ETSI TS 129 163 V13.4.0 (2016-01)

**Digital cellular telecommunications system (Phase 2+);
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 13.4.0 Release 13)**

A GLOBAL INITIATIVE

Reference
RTS/TSGC-0329163vd40

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
http://portal.etsi.org/tb/status/status.asp

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

Intellectual Property Rights

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....44
7.2.3.1.2.8 User service information....44
7.2.3.1.2.9 Hop Counter (National option) 44
7.2.3.1.2.10 Progress Indicator 45
7.2.3.1.2.11 Location Number 45
7.2.3.1.2.12 Support of ICS call 46
7.2.3.1.2.13 UID capability indicators (National option) 47
7.2.3.1.2.14 Called IN number and original called IN number (optional) 47
7.2.3.1.2A Coding of the IAM when Number Portability is supported....47
7.2.3.1.2A.1 Coding of the IAM when a Number Portability Routing Number is available....47
7.2.3.1.2A.1.1 Separate Directory Number Addressing Method 48
7.2.3.1.2A.1.2 Concatenated Addressing Method....49
7.2.3.1.2A.1.3 Separate Network Routing Number Addressing Method....49
7.2.3.1.2A.2 Number Portability Forward Information 50
7.2.3.1.2B Coding of the IAM for Carrier Routeing....50
7.2.3.1.2B.1 Coding of the IAM when a Carrier Identification Code (CIC) is present....50
7.2.3.1.2B.2 Void 50
7.2.3.1.3 Sending of COT....50
7.2.3.1.3A Sending of SAM....51
7.2.3.1.3A.1 General....51
7.2.3.1.3A.2 Additional digits received in in-dialog SIP INFO requests....51
7.2.3.1.3A.3 Additional digits received in SIP INVITE requests....51
7.2.3.1.4 Sending of 180 ringing 52
7.2.3.1.4.0 General....52
7.2.3.1.4.0a PSTN XML body....53
7.2.3.1.4.0b Fallback by I-MGCF....54
7.2.3.1.4.1 Fallback in a succeeding network: Transmission Medium Used parameter (TMU) received....55
7.2.3.1.4A Sending of 183 Session Progress for early media scenarios....55
7.2.3.1.4B Sending of 181Call is being forwarded 58
7.2.3.1.4C Sending of 183 Session Progress for overlap signalling using the in-dialog method....59
7.2.3.1.4D Sending of 183 Session Progress to carry ISUP Cause 59
7.2.3.1.4E Sending of 183 Session Progress for ICS call 59
7.2.3.1.5 Sending of the 200 OK (INVITE)....60
7.2.3.1.6 Sending of the Release message (REL)....62
7.2.3.1.7 Coding of the REL....62
7.2.3.1.8 Receipt of the Release Message 63
7.2.3.1.9 Receipt of RSC, GRS or CGB (H/W oriented) 65
7.2.3.1.9a Receipt of REFER....66
7.2.3.1.10 Autonomous Release at I-MGCF 66
7.2.3.1.11 Internal through connection of the bearer path....67
7.2.3.2 Outgoing Call Interworking from ISUP to SIP at O-MGCF....67
7.2.3.2.1 Sending of INVITE 67
7.2.3.2.1.1 General....67
7.2.3.2.1.2 Interaction with continuity check....67
7.2.3.2.1.3 IAM without calling party number 68
7.2.3.2.1.4 Terminating overlap signalling at MGCF....69
7.2.3.2.1.5 Fallback (optional)....69
7.2.3.2.1a Sending of INVITE without determining the end of address signalling....70
7.2.3.2.1a.1 General....70
7.2.3.2.1a.2 Additional digits sent with in-dialog overlap method....70
7.2.3.2.1a.3 Additional digits sent using the multiple INVITEs overlap method....71
7.2.3.2.2 Coding of the INVITE....72
7.2.3.2.2.0 Overview....72
7.2.3.2.2.1 Request-URI and To header field 72
7.2.3.2.2.2 SDP Media Description 73
7.2.3.2.2.3 P-Asserted-Identity, From and Privacy header fields 76
7.2.3.2.2.3A "cpc" URI Parameter in P-Asserted-Identity Header....80
7.2.3.2.2.3B "oli" URI Parameter in P-Asserted-Identity Header 80
7.2.3.2.2.4 Max Forwards header 81
7.2.3.2.2.5 IMS Communication Service Identifier....81

7.2.3.2.2.6 P-Early-Media header field ... 81
7.2.3.2.2.7 PSTN XML elements ... 81
7.2.3.2.2.8 Progress indicator ... 82
7.2.3.2.2.9 P-Access-Network-Info ... 83
7.2.3.2.2A Coding of the INVITE when Number Portability is supported ... 83
7.2.3.2.2A.1 Request-URI and To header field ... 83
7.2.3.2.2A.1.1 Separate Directory Number Addressing Method ... 84
7.2.3.2.2A.1.2 Concatenated Addressing Method ... 84
7.2.3.2.2A.1.3 Separate Network Routing Number Addressing Method ... 85
7.2.3.2.2B Coding of the INVITE for Carrier Routeing ... 85
7.2.3.2.2B.1 Mapping of "cic" in Request-URI ... 85
7.2.3.2.2B.2 Void ... 85
7.2.3.2.2C Coding of INVITE with instance-id in form of IMEI URN ... 85
7.2.3.2.2.10 PSAP Call-back indication ... 86
7.2.3.2.2.11 History-Info header field (optional) ... 86
7.2.3.2.3 Receipt of CONTINUITY ... 86
7.2.3.2.4 Sending of ACM and awaiting answer indication ... 87
7.2.3.2.5 Coding of the ACM ... 89
7.2.3.2.5.0 General ... 89
7.2.3.2.5.1 Backward call indicators ... 89
7.2.3.2.5.2 Optional Backward call indicators ... 91
7.2.3.2.5.3 Access Transport Parameter, Transmission medium used parameter ... 91
7.2.3.2.5.4 Progress indicator ... 92
7.2.3.2.5.5 Cause Value ... 92
7.2.3.2.6 Sending of the Call Progress message (CPG) ... 92
7.2.3.2.6.0 General ... 92
7.2.3.2.6.1 Handling of the progress indicator ... 94
7.2.3.2.7 Coding of the CPG ... 94
7.2.3.2.7.0 General ... 94
7.2.3.2.7.1 Event information ... 95
7.2.3.2.7.2 Access Transport Parameter ... 95
7.2.3.2.7.3 Void ... 95
7.2.3.2.7.4 Handling of Backward Call indicators ... 95
7.2.3.2.7.5 Optional Backward call indicators ... 95
7.2.3.2.7.6 Cause Value ... 95
7.2.3.2.7a Receipt of 200 OK(INVITE) ... 96
7.2.3.2.7b Internal through connection of the bearer path ... 96
7.2.3.2.8 Sending of the Answer Message (ANM) ... 96
7.2.3.2.9 Coding of the ANM ... 96
7.2.3.2.9.1 Backwards Call Indicators ... 96
7.2.3.2.9.2 Access Transport Parameter ... 96
7.2.3.2.9.3 Transmission Medium Used parameter (TMU) ... 97
7.2.3.2.10 Sending of the Connect message (CON) ... 98
7.2.3.2.11 Coding of the CON ... 98
7.2.3.2.11.0 General ... 98
7.2.3.2.11.1 Backward call indicators ... 98
7.2.3.2.11.2 Access Transport Parameter ... 98
7.2.3.2.11.3 Transmission medium used parameter ... 98
7.2.3.2.11A Receipt of a reINVITE request ... 98
7.2.3.2.12 Receipt of Status Codes 4xx, 5xx or 6xx ... 98
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 ... 100
7.2.3 2.13 Receipt of a BYE ... 101
7.2.3.2.14 Receipt of the Release Message ... 101
7.2.3.2.15 Receipt of RSC, GRS or CGB (H/W oriented) ... 101
7.2.3.2.16 Autonomous Release at O-MGCF ... 101
7.2.3.2.17 Special handling of 580 precondition failure received in response to either an INVITE or UPDATE ... 102
7.2.3.2.17.1 580 Precondition failure response to an INVITE ... 102
7.2.3.2.17.2 580 Precondition failure response to an UPDATE within an early dialog ... 102
7.2.3.2.18 Sending of CANCEL ... 102
7.2.3.2.19 Receipt of SIP redirect (3xx) response ... 103

7.2.3.2.20 Sending of INFO for overlap signalling using the in-dialog method 103
7.2.3.2.20.1 General....103
7.2.3.2.20.2 Encoding of the INFO....103
7.2.3.3 Timers 104
7.3 Interworking between CS networks supporting BICC and the IM CN subsystem....104
7.3.1 Services performed by network entities in the control plane 105
7.3.2 Signalling interactions between network entities in the control plane....105
7.3.2.1 Signalling between the SS7 signalling function and MGCF....105
7.3.2.1.1 Signalling from MGCF to SS7 signalling function 105
7.3.2.1.2 Signalling from SS7 signalling function to MGCF 105
7.3.2.1.3 Services offered by STC, SCTP and M3UA 106
7.3.2.1.3.1 Services offer by SCTP....106
7.3.2.1.3.2 Services offered by M3UA 106
7.3.2.1.3.3 Services offered by STC 106
7.3.2.2 Signalling between the MGCF and SIP signalling function....106
7.3.3 SIP-BICC protocol interworking 106
7.3.3.1 Incoming call interworking from SIP to BICC at I-MGCF....106
7.3.3.1.1 Sending of IAM....106
7.3.3.1.2 Coding of IAM 106
7.3.3.1.2.1 Called party number....106
7.3.3.1.2.2 Nature of connection indicators....106
7.3.3.1.2.3 Forward call indicators 107
7.3.3.1.2.4 Calling party's category 107
7.3.3.1.2.4A Originating Line Information....107
7.3.3.1.2.5 Transmission medium requirement....107
7.3.3.1.2.6 Calling party number 107
7.3.3.1.2.7 Generic number....107
7.3.3.1.2.8 User service information....107
7.3.3.1.2.9 Hop counter (National option)....107
7.3.3.1.2.10 Location Number 107
7.3.3.1.2.11 Support of ICS call 107
7.3.3.1.2.12 UID capability indicators (National option) 107
7.3.3.1.2A Coding of the IAM when Number Portability is supported....107
7.3.3.1.2B Coding of the IAM for Carrier Routeing....107
7.3.3.1.2.13 Called IN number and original called IN number (optional) 107
7.3.3.1.3 Sending of COT....108
7.3.3.1.3A Sending of SAM....108
7.3.3.1.4 Sending of 180 Ringing....108
7.3.3.1.4A Sending of 183 Session Progress for early media scenarios....108
7.3.3.1.4B Sending of 181 Call is being forwarded 108
7.3.3.1.4C Sending of 183 Session Progress for overlap signalling using the in-dialog method....108
7.3.3.1.4D Sending of 183 Session Progress to carry ISUP Cause 108
7.3.3.1.4E Sending of 183 Session Progress for ICS call 108
7.3.3.1.5 Sending of the 200 OK (INVITE)....109
7.3.3.1.6 Sending of the Release message (REL)....109
7.3.3.1.7 Coding of the REL....109
7.3.3.1.8 Receipt of the Release Message 109
7.3.3.1.9 Receipt of RSC, GRS or CGB (H/W oriented) 109
7.3.3.1.9a Receipt of REFER....109
7.3.3.1.9b Autonomous Release at I-MGCF 109
7.3.3.1.10 Internal through connection of the bearer path....109
7.3.3.1.11 Out of Band DTMF 109
7.3.3.2 Outgoing Call Interworking from BICC to SIP at O-MGCF 109
7.3.3.2.1 Sending of INVITE 109
7.3.3.2.1a Sending of INVITE without determining the end of address signalling....110
7.3.3.2.2 Coding of the INVITE....110
7.3.3.2.2.1 Request-URI and To header field 110
7.3.3.2.2.2 SDP Media Description 110
7.3.3.2.2.3 P-Asserted-Identity and privacy header fields 110
7.3.3.2.2.3A "cpc" URI Parameter in P-Asserted-Identity Header....110
7.3.3.2.2.3B "oli" URI Parameter in P-Asserted-Identity Header 110
7.3.3.2.2.4 Max Forwards header 110

7.3.3.2.2.5 IMS Communication Service Identifier 111
7.3.3.2.2.6 P-Access-Network-Info 111
7.3.3.2.2A Coding of the INVITE when number portability is supported 111
7.3.3.2.2B Coding of the INVITE for Carrier Routeing 111
7.3.3.2.2C Coding of INVITE with instance-id in form of IMEI URN 111
7.3.3.2.2.7 PSAP Call-back indication 111
7.3.3.2.2.8 History-Info header field (optional) 111
7.3.3.2.3 Sending of UPDATE 111
7.3.3.2.4 Sending of ACM and Awaiting Answer indication 112
7.3.3.2.5 Coding of the ACM 112
7.3.3.2.5.1 Backward call indicators 112
7.3.3.2.6 Sending of the Call Progress message (CPG) 112
7.3.3.2.7 Coding of the CPG 112
7.3.3.2.7.1 Event information 112
7.3.3.2.7.2 Optional Backward call indicators 112
7.3.3.2.7a Receipt of 200 OK (INVITE) 112
7.3.3.2.7b Internal through connection of the bearer path 112
7.3.3.2.8 Sending of the Answer Message (ANM) 112
7.3.3.2.9 Coding of the ANM 112
7.3.3.2.10 Sending of the Connect message (CON) 112
7.3.3.2.11 Coding of the CON 112
7.3.3.2.11.1 Void 113
7.3.3.2.11A Receipt of re-INVITE requests 113
7.3.3.2.12 Receipt of Status Codes 4xx, 5xx or 6xx 113
7.3 3.2.13 Receipt of a BYE 113
7.3.3.2.14 Receipt of the Release Message 113
7.3.3.2.15 Receipt of RSC, GRS or CGB (H/W oriented) 113
7.3.3.2.16 Out of Band DTMF 113
7.3.3.2.17 Sending of CANCEL 113
7.3.3.2.18 Autonomous Release at O-MGCF 113
7.3.3.2.19 Special handling of 580 precondition failure received in response to either an INVITE or UPDATE 113
7.3.3.2.20 Receipt of SIP redirect (3xx) response 113
7.3.3.2.21 Sending of INFO for overlap signalling using the in-dialog method 113
7.3.3.3 Timers 114
7.4 Supplementary Services 114
7.4.1 Calling line identification presentation/restriction (CLIP/CLIR) 114
7.4.2 Connected line presentation and restriction (COLP/COLR) 114
7.4.2.0 General 114
7.4.2.1 Incoming Call Interworking from SIP to BICC/ISUP at the I-MGCF 114
7.4.2.1.1 INVITE to IAM interworking (SIP to ISUP/BICC calls) 114
7.4.2.1.2 ANM/CON to 200 OK (INVITE) 114
7.4.2.2 Outgoing Call Interworking from BICC/ISUP to SIP at O-MGCF 115
7.4.2.2.1 IAM to INVITE interworking (ISUP to SIP calls) 115
7.4.2.2.2 1XX to ANM or CON interworking 116
7.4.2.2.3 200 OK (INVITE) to ANM/CON interworking 116
7.4.3 Direct Dialling In (DDI) 117
7.4.4 Malicious call identification 117
7.4.5 Subaddressing (SUB) 117
7.4.5.1 General 117
7.4.5.2 Interworking at I-MGCF 118
7.4.5.3 Interworking at O-MGCF 119
7.4.6 Call Forwarding Busy (CFB)/ Call Forwarding No Reply (CFNR) / Call Forwarding Unconditional (CFU) / Call Deflection (CD) 120
7.4.6.1 General 120
7.4.6.2 Interworking at the O-MGCF 120
7.4.6.2.1 General 120
7.4.6.2.2 Interworking SIP to ISUP 120
7.4.6.2.3 Interworking ISUP to SIP 124
7.4.6.3 Interworking at the I-MGCF 126
7.4.6.3.1 General 126
7.4.6.3.2 Interworking from SIP to ISUP 126

7.4.6.3.3 Interworking from ISUP to SIP 129
7.4.7 Void 133
7.4.8 Explicit Call Transfer (ECT) 133
7.4.9 Call Waiting....133
7.4.10 Call Hold....133
7.4.10.1 Session hold initiated from the IM CN subsystem side....133
7.4.10.2 Session hold initiated from the CS network side....134
7.4.11 Call Completion on busy subscriber....137
7.4.12 Completion of Calls on No Reply (CCNR) 137
7.4.13 Terminal Portability (TP)....137
7.4.14 Conference calling (CONF) / Three-Party Service (3PTY)....137
7.4.15 Void 138
7.4.16 Closed User Group (CUG) 138
7.4.17 Multi-Level Precedence and Pre-emption (MLPP)....138
7.4.18 Global Virtual Network Service (GVNS)....138
7.4.19 International telecommunication charge card (ITCC) 138
7.4.20 Reverse charging (REV)....138
7.4.21 User-to-User Signalling (UUS)....138
7.4.21.0 General....138
7.4.21.1 User-to-User Signalling (UUS) service 1 (implicit)....138
7.4.21.1.0 General 138
7.4.21.1.1 Void....139
7.4.21.1.2 User-to-user information Interworking from SIP to ISUP 139
7.4.21.1.3 User-to-user information Interworking from ISUP to SIP 139
7.4.21.2 User-to-User Signalling (UUS) service 1 (explicit) 139
7.4.21.3 User-to-User Signalling (UUS) service 2 (explicit) 139
7.4.21.4 User-to-User Signalling (UUS) service 3 (explicit) 139
7.4.22 Multiple Subscriber Number (MSN) 140
7.4.23 Anonymous Call rejection 140
7.4.23.0 General....140
7.4.23.1 ISUP-SIP protocol interworking at the I-MGCF....140
7.4.23.2 SIP-ISUP protocol interworking at the O-MGCF....140
7.4.24 Customized Alerting Tones (CAT) in the 3GPP CS domain....140
7.5 IMS Supplementary Services 140
7.5.1 Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR)....140
7.5.2 Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR)....140
7.5.2.1 General....140
7.5.2.2 Interworking at the O-MGCF....140
7.5.2.3 Interworking at the I-MGCF 142
7.5.2.4 Timer....143
7.5.3 void 143
7.5.4 Communication Diversion (CDIV)....143
7.5.4.1 General....143
7.5.4.2 Interworking at the O-MGCF....144
7.5.4.2.1 General 144
7.5.4.2.2 Call Diversion within the ISUP Network appeared....149
7.5.4.3 Interworking at the I-MGCF 151
7.5.5 Communication Hold (HOLD)....159
7.5.6 Conference call (CONF)....159
7.5.6.1 General....159
7.5.6.2 Subscribing for the conference event package 159
7.5.6.3 Interworking the notification....160
7.5.7 Anonymous Communication Rejection (ACR) and Communication Barring (CB)....160
7.5.8 Message Waiting Indication (MWI) 161
7.5.9 Malicious Communication Identification (MCID) 161
7.5.9.0 General....161
7.5.9.1 Interworking at the O-MGCF....161
7.5.9.1.0 General 161
7.5.9.1.1 Interworking of the MCID XML Request schema with the ISUP MCID request indicators 161
7.5.9.1.2 Interworking of the ISUP MCID response indicators with the MCID XML Response schema ...162
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....162

7.5.9.1.4 Interworking of the ISUP Generic Number in an Identification Response with the GenericNumber within the MCID XML Response schema 162
7.5.9.2 Interworking at the I-MGCF 162
7.5.9.2.1 General 162
7.5.9.2.2 Interworking of identification Request 162
7.5.9.2.3 Interworking of identification Response 163
7.5.10 Closed User Group (CUG) 163
7.5.10.0 General 163
7.5.10.1 Interworking at the I-MGCF 163
7.5.10.2 Interworking at the O-MGCF 164
7.5.11 CCBS/CCNR 165
7.5.11.0 General 165
7.5.11.1 Interworking at the I-MGCF 165
7.5.11.2 Interworking at the O-MGCF 167
7.5.12 Communication Waiting (CW) 169
7.5.12.0 General 169
7.5.12.1 Interworking at the I-MGCF 169
7.5.12.2 Interworking at the O-MGCF 169
7.5.13 Customized Alerting Tones (CAT) 169
7.5.13.1 General 169
7.5.13.2 Early session model 169
7.5.13.2.1 Interworking at the O-MGCF 169
7.5.13.2.2 MGCF - IM-MGW Interaction for CAT service 170
7.5.13.2.2.1 General 170
7.5.13.2.2.2 IM CN subsystem side Early Session establishment 170
7.5.13.2.2.3 IM CN subsystem side normal Session establishment 170
7.5.13.2.2.4 Called party alerting 171
7.5.13.2.2.5 Message sequence chart 171
7.5.13.3 Forking model 174
7.5.13.4 Gateway model 174
8 User plane interworking 174
8.1 Interworking between IM CN subsystem and bearer independent CS network 174
8.1.1 Transcoder-less Mb to Nb Interworking 175
8.1.1.1 Initialisation 175
8.1.1.2 Time alignment 175
8.1.1.3 Rate control 176
8.1.1.3.1 General 176
8.1.1.3.2 Rate control for AMR and AMR-WB codecs 176
8.1.1.3.3 Rate and Mode control for the EVS codec 176
8.1.1.3.4 Interworking of rate control between compatible AMR-WB and EVS codec configurations 177
8.1.1.3.5 Interworking between rate control and RTCP-APP-CMR messages for MTSI 177
8.1.1.4 Frame quality indication 177
8.1.1.5 Framing 178
8.1.1.6 Transcoding 178
8.1.1.7 Discontinuous transmission 178
8.1.1.8 Timing and sequence information 178
8.2 Interworking between IM CN subsystem and TDM-based CS network 178
8.3 Transcoding requirements 179
8.4 Diffserv code point requirements 179
8.5 DTMF handling 179
9 MGCF - IM-MGW Interaction 180
9.1 Overview 180
9.2 Mn signalling interactions 180
9.2.1 Network model 180
9.2.2 Basic IM CN subsystem originated session 180
9.2.2.1 BICC forward bearer establishment 180
9.2.2.1.1 IM-MGW selection 180
9.2.2.1.2 CS network side bearer establishment 181
9.2.2.1.3 IM CN subsystem side termination reservation 181
9.2.2.1.4 IM CN subsystem side session establishment 181

9.2.2.1.5 Through-connection 182
9.2.2.1.6 Codec handling....182
9.2.2.1.7 Failure handling in MGCF 182
9.2.2.1.8 Message sequence chart....182
9.2.2.2 BICC backward bearer establishment....183
9.2.2.2.1 IM-MGW selection 183
9.2.2.2.2 IM CN subsystem side termination reservation....184
9.2.2.2.3 IM CN subsystem side session establishment 184
9.2.2.2.4 CS network side bearer establishment....184
9.2.2.2.5 Through-connection 185
9.2.2.2.6 Codec handling....185
9.2.2.2.7 Failure handling in MGCF 185
9.2.2.2.8 Message sequence chart....185
9.2.2.3 ISUP....186
9.2.2.3.1 IM-MGW selection 186
9.2.2.3.2 IM CN subsystem side termination reservation....186
9.2.2.3.3 IM CN subsystem side session establishment 187
9.2.2.3.4 CS network side circuit reservation....187
9.2.2.3.5 Through-connection 187
9.2.2.3.6 Continuity check....188
9.2.2.3.7 Codec handling....188
9.2.2.3.8 Voice processing function 188
9.2.2.3.9 Failure handling in MGCF 188
9.2.2.3.10 Message sequence chart....188
9.2.3 Basic CS network originated session....190
9.2.3.1 BICC forward bearer establishment....190
9.2.3.1.1 IM-MGW selection 190
9.2.3.1.2 IM CN subsystem side termination reservation....190
9.2.3.1.3 IM CN subsystem side session establishment 190
9.2.3.1.4 CS network side bearer establishment....190
9.2.3.1.5 Called party alerting 191
9.2.3.1.6 Called party answer 191
9.2.3.1.7 Through-Connection....191
9.2.3.1.8 Codec handling....191
9.2.3.1.9 Failure handling in MGCF 191
9.2.3.1.10 Message sequence chart....191
9.2.3.2 BICC Backward bearer establishment 193
9.2.3.2.1 IM-MGW selection 193
9.2.3.2.2 CS network side bearer establishment....193
9.2.3.2.3 IM CN subsystem side termination reservation....193
9.2.3.2.4 IM CN subsystem side session establishment 194
9.2.3.2.5 Called party alerting 194
9.2.3.2.6 Called party answer 194
9.2.3.2.7 Through-Connection....194
9.2.3.2.8 Codec handling....194
9.2.3.2.9 Failure handling in MGCF 195
9.2.3.2.10 Message sequence chart....195
9.2.3.3 ISUP....197
9.2.3.3.1 IM-MGW selection 197
9.2.3.3.2 CS network side circuit reservation....197
9.2.3.3.3 IM CN subsystem side termination reservation....197
9.2.3.3.4 IM CN subsystem side session establishment 197
9.2.3.3.5 Called party alerting 198
9.2.3.3.6 Called party answer 198
9.2.3.3.7 Through-Connection....198
9.2.3.3.8 Continuity Check....198
9.2.3.3.9 Codec handling....198
9.2.3.3.10 Voice Processing function....199
9.2.3.3.11 Failure handling in MGCF 199
9.2.3.3.12 Message sequence chart....199
9.2.3.4 Handling of Forking....201
9.2.3.4.1 Detection of Forking....201

9.2.3.4.2 IM CN subsystem side session establishment 201
9.2.3.4.3 IM CN subsystem side session establishment completion 202
9.2.3.4.4 Message sequence chart....202
9.2.4 Session release initiated from IM CN subsystem side 205
9.2.4.1 BICC 205
9.2.4.1.1 Session release in the IM CN subsystem side....205
9.2.4.1.2 Session release in the CS network side....205
9.2.4.1.3 Message sequence chart....205
9.2.4.2 ISUP....206
9.2.4.2.1 Session release in the IM CN subsystem side....206
9.2.4.2.2 Session release in the CS network side....206
9.2.4.2.3 Message sequence chart....206
9.2.5 Session release initiated from CS network side 206
9.2.5.1 BICC 206
9.2.5.1.1 Session release in the CS network side....206
9.2.5.1.2 Session release in the IM CN subsystem side....207
9.2.5.1.3 Message sequence chart....207
9.2.5.2 ISUP....207
9.2.5.2.1 Session release in the CS network side....207
9.2.5.2.2 Session release in the IM CN subsystem side....207
9.2.5.2.3 Message sequence chart....208
9.2.6 Session release initiated by MGCF....208
9.2.6.1 BICC 208
9.2.6.1.1 Session release in the CS network side....208
9.2.6.1.2 Session release in the IM CN subsystem side....208
9.2.6.1.3 Message sequence chart....208
9.2.6.2 ISUP....209
9.2.6.2.1 Session release in the CS network side....209
9.2.6.2.2 Session release in the IM CN subsystem side....209
9.2.6.2.3 Message sequence chart....209
9.2.7 Session release initiated by IM-MGW....210
9.2.7.1 BICC 210
9.2.7.1.1 Session release in the CS network side....210
9.2.7.1.2 Session release in the IM CN subsystem side....210
9.2.7.1.3 Message sequence chart....210
9.2.7.2 ISUP....211
9.2.7.2.1 Session release in the CS network side....211
9.2.7.2.2 Session release in the IM CN subsystem side....211
9.2.7.2.3 Message sequence chart....212
9.2.8 Handling of RTP telephone events 212
9.2.8.1 Sending DTMF digits out-of-band to CS CN (BICC)....213
9.2.8.2 Sending and receiving DTMF digits inband to/from CS CN (ISUP or BICC) 215
9.2.8.3 Receiving DTMF digits out-of-band from CS CN (BICC)....216
9.2.9 Session hold initiated from IM CN subsystem 217
9.2.10 Session hold initiated from CS network 219
9.2.11 Explicit Congestion Notification Support....221
9.2.11.1 General....221
9.2.11.1a Incoming Call Interworking from SIP to ISUP/BICC at I-MGCF....221
9.2.11.1b Outgoing Call Interworking from ISUP/BICC to SIP at O-MGCF 222
9.2.11.1c Detection of ECN failures by IM-MGW....222
9.2.11.2 Message sequence chart 222
9.2.12 Interactive Connectivity Establishment Support....223
9.2.12.1 General....223
9.2.12.2 ICE lite....223
9.2.12.3 Full ICE....224
9.2.12.4 Procedures for Interactive Connectivity Establishment (ICE) 226
9.2.12.4.1 ICE lite 226
9.2.12.4.2 Full ICE 227
9.2.12.4.3 Connectivity check result notification (full ICE) 228
9.2.12.4.4 New peer reflexive candidate notification (full ICE) 228
9.2.13 Codec Parameters Handling....229
9.2.13.1 Handling of common codec parameters....229

9.2.13.2 Handling of EVS speech codec parameters when interworking with a different codec 230
9.2.13.3 Handling of EVS speech codec parameters when the EVS codec is used end-to-end 239
9.2.14 SDP Capability Negotiation (SDPCapNeg) 246
9.3 Mn Signalling procedures 246
9.3.1 Procedures related to terminations towards the IM CN Subsystem 246
9.3.1.1 Reserve IMS connection point 247
9.3.1.2 Configure IMS resources 248
9.3.1.3 Reserve IMS Connection point and configure remote resources 250
9.3.1.4 Release IMS termination 252
9.3.1.5 Detect IMS RTP Tel event 252
9.3.1.6 Notify IMS RTP Tel event 252
9.3.1.7 Void 253
9.3.1.8 Send IMS RTP Tel event 253
9.3.1.9 Stop IMS RTP Tel event 253
9.3.1.10 Termination heartbeat indication 253
9.3.1.11 IMS Bearer Released 253
9.3.1.12 End IMS RTP Tel event 253
9.3.1.13 IMS Send Tone 253
9.3.1.14 IMS Stop Tone 253
9.3.1.15 IMS Tone Completed 254
9.3.1.16 ECN Failure Indication 254
9.3.1.17 ICE Connectivity Check Result Notification 254
9.3.1.18 ICE New Peer Reflexive Candidate Notification 255
9.3.2 Procedures related to a termination towards an ISUP network 255
9.3.2.1 Reserve TDM circuit 255
9.3.2.2 Change TDM through-connection 256
9.3.2.3 Activate TDM voice-processing function 256
9.3.2.4 Send TDM tone 256
9.3.2.5 Stop TDM tone 256
9.3.2.6 Play TDM announcement 256
9.3.2.7 TDM announcement completed 256
9.3.2.8 Stop TDM announcement 256
9.3.2.9 Continuity check 256
9.3.2.10 Continuity check verify 257
9.3.2.11 Continuity check response 257
9.3.2.12 Release TDM termination 257
9.3.2.13 Termination Out-of-Service 258
9.3.2.14 Termination heartbeat indication 258
9.3.2.15 Bearer Released 258
9.3.2.16 TDM tone completed 258
9.3.3 Procedures related to a termination towards a BICC network 258
9.3.4 Non-call related procedures 259
9.3.5 Multiple IP Realms 260
9.4 Multimedia Priority Service (MPS) Support 261
9.4.1 General 261
9.4.2 IM-MGW Resource Congestion in ADD response, request is queued 261
9.4.3 IM-MGW Resource Congestion in ADD response, MGCF seizes new IM-MGW 262
9.4.4 IM-MGW Priority Resource Allocation 262
9.4.5 IM-MGW Priority User Data marking 263

Annex A (informative): Void 264

Annex B (normative): Codec Negotiation between a BICC CS network and the IM CN subsystem 265

B.1 Introduction 265

B.2 Control plane interworking 265
B.2.1 Incoming call interworking from SIP to BICC at I-MGCF 265
B.2.1.1 Sending of IAM 265
B.2.1.2 Sending of SDP answer 265
B.2.2 Outgoing call interworking from BICC to SIP at O-MGCF 265
B.2.2.1 Sending of INVITE 265

B.2.2.2 Responding to serving node initiating codec negotiation 266
B.2.3 Mid-call interworking from SIP to BICC at I-MGCF or O-MGCF 266
B.2.3.1 Receipt of SDP offer 266
B.2.3.2 Generating SDP answer 267
B.2.4 Mid-call interworking from BICC to SIP at I-MGCF or O-MGCF 267
B.2.4.1 Receipt of mid-call codec negotiation request 267
B.2.4.2 Responding to serving node initiating mid-call codec negotiation 267
B.2.4.3 Receipt of codec modification request 268
B.2.5 Codec parameter translation between BICC CS network and the IM CN subsystem 268
B.2.5.1 Codec parameters for 3GPP AMR-NB codecs 269
B.2.5.2 Codec parameters for 3GPP AMR-WB codecs 270
B.2.5.3 Codec parameters for 3GPP non-AMR codecs 272
B.2.5.4 Codec parameters for ITU-T codecs 272
B.2.5.5 Codec parameters for 3GPP EVS codec 273

B.3 MGCF - IM-MGW interaction during interworking of codec negotiation 276
B.3.1 Basic IM CN subsystem originated session 276
B.3.1.1 BICC forward bearer establishment 276
B.3.1.1.1 IM-MGW selection 276
B.3.1.1.2 CS network side bearer establishment 276
B.3.1.1.3 IM CN subsystem side session establishment 276
B.3.1.1.4 Through-connection 277
B.3.1.1.5 Codec handling 277
B.3.1.1.6 Failure handling in MGCF 277
B.3.1.1.7 Message sequence chart 277
B.3.2 Basic CS network originated session 279
B.3.2.1 BICC forward bearer establishment 279
B.3.2.1.1 IM-MGW selection 279
B.3.2.1.2 IM CN subsystem side termination reservation 279
B.3.2.1.3 IM CN subsystem side session establishment 279
B.3.2.1.4 CS network side bearer establishment 279
B.3.2.1.5 Called party alerting 280
B.3.2.1.6 Called party answer 280
B.3.2.1.7 Through-Connection 280
B.3.2.1.8 Codec handling 280
B.3.2.1.9 Failure handling in MGCF 280
B.3.2.1.10 Message sequence chart 280
B.3.3 CS network initiated mid-call codec negotiation 283
B.3.4 IM CN subsystem initiated mid-call codec negotiation 284

Annex C (normative): Interworking of CPC parameter 285

C.1 Interworking SIP to ISUP 285

C.2 Interworking ISUP to SIP 286

Annex D: Void 287

Annex E (normative): Multimedia interworking between the IP Multimedia Core Network (CN) Subsystem (IMS) and Circuit Switched (CS) networks 288

E.1 Basic Multimedia calls interworking between the IMS and CS Networks scenarios 288

E.2 Control plane interworking 288
E.2.1 General 288
E.2.2 Functionalities required in the MGCF for multimedia calls support 289
E.2.3 IM CN subsystem originated session 289
E.2.3.1 Preconditions used at IMS side 289
E.2.3.1.1 Interactions between H.245 or MONA and SIP/SDP 289
E.2.3.2 Preconditions not used at IMS side 291
E.2.3.2.1 Interactions between H.245 or MONA and SIP/SDP 291
E.2.3.3 Fallback to speech at session establishment 293
E.2.4 CS network originated session 293
E.2.4.1 Interactions between SIP/SDP and H.245 or MONA 293

E.2.4.1.1 Normal Call setup 293
E.2.4.1.2 Call setup if multimedia call can not be recognized in an unambiguous manner....................295
E.2.4.1.3 Fallback to speech during call setup 295
E.2.4.2 CS originated - IM CN transit - CS terminated 296
E.2.5 Service change....................297
E.2.5.2.1 SCUDIF 297
E.2.5.2.1.0 General 297
E.2.5.2.1.1 IM CN subsystem originated change....................297
E.2.5.2.1.1.1 Change from multimedia to speech 297
E.2.5.2.1.1.2 Change from speech to multimedia 298
E.2.5.2.1.2 CS network originated change....................299
E.2.5.2.1.2.1 Change from multimedia to speech 299
E.2.5.2.1.2.2 Change from speech to multimedia 299
E.2.5.2.2 Non-SCUDIF case (ISUP or BICC without SCUDIF)....................300
E.2.5.2.2.1 Change from multimedia to audio....................300
E.2.5.2.2.2 Change from speech to multimedia 301
E.2.6 Call release 301
E.2.6.1 Call release initiated from the IM CN subsystem side....................301
E.2.6.2 Call release initiated from the CS network side....................302
E.2.6.3 Call release initiated from the interworking node....................303

E.3 User plane interworking....................304
E.3.1 Functionalities required in the IM-MGW for multimedia calls support....................304

E.4 MGCF and IM-MGW interactions....................304
E.4.1 Introduction....................304
E.4.2 Mn signalling interactions....................305
E.4.2.1 Introduction....................305
E.4.2.2 H.248 Context Model 305
E.4.2.3 Transport of H.245 messages between the MGCF and IM-MGW 305
E.4.2.3.1 General....................305
E.4.2.3.2 Transport from MGCF to IM-MGW....................306
E.4.2.3.3 Transport from IM-MGW to MGCF....................306
E.4.2.4 Call establishment procedure....................306
E.4.2.5 Handling of H.245 indication message....................309
E.4.2.5.1 Overview....................309
E.4.2.5.2 Function Not Understood / Function Not Supported message....................309
E.4.2.5.3 User Input Indication message 310
E.4.2.6 Handling of H.245 Command message 310
E.4.2.6.1 Overview....................310
E.4.2.6.2 Flow control command 310
E.4.2.6.3 End Session Command 311
E.4.2.6.4 videoFastUpdatePicture 311
E.4.2.7 Mn Signalling Interactions to support the Media Oriented Negotiation Acceleration (MONA)....................312
E.4.2.7.1 Overview....................312
E.4.2.7.2 Mn Interactions for MONA preference messages....................314
E.4.2.7.3 Mn Interactions for MONA MPCs....................316
E.4.2.7.4 Mn Interactions for MONA SPCs....................318
E.4.2.7.4.1 General 318
E.4.2.7.4.2 Transport from MGCF to IM-MGW 318
E.4.2.7.4.3 Transport from IM-MGW to MGCF 319
E.4.2.7.4.4 Termination of SPC procedure....................320
E.4.2.7.5 Mn Interactions for fallback from MONA procedures to standard H.324 setup....................321
E.4.2.8 Interworking between RTCP messages and H.245 messages....................322
E.4.2.8.1 Overview....................322
E.4.2.8.2 IM CN subsystem originated RTCP messages....................322
E.4.2.8.3 CS network originated H.245 messages....................323
E.4.3 Mn Signalling procedures 324
E.4.3.1 Overview 324
E.4.3.2 Add Multiplex Termination....................325
E.4.3.3 Configure Multiplex Termination....................326
E.4.3.4 Signal H245 Message 327

E.4.3.5 Notify H245 Message 327
E.4.3.6 Notify MONA Preference Reception....328
E.4.3.7 Notify MONA Preference Completed 328
E.4.3.8 Signal SPC....329
E.4.3.9 Notify SPC....329
E.4.3.10 Notify MPC 330
E.4.3.11 Notify Detection of Legacy Interworking....330
E.4.3.12 Request RTCP-Interworking 331
E.4.3.13 Notify RTCP-Interworking....331
E.4.3.14 Signal-H.245-Interworking....332
E.4.3.15 Stop MPC 332
E.4.3.16 Stop SPC....332
E.4.3.17 Stop MONA Negotiation....333

Annex F (normative): PSTN XML Scheme 335

F.1 Scope....335

F.2 MIME type 335

F.3 XML Schema definition....335

F.4 IANA registration template....340

Annex G (informative): Void 343

Annex H (normative): Interworking of Originating Line Information (OLI) parameter (network option)....344

H.1 Interworking SIP to ISUP....344

H.2 Interworking ISUP to SIP....344

Annex I (normative): GTT interworking between the IP Multimedia Core Network (CN) Subsystem (IMS) and Circuit Switched (CS) networks 345

I.1 Overview of GTT interworking between the IMS and Circuit Switched (CS) networks 345

I.2 Control plane interworking 346
I.2.1 General 346
I.2.2 Functionalities required in the MGCF for GTT calls support 346
I.2.3 IM CN subsystem originated session 346
I.2.3.1 Initial INVITE with an SDP offer including a text media line 346
I.2.4 CS network originated session 347
I.2.4.1 General....347
I.2.4.2 Initial INVITE with an SDP offer including audio only....347
I.2.5 Subsequent SDP offer/answer exchange adding text to an audio session 348

I.3 User plane interworking....348
I.3.1 Functionalities required in the IM-MGW for GTT support 348
I.3.2 Monitoring of text/modem signals on the CS side 348
I.3.3 Multiplexing between the CS and IMS streams 349
I.3.4 Conversion between text/modem and Real-Time Text over RTP 349

I.4 MGCF and IM-MGW interactions....349
I.4.1 Introduction 349
I.4.2 Mn signalling interactions 349
I.4.2.1 Introduction....349
I.4.2.2 H.248 Context Model 350
I.4.2.3 Specific Mn signalling for GTT....350
I.4.2.4 IM CN subsystem originated session between the MGCF and IM-MGW 351
I.4.2.4.1 Initial INVITE with an SDP offer including Real-Time Text....351
I.4.2.5 CS network originated session....353
I.4.2.5.1 Initial INVITE with an SDP offer only including audio 353
I.4.3 Mn Signalling procedures 354
I.4.3.1 Overview 354

Annex J (informative): Call Flow for Customized Alerting Tone (CAT) interworking between CS network and IMS ... 355

J.1 Introduction ... 355

J.2 IMS CAT provided by the terminating IMS user using gateway model towards user served in CS networks ... 355

J.2.1 Callflow without using precondition mechanism ... 356

Annex K (normative): T.38 interworking between the IP Multimedia Core Network (CN) Subsystem (IMS) and Circuit Switched (CS) networks ... 358

K.1 General ... 358

K.2 Recommend T.38 configuration ... 358

K.3 Handling T.38 SDP attributes at the I-MGCF ... 359

K.4 Handling T.38 SDP attributes at the O-MGCF ... 360

K.5 Applicable T.38 SDP attributes at the Mn Interface ... 361

Annex L (normative): Interworking of called IN number and original called IN number parameters ... 362

L.0 General ... 362

L.1 Interworking SIP to ISUP ... 362

L.2 Interworking ISUP to SIP ... 363

Annex M (informative): Change history ... 367

History ... 368

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 26.235 [80] and 3GPP TS 26.236 [32], 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.761to 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".