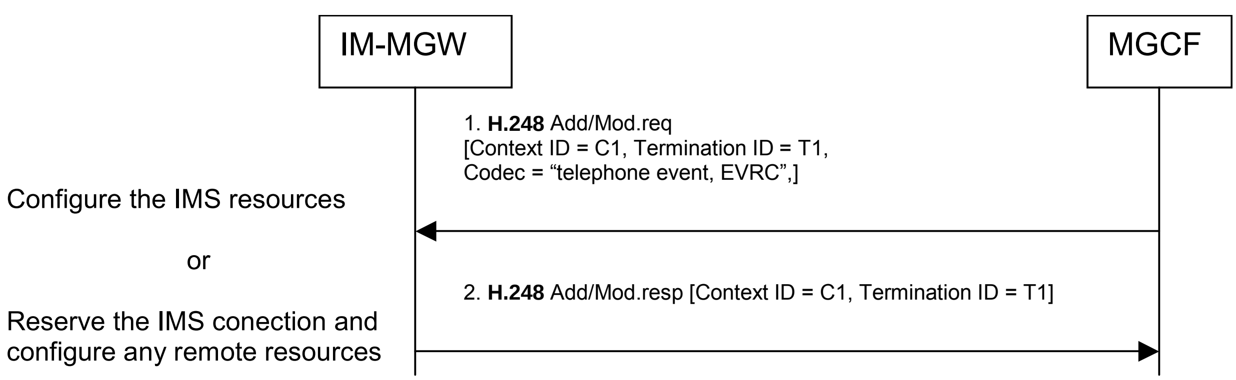


Figure 34 Activation of Processing of DTMF Digits Received in RTP for Sending the Digits inband to CS CN (Message Sequence Chart)

9.2.9 Session Hold Initiated from IM CN Subsystem

The network model in the clause 9.2.1 shall apply here.

9.2.9.1 Hold Request

When the IMS network makes a hold request by sending an UPDATE or re-INVITE message (signal 1 of Figure 35), the MGCF shall request the IM-MGW to suspend sending media towards the IMS side by changing the through-connection of the IM CN subsystem side termination to 'not through-connected' (signal 2 of Figure 35). If the IMS side provides modified SDP RR or RS bandwidth modifiers, as specified in [59], within the hold request, the MGCF shall configure the IMS resources to forward this information to the IM-MGW (not depicted in Figure 35, 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 35). Simultaneously a SIP message acknowledging the Hold request is sent to the IMS side (signal 7 of Figure 35, acknowledged by signal 7.a if the INVITE method is used). Announcements may be applied to the party on hold, depending on the held party's status (for ISUP, signal 5 in Figure 35). The hold operation shall not block RTCP flows.

9.2.9.2 Resume Request

When the IMS network makes a request to retrieve the session on hold by sending an UPDATE or re-INVITE message (signal 8 of Figure 35), 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 35). If the IMS side provides modified SDP RR or RS bandwidth modifiers, as specified in [59], within the retrieve request, the MGCF shall configure the IMS resources to forward this information to the IM-MGW (not depicted in Figure 35, but may be combined with signal 11). Possible announcements to the party on hold shall be stopped (for ISUP, signal 9 in Figure 35). 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 35).

9.2.9.3 Message Sequence Chart

Figure 35 shows the message sequence chart for the call hold and hold-release procedures.

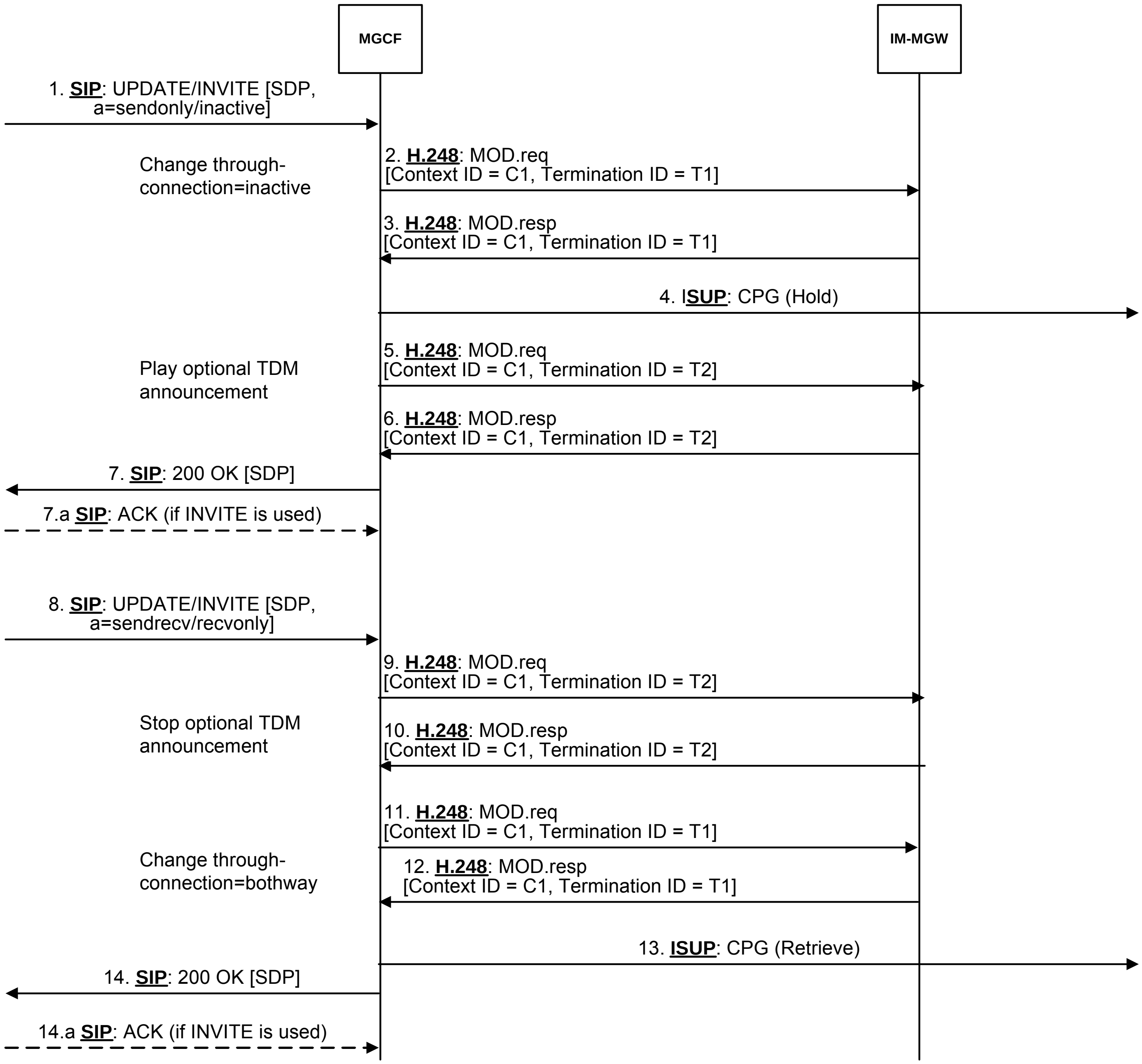


Figure 35 Session Hold from IM CN Subsystem

9.2.10 Session Hold Initiated from CS Network

When an MGCF receives a CPG message with a ‘remote hold’ Generic notification indicator (signal 1 of Figure 36), the MGCF forwards the hold request by sending an UPDATE or re-INVITE message containing SDP with “sendonly” or “inactive” media (signal 4 of Figure 36).

When an MGCF receives a CPG message with a ’remote hold release’ Generic notification indicator (signal 6 of Figure 36), the MGCF forwards the resume request by sending an UPDATE or re-INVITE message containing SDP with “sendrecv” or “recvonly” media (signal 9 of Figure 36).

If the MGCF receives a CPG with ‘remote hold’ or ‘remote hold release’ before answer, it shall forward the request using an UPDATE message. If the MGCF receives a CPG with ‘remote hold’ or ‘remote hold release’ after answer, it should forward the request using re-INVITE but may use UPDATE.

If link aliveness information is required at the IM-MGW while the media are on hold, the MGCF should provide to the modified SDP RR and RS bandwidth modifiers specified in [59] within the SDP offers in the UPDATE or re-INVITE messages holding and retrieving the media to temporarily enable RTCP while the media are on hold. If no link aliveness information is required at the IM-MGW, the 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 or re-INVITE messages. If the MGCF provides modified SDP RR and RS bandwidth modifiers in the UPDATE or re-INVITE messages, the MGCF shall also provide modified SDP RR and RS bandwidths to the IM-MGW using the "Configure IMS Resources" procedures (signals 2-3 and 7-8 of Figure 36).

9.2.10.1 Message Sequence Chart

Figure 36 shows the message sequence chart for the call hold and hold-release procedures.

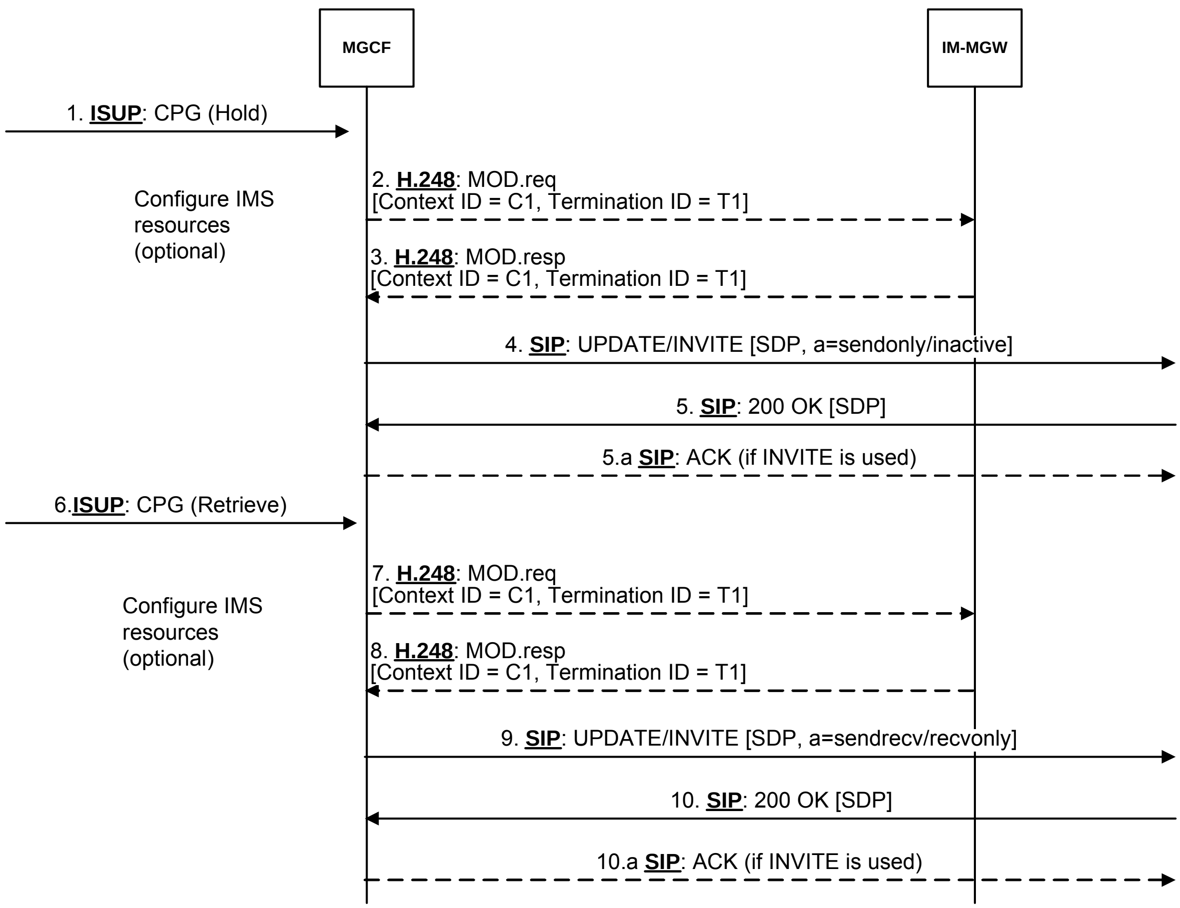


Figure 36 Session Hold from CS Network