

# ETSI TS 129 163 V14.3.0 (2017-04)



**Digital cellular telecommunications system (Phase 2+) (GSM);  
Universal Mobile Telecommunications System (UMTS);  
LTE;  
Interworking between the IP Multimedia (IM) Core Network (CN)  
subsystem and Circuit Switched (CS) networks  
(3GPP TS 29.163 version 14.3.0 Release 14)**



---

Reference

RTS/TSGC-0329163ve30

---

Keywords

GSM,LTE,UMTS

**ETSI**

650 Route des Lucioles  
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C  
Association à but non lucratif enregistrée à la  
Sous-Préfecture de Grasse (06) N° 7803/88

---

**Important notice**

The present document can be downloaded from:  
<http://www.etsi.org/standards-search>

The present document may be made available in electronic versions and/or in print. The content of any electronic and/or print versions of the present document shall not be modified without the prior written authorization of ETSI. In case of any existing or perceived difference in contents between such versions and/or in print, the only prevailing document is the print of the Portable Document Format (PDF) version kept on a specific network drive within ETSI Secretariat.

Users of the present document should be aware that the document may be subject to revision or change of status. Information on the current status of this and other ETSI documents is available at  
<https://portal.etsi.org/TB/ETSIDeliverableStatus.aspx>

If you find errors in the present document, please send your comment to one of the following services:  
<https://portal.etsi.org/People/CommiteeSupportStaff.aspx>

---

**Copyright Notification**

No part may be reproduced or utilized in any form or by any means, electronic or mechanical, including photocopying and microfilm except as authorized by written permission of ETSI.

The content of the PDF version shall not be modified without the written authorization of ETSI.  
The copyright and the foregoing restriction extend to reproduction in all media.

© European Telecommunications Standards Institute 2017.  
All rights reserved.

**DECT™**, **PLUGTESTS™**, **UMTS™** and the ETSI logo are Trade Marks of ETSI registered for the benefit of its Members.  
**3GPP™** and **LTE™** are Trade Marks of ETSI registered for the benefit of its Members and of the 3GPP Organizational Partners.  
**GSM®** and the GSM logo are Trade Marks registered and owned by the GSM Association.

---

## Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to ETSI. The information pertaining to these essential IPRs, if any, is publicly available for **ETSI members and non-members**, and can be found in ETSI SR 000 314: *"Intellectual Property Rights (IPRs); Essential, or potentially Essential, IPRs notified to ETSI in respect of ETSI standards"*, which is available from the ETSI Secretariat. Latest updates are available on the ETSI Web server (<https://ipr.etsi.org>).

Pursuant to the ETSI IPR Policy, no investigation, including IPR searches, has been carried out by ETSI. No guarantee can be given as to the existence of other IPRs not referenced in ETSI SR 000 314 (or the updates on the ETSI Web server) which are, or may be, or may become, essential to the present document.

---

## Foreword

This Technical Specification (TS) has been produced by ETSI 3rd Generation Partnership Project (3GPP).

The present document may refer to technical specifications or reports using their 3GPP identities, UMTS identities or GSM identities. These should be interpreted as being references to the corresponding ETSI deliverables.

The cross reference between GSM, UMTS, 3GPP and ETSI identities can be found under <http://webapp.etsi.org/key/queryform.asp>.

---

## Modal verbs terminology

In the present document "**shall**", "**shall not**", "**should**", "**should not**", "**may**", "**need not**", "**will**", "**will not**", "**can**" and "**cannot**" are to be interpreted as described in clause 3.2 of the [ETSI Drafting Rules](#) (Verbal forms for the expression of provisions).

"**must**" and "**must not**" are **NOT** allowed in ETSI deliverables except when used in direct citation.

# Contents

Intellectual Property Rights .....	2
Foreword.....	2
Modal verbs terminology.....	2
Foreword.....	17
1 Scope .....	18
2 References .....	18
3 Definitions and abbreviations.....	24
3.1 Definitions .....	24
3.2 Abbreviations .....	25
4 General .....	27
4.1 General interworking overview .....	27
5 Network characteristics .....	27
5.1 Key characteristics of ISUP/BICC based CS networks .....	27
5.2 Key characteristics of IM CN subsystem .....	27
6 Interworking with CS networks .....	27
6.1 Interworking reference model .....	27
6.1.1 Interworking reference points and interfaces .....	28
6.1.2 Interworking functional entities .....	28
6.1.2.1 Signalling Gateway Function (SGW) .....	28
6.1.2.2 Media Gateway Control Function (MGCF) .....	28
6.1.2.3 IP Multimedia - Media Gateway Function (IM-MGW) .....	28
6.2 Control plane interworking model.....	29
6.3 User plane interworking model .....	29
7 Control plane interworking .....	29
7.1 General .....	29
7.2 Interworking between CS networks supporting ISUP and the IM CN subsystem .....	30
7.2.1 Services performed by network entities in the control plane .....	30
7.2.1.1 Services performed by the SS7 signalling function .....	30
7.2.1.2 Services of the SGW .....	30
7.2.1.3 Services of the MGCF.....	31
7.2.1.4 Services of the SIP signalling function .....	31
7.2.2 Signalling interactions between network entities in the control plane.....	31
7.2.2.1 Signalling between the SS7 signalling function and MGCF .....	31
7.2.2.1.1 Signalling from MGCF to SS7 signalling function .....	31
7.2.2.1.2 Signalling from SS7 signalling function to MGCF .....	31
7.2.2.1.3 Services offered by SCTP and M3UA.....	31
7.2.2.1.3.1 Services offered by SCTP.....	31
7.2.2.1.3.2 Services offered by M3UA .....	31
7.2.2.2 Signalling between the MGCF and SIP signalling function.....	32
7.2.3 SIP-ISUP protocol interworking.....	32
7.2.3.1 Incoming call interworking from SIP to ISUP at I-MGCF .....	32
7.2.3.1.1 Sending of IAM.....	32
7.2.3.1.2 Coding of the IAM .....	33
7.2.3.1.2.0 General.....	33
7.2.3.1.2.1 Called party number.....	33
7.2.3.1.2.2 Nature of connection indicators .....	34
7.2.3.1.2.3 Forward call indicators .....	35
7.2.3.1.2.4 Calling party's category .....	36
7.2.3.1.2.4A Originating Line Information.....	36
7.2.3.1.2.5 Transmission medium requirement.....	36
7.2.3.1.2.5a Transmission medium requirement prime and USI prime (optional) .....	38
7.2.3.1.2.6 Calling party number .....	40

7.2.3.1.2.7	Generic number.....	45
7.2.3.1.2.8	User service information.....	45
7.2.3.1.2.9	Hop Counter (National option) .....	45
7.2.3.1.2.10	Progress Indicator .....	46
7.2.3.1.2.11	Location Number .....	46
7.2.3.1.2.12	Support of ICS call .....	48
7.2.3.1.2.13	UID capability indicators (National option) .....	48
7.2.3.1.2.14	Called IN number and original called IN number (optional) .....	49
7.2.3.1.2A	Coding of the IAM when Number Portability is supported.....	49
7.2.3.1.2A.1	Coding of the IAM when a Number Portability Routing Number is available.....	49
7.2.3.1.2A.1.1	Separate Directory Number Addressing Method .....	50
7.2.3.1.2A.1.2	Concatenated Addressing Method.....	51
7.2.3.1.2A.1.3	Separate Network Routing Number Addressing Method .....	51
7.2.3.1.2A.2	Number Portability Forward Information .....	52
7.2.3.1.2B	Coding of the IAM for Carrier Routeing .....	52
7.2.3.1.2B.1	Coding of the IAM when a Carrier Identification Code (CIC) is present .....	52
7.2.3.1.2B.2	Void .....	52
7.2.3.1.3	Sending of COT.....	52
7.2.3.1.3A	Sending of SAM .....	53
7.2.3.1.3A.1	General.....	53
7.2.3.1.3A.2	Additional digits received in in-dialog SIP INFO requests.....	53
7.2.3.1.3A.3	Additional digits received in SIP INVITE requests .....	53
7.2.3.1.4	Sending of 180 ringing .....	54
7.2.3.1.4.0	General.....	54
7.2.3.1.4.0a	PSTN XML body.....	55
7.2.3.1.4.0b	Fallback by I-MGCF.....	56
7.2.3.1.4.1	Fallback in a succeeding network: Transmission Medium Used parameter (TMU) received.....	57
7.2.3.1.4A	Sending of 183 Session Progress for early media scenarios.....	57
7.2.3.1.4B	Sending of 181Call is being forwarded .....	60
7.2.3.1.4C	Sending of 183 Session Progress for overlap signalling using the in-dialog method .....	61
7.2.3.1.4D	Sending of 183 Session Progress to carry ISUP Cause .....	61
7.2.3.1.4E	Sending of 183 Session Progress for ICS call .....	61
7.2.3.1.5	Sending of the 200 OK (INVITE) .....	62
7.2.3.1.6	Sending of the Release message (REL).....	64
7.2.3.1.7	Coding of the REL.....	64
7.2.3.1.8	Receipt of the Release Message .....	65
7.2.3.1.9	Receipt of RSC, GRS or CGB (H/W oriented) .....	68
7.2.3.1.9a	Receipt of REFER .....	69
7.2.3.1.10	Autonomous Release at I-MGCF .....	69
7.2.3.1.11	Internal through connection of the bearer path.....	70
7.2.3.2	Outgoing Call Interworking from ISUP to SIP at O-MGCF.....	70
7.2.3.2.1	Sending of INVITE .....	70
7.2.3.2.1.1	General.....	70
7.2.3.2.1.2	Interaction with continuity check.....	70
7.2.3.2.1.3	IAM without calling party number .....	71
7.2.3.2.1.4	Terminating overlap signalling at MGCF.....	72
7.2.3.2.1.5	Fallback (optional) .....	72
7.2.3.2.1a	Sending of INVITE without determining the end of address signalling.....	73
7.2.3.2.1a.1	General.....	73
7.2.3.2.1a.2	Additional digits sent with in-dialog overlap method.....	73
7.2.3.2.1a.3	Additional digits sent using the multiple INVITEs overlap method.....	74
7.2.3.2.2	Coding of the INVITE.....	75
7.2.3.2.2.0	Overview.....	75
7.2.3.2.2.1	Request-URI and To header field .....	75
7.2.3.2.2.2	SDP Media Description .....	76
7.2.3.2.2.3	P-Asserted-Identity, From and Privacy header fields .....	80
7.2.3.2.2.3A	"cp" URI Parameter in P-Asserted-Identity Header .....	85
7.2.3.2.2.3B	"oli" URI Parameter in P-Asserted-Identity Header .....	85
7.2.3.2.2.4	Max Forwards header .....	86
7.2.3.2.2.5	IMS Communication Service Identifier .....	86
7.2.3.2.2.6	P-Early-Media header field.....	86

7.2.3.2.2.7	PSTN XML elements.....	86
7.2.3.2.2.8	Progress indicator .....	87
7.2.3.2.2.9	P-Access-Network-Info .....	88
7.2.3.2.2A	Coding of the INVITE when Number Portability is supported .....	88
7.2.3.2.2A.1	Request-URI and To header field .....	88
7.2.3.2.2A.1.1	Separate Directory Number Addressing Method .....	89
7.2.3.2.2A.1.2	Concatenated Addressing Method.....	89
7.2.3.2.2A.1.3	Separate Network Routing Number Addressing Method .....	90
7.2.3.2.2B	Coding of the INVITE for Carrier Routeing .....	90
7.2.3.2.2B.1	Mapping of "cic" in Request-URI.....	90
7.2.3.2.2B.2	Void .....	90
7.2.3.2.2C	Coding of INVITE with instance-id in form of IMEI URN .....	90
7.2.3.2.2.10	PSAP Call-back indication .....	91
7.2.3.2.2.11	History-Info header field (optional) .....	91
7.2.3.2.3	Receipt of CONTINUITY.....	91
7.2.3.2.4	Sending of ACM and awaiting answer indication .....	92
7.2.3.2.5	Coding of the ACM .....	94
7.2.3.2.5.0	General.....	94
7.2.3.2.5.1	Backward call indicators.....	94
7.2.3.2.5.2	Optional Backward call indicators .....	96
7.2.3.2.5.3	Access Transport Parameter, Transmission medium used parameter .....	96
7.2.3.2.5.4	Progress indicator .....	97
7.2.3.2.5.5	Cause Value .....	97
7.2.3.2.6	Sending of the Call Progress message (CPG).....	97
7.2.3.2.6.0	General.....	97
7.2.3.2.6.1	Handling of the progress indicator.....	99
7.2.3.2.7	Coding of the CPG .....	99
7.2.3.2.7.0	General.....	99
7.2.3.2.7.1	Event information .....	100
7.2.3.2.7.2	Access Transport Parameter.....	100
7.2.3.2.7.3	Void .....	100
7.2.3.2.7.4	Handling of Backward Call indicators.....	100
7.2.3.2.7.5	Optional Backward call indicators .....	100
7.2.3.2.7.6	Cause Value .....	100
7.2.3.2.7a	Receipt of 200 OK(INVITE).....	101
7.2.3.2.7b	Internal through connection of the bearer path.....	101
7.2.3.2.8	Sending of the Answer Message (ANM).....	101
7.2.3.2.9	Coding of the ANM.....	101
7.2.3.2.9.1	Backwards Call Indicators .....	101
7.2.3.2.9.2	Access Transport Parameter.....	101
7.2.3.2.9.3	Transmission Medium Used parameter (TMU).....	102
7.2.3.2.10	Sending of the Connect message (CON) .....	103
7.2.3.2.11	Coding of the CON.....	103
7.2.3.2.11.0	General.....	103
7.2.3.2.11.1	Backward call indicators.....	103
7.2.3.2.11.2	Access Transport Parameter.....	103
7.2.3.2.11.3	Transmission medium used parameter.....	103
7.2.3.2.11A	Receipt of a reINVITE request.....	103
7.2.3.2.12	Receipt of Status Codes 4xx, 5xx or 6xx.....	103
7.2.3.2.12.1	Special handling of 404 Not Found and 484 Address Incomplete responses after sending of INVITE without determining the end of address signalling .....	106
7.2.3.2.13	Receipt of a BYE.....	106
7.2.3.2.14	Receipt of the Release Message .....	106
7.2.3.2.15	Receipt of RSC, GRS or CGB (H/W oriented) .....	106
7.2.3.2.16	Autonomous Release at O-MGCF.....	107
7.2.3.2.17	Special handling of 580 precondition failure received in response to either an INVITE or UPDATE .....	107
7.2.3.2.17.1	580 Precondition failure response to an INVITE.....	107
7.2.3.2.17.2	580 Precondition failure response to an UPDATE within an early dialog.....	107
7.2.3.2.18	Sending of CANCEL.....	107
7.2.3.2.19	Receipt of SIP redirect (3xx) response .....	108
7.2.3.2.20	Sending of INFO for overlap signalling using the in-dialog method .....	108

7.2.3.2.20.1	General.....	108
7.2.3.2.20.2	Encoding of the INFO.....	108
7.2.3.3	Timers .....	109
7.3	Interworking between CS networks supporting BICC and the IM CN subsystem.....	109
7.3.1	Services performed by network entities in the control plane .....	110
7.3.2	Signalling interactions between network entities in the control plane.....	110
7.3.2.1	Signalling between the SS7 signalling function and MGCF.....	110
7.3.2.1.1	Signalling from MGCF to SS7 signalling function .....	110
7.3.2.1.2	Signalling from SS7 signalling function to MGCF .....	110
7.3.2.1.3	Services offered by STC, SCTP and M3UA .....	111
7.3.2.1.3.1	Services offer by SCTP.....	111
7.3.2.1.3.2	Services offered by M3UA .....	111
7.3.2.1.3.3	Services offered by STC .....	111
7.3.2.2	Signalling between the MGCF and SIP signalling function.....	111
7.3.3	SIP-BICC protocol interworking .....	111
7.3.3.1	Incoming call interworking from SIP to BICC at I-MGCF.....	111
7.3.3.1.1	Sending of IAM.....	111
7.3.3.1.2	Coding of IAM .....	111
7.3.3.1.2.1	Called party number.....	111
7.3.3.1.2.2	Nature of connection indicators .....	111
7.3.3.1.2.3	Forward call indicators .....	112
7.3.3.1.2.4	Calling party's category .....	112
7.3.3.1.2.4A	Originating Line Information.....	112
7.3.3.1.2.5	Transmission medium requirement.....	112
7.3.3.1.2.6	Calling party number .....	112
7.3.3.1.2.7	Generic number.....	112
7.3.3.1.2.8	User service information.....	112
7.3.3.1.2.9	Hop counter (National option) .....	112
7.3.3.1.2.10	Location Number .....	112
7.3.3.1.2.11	Support of ICS call .....	112
7.3.3.1.2.12	UID capability indicators (National option) .....	112
7.3.3.1.2A	Coding of the IAM when Number Portability is supported.....	112
7.3.3.1.2B	Coding of the IAM for Carrier Routeing.....	112
7.3.3.1.2.13	Called IN number and original called IN number (optional) .....	112
7.3.3.1.3	Sending of COT.....	113
7.3.3.1.3A	Sending of SAM.....	113
7.3.3.1.4	Sending of 180 Ringing.....	113
7.3.3.1.4A	Sending of 183 Session Progress for early media scenarios.....	113
7.3.3.1.4B	Sending of 181 Call is being forwarded.....	113
7.3.3.1.4C	Sending of 183 Session Progress for overlap signalling using the in-dialog method .....	113
7.3.3.1.4D	Sending of 183 Session Progress to carry ISUP Cause .....	113
7.3.3.1.4E	Sending of 183 Session Progress for ICS call .....	113
7.3.3.1.5	Sending of the 200 OK (INVITE).....	114
7.3.3.1.6	Sending of the Release message (REL).....	114
7.3.3.1.7	Coding of the REL.....	114
7.3.3.1.8	Receipt of the Release Message .....	114
7.3.3.1.9	Receipt of RSC, GRS or CGB (H/W oriented) .....	114
7.3.3.1.9a	Receipt of REFER .....	114
7.3.3.1.9b	Autonomous Release at I-MGCF .....	114
7.3.3.1.10	Internal through connection of the bearer path.....	114
7.3.3.1.11	Out of Band DTMF .....	114
7.3.3.2	Outgoing Call Interworking from BICC to SIP at O-MGCF .....	114
7.3.3.2.1	Sending of INVITE .....	114
7.3.3.2.1a	Sending of INVITE without determining the end of address signalling.....	115
7.3.3.2.2	Coding of the INVITE.....	115
7.3.3.2.2.1	Request-URI and To header field .....	115
7.3.3.2.2.2	SDP Media Description .....	115
7.3.3.2.2.3	P-Asserted-Identity and privacy header fields .....	115
7.3.3.2.2.3A	"cpc" URI Parameter in P-Asserted-Identity Header .....	115
7.3.3.2.2.3B	"oli" URI Parameter in P-Asserted-Identity Header .....	115
7.3.3.2.2.4	Max Forwards header .....	115
7.3.3.2.2.5	IMS Communication Service Identifier.....	116

7.3.3.2.2.6	P-Access-Network-Info .....	116
7.3.3.2.2A	Coding of the INVITE when number portability is supported .....	116
7.3.3.2.2B	Coding of the INVITE for Carrier Routeing .....	116
7.3.3.2.2C	Coding of INVITE with instance-id in form of IMEI URN .....	116
7.3.3.2.2.7	PSAP Call-back indication .....	116
7.3.3.2.2.8	History-Info header field (optional) .....	116
7.3.3.2.3	Sending of UPDATE .....	116
7.3.3.2.4	Sending of ACM and Awaiting Answer indication .....	117
7.3.3.2.5	Coding of the ACM .....	117
7.3.3.2.5.1	Backward call indicators .....	117
7.3.3.2.6	Sending of the Call Progress message (CPG) .....	117
7.3.3.2.7	Coding of the CPG .....	117
7.3.3.2.7.1	Event information .....	117
7.3.3.2.7.2	Optional Backward call indicators .....	117
7.3.3.2.7a	Receipt of 200 OK (INVITE) .....	117
7.3.3.2.7b	Internal through connection of the bearer path .....	117
7.3.3.2.8	Sending of the Answer Message (ANM) .....	117
7.3.3.2.9	Coding of the ANM .....	117
7.3.3.2.10	Sending of the Connect message (CON) .....	117
7.3.3.2.11	Coding of the CON .....	117
7.3.3.2.11.1	Void .....	118
7.3.3.2.11A	Receipt of re-INVITE requests .....	118
7.3.3.2.12	Receipt of Status Codes 4xx, 5xx or 6xx .....	118
7.3.3.2.13	Receipt of a BYE .....	118
7.3.3.2.14	Receipt of the Release Message .....	118
7.3.3.2.15	Receipt of RSC, GRS or CGB (H/W oriented) .....	118
7.3.3.2.16	Out of Band DTMF .....	118
7.3.3.2.17	Sending of CANCEL .....	118
7.3.3.2.18	Autonomous Release at O-MGCF .....	118
7.3.3.2.19	Special handling of 580 precondition failure received in response to either an INVITE or UPDATE .....	118
7.3.3.2.20	Receipt of SIP redirect (3xx) response .....	118
7.3.3.2.21	Sending of INFO for overlap signalling using the in-dialog method .....	118
7.3.3.3	Timers .....	119
7.4	Supplementary Services .....	119
7.4.1	Calling line identification presentation/restriction (CLIP/CLIR) .....	119
7.4.2	Connected line presentation and restriction (COLP/COLR) .....	119
7.4.2.0	General .....	119
7.4.2.1	Incoming Call Interworking from SIP to BICC/ISUP at the I-MGCF .....	119
7.4.2.1.1	INVITE to IAM interworking (SIP to ISUP/BICC calls) .....	119
7.4.2.1.2	ANM/CON to 200 OK (INVITE) .....	119
7.4.2.2	Outgoing Call Interworking from BICC/ISUP to SIP at O-MGCF .....	120
7.4.2.2.1	IAM to INVITE interworking (ISUP to SIP calls) .....	120
7.4.2.2.2	1XX to ANM or CON interworking .....	121
7.4.2.2.3	200 OK (INVITE) to ANM/CON interworking .....	121
7.4.3	Direct Dialling In (DDI) .....	122
7.4.4	Malicious call identification .....	122
7.4.5	Subaddressing (SUB) .....	122
7.4.5.1	General .....	122
7.4.5.2	Interworking at I-MGCF .....	123
7.4.5.3	Interworking at O-MGCF .....	124
7.4.6	Call Forwarding Busy (CFB)/ Call Forwarding No Reply (CFNR) / Call Forwarding Unconditional (CFU) / Call Deflection (CD) .....	125
7.4.6.1	General .....	125
7.4.6.2	Interworking at the O-MGCF .....	125
7.4.6.2.1	General .....	125
7.4.6.2.2	Interworking SIP to ISUP .....	125
7.4.6.2.3	Interworking ISUP to SIP .....	129
7.4.6.3	Interworking at the I-MGCF .....	132
7.4.6.3.1	General .....	132
7.4.6.3.2	Interworking from SIP to ISUP .....	133
7.4.6.3.3	Interworking from ISUP to SIP .....	135



7.4.7	Void .....	141
7.4.8	Explicit Call Transfer (ECT) .....	141
7.4.9	Call Waiting .....	141
7.4.10	Call Hold.....	141
7.4.10.1	Session hold initiated from the IM CN subsystem side.....	141
7.4.10.2	Session hold initiated from the CS network side.....	142
7.4.11	Call Completion on busy subscriber .....	145
7.4.12	Completion of Calls on No Reply (CCNR) .....	145
7.4.13	Terminal Portability (TP).....	145
7.4.14	Conference calling (CONF) / Three-Party Service (3PTY).....	145
7.4.15	Void .....	146
7.4.16	Closed User Group (CUG) .....	146
7.4.17	Multi-Level Precedence and Pre-emption (MLPP).....	146
7.4.18	Global Virtual Network Service (GVNS).....	146
7.4.19	International telecommunication charge card (ITCC) .....	146
7.4.20	Reverse charging (REV).....	146
7.4.21	User-to-User Signalling (UUS).....	146
7.4.21.0	General .....	146
7.4.21.1	User-to-User Signalling (UUS) service 1 (implicit).....	146
7.4.21.1.0	General .....	146
7.4.21.1.1	Void .....	147
7.4.21.1.2	User-to-user information Interworking from SIP to ISUP .....	147
7.4.21.1.3	User-to-user information Interworking from ISUP to SIP .....	147
7.4.21.2	User-to-User Signalling (UUS) service 1 (explicit) .....	147
7.4.21.3	User-to-User Signalling (UUS) service 2 (explicit) .....	147
7.4.21.4	User-to-User Signalling (UUS) service 3 (explicit) .....	147
7.4.22	Multiple Subscriber Number (MSN) .....	148
7.4.23	Anonymous Call rejection .....	148
7.4.23.0	General .....	148
7.4.23.1	ISUP-SIP protocol interworking at the I-MGCF.....	148
7.4.23.2	SIP-ISUP protocol interworking at the O-MGCF.....	148
7.4.24	Customized Alerting Tones (CAT) in the 3GPP CS domain.....	148
7.5	IMS Supplementary Services .....	148
7.5.1	Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR).....	148
7.5.2	Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR).....	148
7.5.2.1	General .....	148
7.5.2.2	Interworking at the O-MGCF.....	148
7.5.2.3	Interworking at the I-MGCF .....	150
7.5.2.4	Timer.....	151
7.5.3	void .....	151
7.5.4	Communication Diversion (CDIV).....	151
7.5.4.1	General .....	151
7.5.4.2	Interworking at the O-MGCF.....	152
7.5.4.2.1	General .....	152
7.5.4.2.2	Call Diversion within the ISUP Network appeared.....	157
7.5.4.3	Interworking at the I-MGCF .....	160
7.5.5	Communication Hold (HOLD).....	168
7.5.6	Conference call (CONF).....	168
7.5.6.1	General .....	168
7.5.6.2	Subscribing for the conference event package .....	168
7.5.6.3	Interworking the notification.....	169
7.5.7	Anonymous Communication Rejection (ACR) and Communication Barring (CB).....	169
7.5.8	Message Waiting Indication (MWI) .....	170
7.5.9	Malicious Communication Identification (MCID) .....	170
7.5.9.0	General .....	170
7.5.9.1	Interworking at the O-MGCF.....	170
7.5.9.1.0	General .....	170
7.5.9.1.1	Interworking of the MCID XML Request schema with the ISUP MCID request indicators .....	170
7.5.9.1.2	Interworking of the ISUP MCID response indicators with the MCID XML Response schema ...	171
7.5.9.1.3	Interworking of the ISUP Calling Party Number in an Identification Response with the OrigPartyIdentity within the MCID XML Response schema.....	171

7.5.9.1.4	Interworking of the ISUP Generic Number in an Identification Response with the GenericNumber within the MCID XML Response schema .....	171
7.5.9.2	Interworking at the I-MGCF .....	171
7.5.9.2.1	General .....	171
7.5.9.2.2	Interworking of identification Request .....	171
7.5.9.2.3	Interworking of identification Response.....	172
7.5.10	Closed User Group (CUG) .....	172
7.5.10.0	General .....	172
7.5.10.1	Interworking at the I-MGCF .....	172
7.5.10.2	Interworking at the O-MGCF.....	173
7.5.11	CCBS/CCNR .....	174
7.5.11.0	General .....	174
7.5.11.1	Interworking at the I-MGCF .....	174
7.5.11.2	Interworking at the O-MGCF.....	176
7.5.12	Communication Waiting (CW).....	178
7.5.12.0	General .....	178
7.5.12.1	Interworking at the I-MGCF .....	178
7.5.12.2	Interworking at the O-MGCF.....	178
7.5.13	Customized Alerting Tones (CAT).....	178
7.5.13.1	General .....	178
7.5.13.2	Early session model.....	178
7.5.13.2.1	Interworking at the O-MGCF .....	178
7.5.13.2.2	MGCF - IM-MGW Interaction for CAT service .....	179
7.5.13.2.2.1	General.....	179
7.5.13.2.2.2	IM CN subsystem side Early Session establishment .....	179
7.5.13.2.2.3	IM CN subsystem side normal Session establishment.....	179
7.5.13.2.2.4	Called party alerting.....	180
7.5.13.2.2.5	Message sequence chart.....	180
7.5.13.3	Forking model.....	183
7.5.13.4	Gateway model .....	183
8	User plane interworking .....	183
8.1	Interworking between IM CN subsystem and bearer independent CS network .....	183
8.1.1	Transcoder-less Mb to Nb Interworking.....	184
8.1.1.1	Initialisation .....	184
8.1.1.2	Time alignment .....	184
8.1.1.3	Rate control .....	185
8.1.1.3.1	General .....	185
8.1.1.3.2	Rate control for AMR and AMR-WB codecs .....	185
8.1.1.3.3	Rate and Mode control for the EVS codec .....	185
8.1.1.3.4	Interworking of rate control between compatible AMR-WB and EVS codec configurations.....	186
8.1.1.3.5	Interworking between rate control and RTCP-APP-CMR messages for MTSI .....	187
8.1.1.4	Frame quality indication .....	187
8.1.1.5	Framing .....	187
8.1.1.6	Transcoding.....	188
8.1.1.7	Discontinuous transmission .....	188
8.1.1.8	Timing and sequence information.....	188
8.2	Interworking between IM CN subsystem and TDM-based CS network .....	188
8.3	Transcoding requirements .....	189
8.4	Diffserv code point requirements .....	189
8.5	DTMF handling.....	189
9	MGCF - IM-MGW Interaction.....	189
9.1	Overview .....	189
9.2	Mn signalling interactions .....	190
9.2.1	Network model .....	190
9.2.2	Basic IM CN subsystem originated session.....	190
9.2.2.1	BICC forward bearer establishment.....	190
9.2.2.1.1	IM-MGW selection .....	190
9.2.2.1.2	CS network side bearer establishment.....	190
9.2.2.1.3	IM CN subsystem side termination reservation.....	190
9.2.2.1.4	IM CN subsystem side session establishment .....	191

9.2.2.1.5	Through-connection .....	191
9.2.2.1.6	Codec handling.....	192
9.2.2.1.7	Failure handling in MGCF .....	192
9.2.2.1.8	Message sequence chart.....	192
9.2.2.2	BICC backward bearer establishment .....	193
9.2.2.2.1	IM-MGW selection .....	193
9.2.2.2.2	IM CN subsystem side termination reservation.....	194
9.2.2.2.3	IM CN subsystem side session establishment .....	194
9.2.2.2.4	CS network side bearer establishment.....	194
9.2.2.2.5	Through-connection .....	195
9.2.2.2.6	Codec handling.....	195
9.2.2.2.7	Failure handling in MGCF .....	195
9.2.2.2.8	Message sequence chart.....	195
9.2.2.3	ISUP .....	196
9.2.2.3.1	IM-MGW selection .....	196
9.2.2.3.2	IM CN subsystem side termination reservation.....	196
9.2.2.3.3	IM CN subsystem side session establishment .....	197
9.2.2.3.4	CS network side circuit reservation.....	197
9.2.2.3.5	Through-connection .....	197
9.2.2.3.6	Continuity check.....	198
9.2.2.3.7	Codec handling.....	198
9.2.2.3.8	Voice processing function .....	198
9.2.2.3.9	Failure handling in MGCF .....	198
9.2.2.3.10	Message sequence chart.....	198
9.2.3	Basic CS network originated session .....	200
9.2.3.1	BICC forward bearer establishment.....	200
9.2.3.1.1	IM-MGW selection .....	200
9.2.3.1.2	IM CN subsystem side termination reservation.....	200
9.2.3.1.3	IM CN subsystem side session establishment .....	200
9.2.3.1.4	CS network side bearer establishment.....	200
9.2.3.1.5	Called party alerting .....	201
9.2.3.1.6	Called party answer .....	201
9.2.3.1.7	Through-Connection.....	201
9.2.3.1.8	Codec handling.....	201
9.2.3.1.9	Failure handling in MGCF .....	201
9.2.3.1.10	Message sequence chart.....	201
9.2.3.2	BICC Backward bearer establishment .....	203
9.2.3.2.1	IM-MGW selection .....	203
9.2.3.2.2	CS network side bearer establishment.....	203
9.2.3.2.3	IM CN subsystem side termination reservation.....	203
9.2.3.2.4	IM CN subsystem side session establishment .....	204
9.2.3.2.5	Called party alerting .....	204
9.2.3.2.6	Called party answer .....	204
9.2.3.2.7	Through-Connection.....	204
9.2.3.2.8	Codec handling.....	204
9.2.3.2.9	Failure handling in MGCF .....	205
9.2.3.2.10	Message sequence chart.....	205
9.2.3.3	ISUP .....	207
9.2.3.3.1	IM-MGW selection .....	207
9.2.3.3.2	CS network side circuit reservation.....	207
9.2.3.3.3	IM CN subsystem side termination reservation.....	207
9.2.3.3.4	IM CN subsystem side session establishment .....	207
9.2.3.3.5	Called party alerting .....	208
9.2.3.3.6	Called party answer .....	208
9.2.3.3.7	Through-Connection.....	208
9.2.3.3.8	Continuity Check.....	208
9.2.3.3.9	Codec handling.....	208
9.2.3.3.10	Voice Processing function.....	209
9.2.3.3.11	Failure handling in MGCF .....	209
9.2.3.3.12	Message sequence chart.....	209
9.2.3.4	Handling of Forking .....	211
9.2.3.4.1	Detection of Forking.....	211

9.2.3.4.2	IM CN subsystem side session establishment .....	211
9.2.3.4.3	IM CN subsystem side session establishment completion .....	213
9.2.3.4.4	Message sequence chart.....	214
9.2.4	Session release initiated from IM CN subsystem side .....	217
9.2.4.1	BICC .....	217
9.2.4.1.1	Session release in the IM CN subsystem side.....	217
9.2.4.1.2	Session release in the CS network side.....	217
9.2.4.1.3	Message sequence chart.....	217
9.2.4.2	ISUP .....	218
9.2.4.2.1	Session release in the IM CN subsystem side.....	218
9.2.4.2.2	Session release in the CS network side.....	218
9.2.4.2.3	Message sequence chart.....	218
9.2.5	Session release initiated from CS network side .....	218
9.2.5.1	BICC .....	218
9.2.5.1.1	Session release in the CS network side.....	218
9.2.5.1.2	Session release in the IM CN subsystem side.....	219
9.2.5.1.3	Message sequence chart.....	219
9.2.5.2	ISUP .....	219
9.2.5.2.1	Session release in the CS network side.....	219
9.2.5.2.2	Session release in the IM CN subsystem side.....	219
9.2.5.2.3	Message sequence chart.....	220
9.2.6	Session release initiated by MGCF.....	220
9.2.6.1	BICC .....	220
9.2.6.1.1	Session release in the CS network side.....	220
9.2.6.1.2	Session release in the IM CN subsystem side.....	220
9.2.6.1.3	Message sequence chart.....	220
9.2.6.2	ISUP .....	221
9.2.6.2.1	Session release in the CS network side.....	221
9.2.6.2.2	Session release in the IM CN subsystem side.....	221
9.2.6.2.3	Message sequence chart.....	221
9.2.7	Session release initiated by IM-MGW.....	222
9.2.7.1	BICC .....	222
9.2.7.1.1	Session release in the CS network side.....	222
9.2.7.1.2	Session release in the IM CN subsystem side.....	222
9.2.7.1.3	Message sequence chart.....	222
9.2.7.2	ISUP .....	223
9.2.7.2.1	Session release in the CS network side.....	223
9.2.7.2.2	Session release in the IM CN subsystem side.....	223
9.2.7.2.3	Message sequence chart.....	224
9.2.8	Handling of RTP telephone events .....	224
9.2.8.1	Sending DTMF digits out-of-band to CS CN (BICC).....	225
9.2.8.2	Sending and receiving DTMF digits inband to/from CS CN (ISUP or BICC) .....	227
9.2.8.3	Receiving DTMF digits out-of-band from CS CN (BICC).....	228
9.2.9	Session hold initiated from IM CN subsystem .....	229
9.2.10	Session hold initiated from CS network .....	231
9.2.11	Explicit Congestion Notification Support.....	233
9.2.11.1	General .....	233
9.2.11.1a	Incoming Call Interworking from SIP to ISUP/BICC at I-MGCF .....	233
9.2.11.1b	Outgoing Call Interworking from ISUP/BICC to SIP at O-MGCF .....	234
9.2.11.1c	Detection of ECN failures by IM-MGW.....	234
9.2.11.2	Message sequence chart .....	234
9.2.12	Interactive Connectivity Establishment Support.....	235
9.2.12.1	General .....	235
9.2.12.2	ICE lite .....	235
9.2.12.3	Full ICE.....	236
9.2.12.4	Procedures for Interactive Connectivity Establishment (ICE) .....	238
9.2.12.4.1	ICE lite .....	238
9.2.12.4.2	Full ICE .....	239
9.2.12.4.3	Connectivity check result notification (full ICE) .....	240
9.2.12.4.4	New peer reflexive candidate notification (full ICE) .....	240
9.2.13	Codec Parameters Handling.....	241
9.2.13.1	Handling of common codec parameters.....	241

9.2.13.2	Handling of EVS speech codec parameters when interworking with a different codec.....	242
9.2.13.3	Handling of EVS speech codec parameters when the EVS codec is used end-to-end .....	254
9.2.14	SDP Capability Negotiation (SDPCapNeg).....	264
9.2.15	Rate adaptation for media endpoints.....	264
9.3	Mn Signalling procedures .....	264
9.3.1	Procedures related to terminations towards the IM CN Subsystem .....	264
9.3.1.1	Reserve IMS connection point .....	265
9.3.1.2	Configure IMS resources .....	268
9.3.1.3	Reserve IMS Connection point and configure remote resources .....	270
9.3.1.4	Release IMS termination .....	273
9.3.1.5	Detect IMS RTP Tel event.....	273
9.3.1.6	Notify IMS RTP Tel event.....	273
9.3.1.7	Void.....	273
9.3.1.8	Send IMS RTP Tel event .....	273
9.3.1.9	Stop IMS RTP Tel event .....	273
9.3.1.10	Termination heartbeat indication .....	274
9.3.1.11	IMS Bearer Released.....	274
9.3.1.12	End IMS RTP Tel event.....	274
9.3.1.13	IMS Send Tone .....	274
9.3.1.14	IMS Stop Tone .....	274
9.3.1.15	IMS Tone Completed.....	274
9.3.1.16	ECN Failure Indication .....	275
9.3.1.17	ICE Connectivity Check Result Notification .....	275
9.3.1.18	ICE New Peer Reflexive Candidate Notification.....	276
9.3.2	Procedures related to a termination towards an ISUP network.....	276
9.3.2.1	Reserve TDM circuit.....	276
9.3.2.2	Change TDM through-connection .....	277
9.3.2.3	Activate TDM voice-processing function .....	277
9.3.2.4	Send TDM tone.....	277
9.3.2.5	Stop TDM tone.....	277
9.3.2.6	Play TDM announcement .....	277
9.3.2.7	TDM announcement completed .....	277
9.3.2.8	Stop TDM announcement .....	277
9.3.2.9	Continuity check .....	277
9.3.2.10	Continuity check verify.....	278
9.3.2.11	Continuity check response .....	278
9.3.2.12	Release TDM termination .....	278
9.3.2.13	Termination Out-of-Service .....	279
9.3.2.14	Termination heartbeat indication .....	279
9.3.2.15	Bearer Released.....	279
9.3.2.16	TDM tone completed .....	279
9.3.3	Procedures related to a termination towards a BICC network.....	279
9.3.4	Non-call related procedures .....	280
9.3.5	Multiple IP Realms .....	281
9.4	Multimedia Priority Service (MPS) Support .....	282
9.4.1	General.....	282
9.4.2	IM-MGW Resource Congestion in ADD response, request is queued .....	282
9.4.3	IM-MGW Resource Congestion in ADD response, MGCF seizes new IM-MGW .....	283
9.4.4	IM-MGW Priority Resource Allocation .....	283
9.4.5	IM-MGW Priority User Data marking.....	284
<b>Annex A (informative):</b>	<b>Void .....</b>	<b>285</b>
<b>Annex B (normative):</b>	<b>Codec Negotiation between a BICC CS network and the IM CN subsystem.....</b>	<b>286</b>
B.1	Introduction .....	286
B.2	Control plane interworking .....	286
B.2.1	Incoming call interworking from SIP to BICC at I-MGCF.....	286
B.2.1.1	Sending of IAM .....	286
B.2.1.2	Sending of SDP answer .....	286
B.2.2	Outgoing call interworking from BICC to SIP at O-MGCF .....	286

B.2.2.1	Sending of INVITE.....	286
B.2.2.2	Responding to serving node initiating codec negotiation .....	287
B.2.3	Mid-call interworking from SIP to BICC at I-MGCF or O-MGCF .....	287
B.2.3.1	Receipt of SDP offer.....	287
B.2.3.2	Generating SDP answer .....	288
B.2.4	Mid-call interworking from BICC to SIP at I-MGCF or O-MGCF .....	288
B.2.4.1	Receipt of mid-call codec negotiation request .....	288
B.2.4.2	Responding to serving node initiating mid-call codec negotiation .....	288
B.2.4.3	Receipt of codec modification request.....	289
B.2.5	Codec parameter translation between BICC CS network and the IM CN subsystem .....	289
B.2.5.1	Codec parameters for 3GPP AMR-NB codecs .....	290
B.2.5.2	Codec parameters for 3GPP AMR-WB codecs .....	291
B.2.5.3	Codec parameters for 3GPP non-AMR codecs.....	293
B.2.5.4	Codec parameters for ITU-T codecs.....	293
B.2.5.5	Codec parameters for 3GPP EVS codec .....	294
B.3	MGCF - IM-MGW interaction during interworking of codec negotiation .....	296
B.3.1	Basic IM CN subsystem originated session .....	296
B.3.1.1	BICC forward bearer establishment .....	297
B.3.1.1.1	IM-MGW selection .....	297
B.3.1.1.2	CS network side bearer establishment.....	297
B.3.1.1.3	IM CN subsystem side session establishment .....	297
B.3.1.1.4	Through-connection .....	297
B.3.1.1.5	Codec handling .....	298
B.3.1.1.6	Failure handling in MGCF .....	298
B.3.1.1.7	Message sequence chart .....	298
B.3.2	Basic CS network originated session.....	300
B.3.2.1	BICC forward bearer establishment.....	300
B.3.2.1.1	IM-MGW selection .....	300
B.3.2.1.2	IM CN subsystem side termination reservation .....	300
B.3.2.1.3	IM CN subsystem side session establishment .....	300
B.3.2.1.4	CS network side bearer establishment.....	300
B.3.2.1.5	Called party alerting .....	301
B.3.2.1.6	Called party answer.....	301
B.3.2.1.7	Through-Connection .....	301
B.3.2.1.8	Codec handling .....	301
B.3.2.1.9	Failure handling in MGCF .....	301
B.3.2.1.10	Message sequence chart .....	301
B.3.3	CS network initiated mid-call codec negotiation.....	304
B.3.4	IM CN subsystem initiated mid-call codec negotiation.....	305
<b>Annex C (normative): Interworking of CPC parameter .....</b>		<b>306</b>
C.1	Interworking SIP to ISUP.....	306
C.2	Interworking ISUP to SIP.....	307
<b>Annex D: Void .....</b>		<b>308</b>
<b>Annex E (normative): Multimedia interworking between the IP Multimedia Core Network (CN) Subsystem (IMS) and Circuit Switched (CS) networks .....</b>		<b>309</b>
E.1	Basic Multimedia calls interworking between the IMS and CS Networks scenarios .....	309
E.2	Control plane interworking .....	310
E.2.1	General .....	310
E.2.2	Functionalities required in the MGCF for multimedia calls support.....	310
E.2.3	IM CN subsystem originated session .....	310
E.2.3.1	Preconditions used at IMS side.....	310
E.2.3.1.1	Interactions between H.245 or MONA and SIP/SDP .....	310
E.2.3.2	Preconditions not used at IMS side.....	312
E.2.3.2.1	Interactions between H.245 or MONA and SIP/SDP .....	312
E.2.3.3	Fallback to speech at session establishment .....	314
E.2.4	CS network originated session .....	314

E.2.4.1	Interactions between SIP/SDP and H.245 or MONA .....	314
E.2.4.1.1	Normal Call setup .....	314
E.2.4.1.2	Call setup if multimedia call can not be recognized in an unambiguous manner.....	316
E.2.4.1.3	Fallback to speech during call setup .....	316
E.2.4.2	CS originated - IM CN transit - CS terminated .....	317
E.2.5	Service change.....	318
E.2.5.2.1	SCUDIF .....	318
E.2.5.2.1.0	General .....	318
E.2.5.2.1.1	IM CN subsystem originated change.....	318
E.2.5.2.1.1.1	Change from multimedia to speech .....	318
E.2.5.2.1.1.2	Change from speech to multimedia .....	319
E.2.5.2.1.2	CS network originated change.....	320
E.2.5.2.1.2.1	Change from multimedia to speech .....	320
E.2.5.2.1.2.2	Change from speech to multimedia .....	320
E.2.5.2.2	Non-SCUDIF case (ISUP or BICC without SCUDIF) .....	321
E.2.5.2.2.1	Change from multimedia to audio .....	321
E.2.5.2.2.2	Change from speech to multimedia .....	322
E.2.6	Call release .....	322
E.2.6.1	Call release initiated from the IM CN subsystem side.....	322
E.2.6.2	Call release initiated from the CS network side.....	323
E.2.6.3	Call release initiated from the interworking node.....	324
E.3	User plane interworking .....	325
E.3.1	Functionalities required in the IM-MGW for multimedia calls support.....	325
E.4	MGCF and IM-MGW interactions.....	325
E.4.1	Introduction .....	325
E.4.2	Mn signalling interactions .....	326
E.4.2.1	Introduction.....	326
E.4.2.2	H.248 Context Model .....	326
E.4.2.3	Transport of H.245 messages between the MGCF and IM-MGW .....	326
E.4.2.3.1	General .....	326
E.4.2.3.2	Transport from MGCF to IM-MGW.....	327
E.4.2.3.3	Transport from IM-MGW to MGCF.....	327
E.4.2.4	Call establishment procedure.....	327
E.4.2.5	Handling of H.245 indication message .....	330
E.4.2.5.1	Overview .....	330
E.4.2.5.2	Function Not Understood / Function Not Supported message .....	330
E.4.2.5.3	User Input Indication message .....	331
E.4.2.6	Handling of H.245 Command message .....	331
E.4.2.6.1	Overview .....	331
E.4.2.6.2	Flow control command .....	331
E.4.2.6.3	End Session Command .....	332
E.4.2.6.4	videoFastUpdatePicture .....	332
E.4.2.7	Mn Signalling Interactions to support the Media Oriented Negotiation Acceleration (MONA).....	333
E.4.2.7.1	Overview .....	333
E.4.2.7.2	Mn Interactions for MONA preference messages.....	335
E.4.2.7.3	Mn Interactions for MONA MPCs.....	337
E.4.2.7.4	Mn Interactions for MONA SPCs.....	339
E.4.2.7.4.1	General .....	339
E.4.2.7.4.2	Transport from MGCF to IM-MGW .....	339
E.4.2.7.4.3	Transport from IM-MGW to MGCF .....	340
E.4.2.7.4.4	Termination of SPC procedure .....	341
E.4.2.7.5	Mn Interactions for fallback from MONA procedures to standard H.324 setup.....	342
E.4.2.8	Interworking between RTCP messages and H.245 messages.....	343
E.4.2.8.1	Overview.....	343
E.4.2.8.2	IM CN subsystem originated RTCP messages.....	343
E.4.2.8.3	CS network originated H.245 messages.....	344
E.4.3	Mn Signalling procedures .....	345
E.4.3.1	Overview .....	345
E.4.3.2	Add Multiplex Termination .....	346
E.4.3.3	Configure Multiplex Termination.....	347

E.4.3.4	Signal H245 Message .....	348
E.4.3.5	Notify H245 Message .....	348
E.4.3.6	Notify MONA Preference Reception.....	349
E.4.3.7	Notify MONA Preference Completed .....	349
E.4.3.8	Signal SPC.....	350
E.4.3.9	Notify SPC.....	350
E.4.3.10	Notify MPC .....	351
E.4.3.11	Notify Detection of Legacy Interworking.....	351
E.4.3.12	Request RTCP-Interworking .....	352
E.4.3.13	Notify RTCP-Interworking.....	352
E.4.3.14	Signal-H.245-Interworking.....	353
E.4.3.15	Stop MPC .....	353
E.4.3.16	Stop SPC.....	354
E.4.3.17	Stop MONA Negotiation .....	355
<b>Annex F (normative): PSTN XML Scheme.....</b>		<b>356</b>
F.1	Scope .....	356
F.2	MIME type .....	356
F.3	XML Schema definition.....	356
F.4	IANA registration template .....	361
<b>Annex G (informative): Void .....</b>		<b>364</b>
<b>Annex H (normative): Interworking of Originating Line Information (OLI) parameter (network option).....</b>		<b>365</b>
H.1	Interworking SIP to ISUP.....	365
H.2	Interworking ISUP to SIP.....	365
<b>Annex I (normative): GTT interworking between the IP Multimedia Core Network (CN) Subsystem (IMS) and Circuit Switched (CS) networks .....</b>		<b>366</b>
I.1	Overview of GTT interworking between the IMS and Circuit Switched (CS) networks .....	366
I.2	Control plane interworking .....	367
I.2.1	General .....	367
I.2.2	Functionalities required in the MGCF for GTT calls support .....	367
I.2.3	IM CN subsystem originated session .....	367
I.2.3.1	Initial INVITE with an SDP offer including a text media line .....	367
I.2.4	CS network originated session .....	368
I.2.4.1	General.....	368
I.2.4.2	Initial INVITE with an SDP offer including audio only.....	368
I.2.5	Subsequent SDP offer/answer exchange adding text to an audio session .....	369
I.3	User plane interworking .....	369
I.3.1	Functionalities required in the IM-MGW for GTT support .....	369
I.3.2	Monitoring of text/modem signals on the CS side .....	369
I.3.3	Multiplexing between the CS and IMS streams .....	370
I.3.4	Conversion between text/modem and Real-Time Text over RTP .....	370
I.4	MGCF and IM-MGW interactions.....	370
I.4.1	Introduction .....	370
I.4.2	Mn signalling interactions .....	370
I.4.2.1	Introduction.....	370
I.4.2.2	H.248 Context Model .....	371
I.4.2.3	Specific Mn signalling for GTT.....	371
I.4.2.4	IM CN subsystem originated session between the MGCF and IM-MGW .....	372
I.4.2.4.1	Initial INVITE with an SDP offer including Real-Time Text.....	372
I.4.2.5	CS network originated session.....	374
I.4.2.5.1	Initial INVITE with an SDP offer only including audio .....	374
I.4.3	Mn Signalling procedures .....	375



I.4.3.1	Overview .....	375
<b>Annex J (informative):</b>	<b>Call Flow for Customized Alerting Tone (CAT) interworking between CS network and IMS .....</b>	<b>376</b>
J.1	Introduction .....	376
J.2	IMS CAT provided by the terminating IMS user using gateway model towards user served in CS networks .....	376
J.2.1	Callflow without using precondition mechanism .....	376
<b>Annex K (normative):</b>	<b>T.38 interworking between the IP Multimedia Core Network (CN) Subsystem (IMS) and Circuit Switched (CS) networks .....</b>	<b>378</b>
K.1	General .....	378
K.2	Recommend T.38 configuration.....	378
K.3	Handling T.38 SDP attributes at the I-MGCF.....	379
K.4	Handling T.38 SDP attributes at the O-MGCF .....	380
K.5	Applicable T.38 SDP attributes at the Mn Interface .....	380
<b>Annex L (normative):</b>	<b>Interworking of called IN number and original called IN number parameters.....</b>	<b>382</b>
L.0	General .....	382
L.1	Interworking SIP to ISUP.....	382
L.2	Interworking ISUP to SIP.....	383
<b>Annex M (informative):</b>	<b>Change history .....</b>	<b>387</b>
History .....		390

---

# Foreword

This Technical Specification has been produced by the 3rd Generation Partnership Project (3GPP).

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

Version x.y.z

where:

x the first digit:

- 1 presented to TSG for information;
- 2 presented to TSG for approval;
- 3 or greater indicates TSG approved document under change control.

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

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

---

# 1 Scope

The present document specifies the principles of interworking between the 3GPP IM CN subsystem and BICC/ISUP based legacy CS networks, in order to support IM basic voice, data and multimedia calls.

The present 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 BICC or ISUP are detailed in terms of the processes and protocol mappings required for the support of both IM originated and terminated voice and multimedia calls.

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

The present document specifies the interworking between 3GPP profile of SIP (as detailed according to 3GPP TS 24.229 [9]) and BICC or ISUP, as specified in ITU-T Recommendations Q.1902.1 to Q.1902.6 [30] and ITU-T Q761 to Q764 [4] respectively.

The present document also specifies the interworking between circuit switched multimedia telephony service, as described in 3GPP TS 26.110 [78] 3GPP TS 26.111 [79], and ITU-T Recommendation H.324 [81] and packet switched multimedia services, as described in 3GPP TS 26.114 [104], in particular and the interworking between the 3GPP profile of SIP and the inband control protocols for multimedia communication as specified in ITU-T Recommendations H.245 [82] and H.324 Annex K [81].

The present document addresses two interworking scenarios with respect to the properties of the CS network:

- The CS network does not use any 3GPP specific additions.
- The CS network uses 3GPP specific additions.

---

# 2 References

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

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

- [1] ITU-T Recommendation G.711 (11/88): "Pulse Code Modulation (PCM) of voice frequencies".
- [2] ITU-T Recommendation H.248.1 (05/02): "Gateway control protocol: Version 2".
- [3] ITU-T Recommendation Q.701 (03/93), Q.702 (11/88), Q.703 (07/96), Q.704 (07/96), Q.705 (03/93), Q.706 (03/93), Q.707 (11/88), Q.708 (03/99), Q.709 (03/93): "Functional description of the message transfer part (MTP) of Signalling System No. 7".
- [4] ITU-T Recommendations Q.761 to Q.764 (12/99): "Specifications of Signalling System No.7 ISDN User Part (ISUP)".
- [5] Void.
- [6] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".
- [7] Void.
- [8] 3GPP TS 24.228: "Signalling flows for the IP multimedia call control based on SIP and SDP".
- [9] 3GPP TS 24.229: "IP Multimedia Call Control Protocol based on SIP and SDP".