Technical Technical Specification of the SIP (Gm) interface between the User Equipment (UE) and the NGN platform of Deutsche Telekom

1 TR 114

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Contents

Foreword

This Technical Specification (Technische Richtlinie, TR) has been produced by the department FMED-321 of Deutsche Telekom Netzproduktion GmbH, Fixed Mobile Engineering Deutschland (in the following named as Deutsche Telekom) and contains the description of the SIP (Gm) interface between an User Equipment (UE) and the NGN platform of Deutsche Telekom.

Annex A of the present document is not longer Valid due to the fact that all changes of TISPAN documents were incorporated into the 3GPP Specifications. Therefore only Annex B of the present document is now applicable.

Annex B of the present document is a delta specification based on the 3GPP Technical Specification TS 24.229 Release 8 [21] (endorsement).

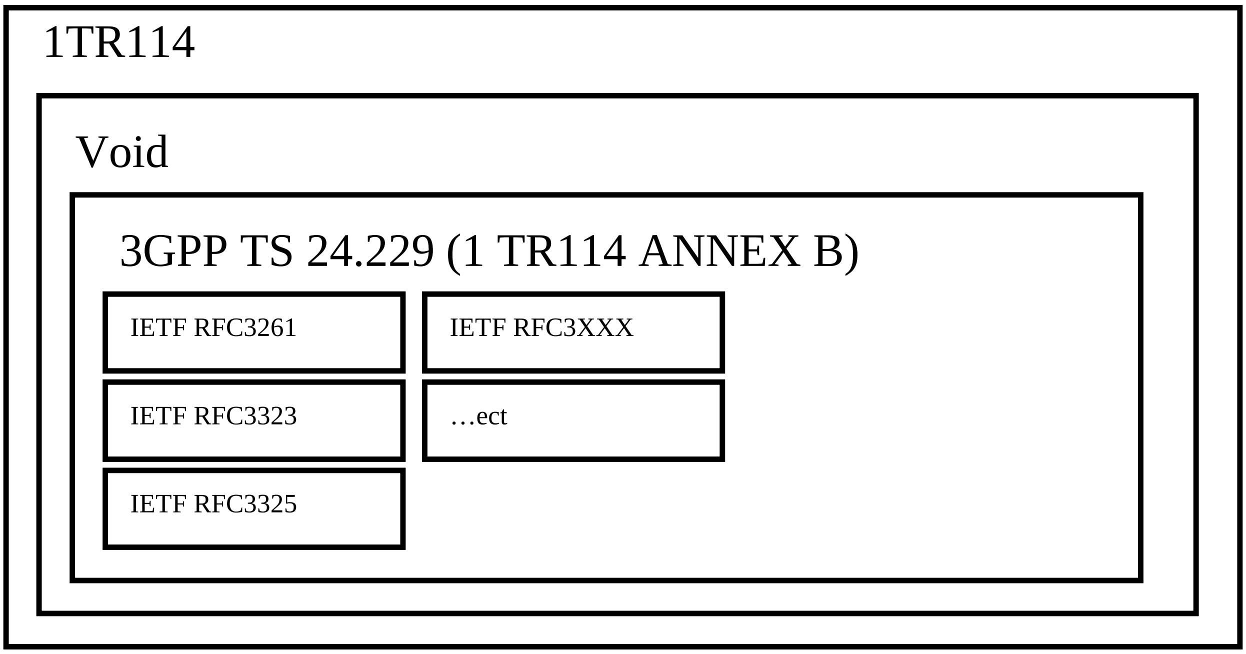
This TR contains the current status of the SIP (Gm) interface which will be supported by the NGN platform of Deutsche Telekom.

Modifications in the main body as well as in the annexes of this document can not be excluded at this point of time due to the still ongoing work on the referenced standards (e.g. 3GPP, ETSI) and some open decisions concerning the supported options.

The present document describes the final NGN platform of Deutsche Telekom; deviations to the currently provided solution of the NGN platform of Deutsche Telekom are possible (e.g. not yet realized service features).

The figure below shows the principle of endorsement used within this document.

Technische Richtlinie 1 TR 114



1 Scope

The present Technical Specification (TR) is applicable to the SIP (Gm) interface between a User Equipment (UE) and the Next Generation Network (NGN) of Deutsche Telekom according to the AGB [1] of Deutsche Telekom.

The present document does not describe the physical characteristics and transmission requirements via the subscriber line.

A possible physical access is e.g. an xDSL interface provided by Deutsche Telekom which is described in the technical specification 1TR112 [2].

Reference points of 1TR114

Figure 1-1 depicts the scope of the relevant technical specifications.

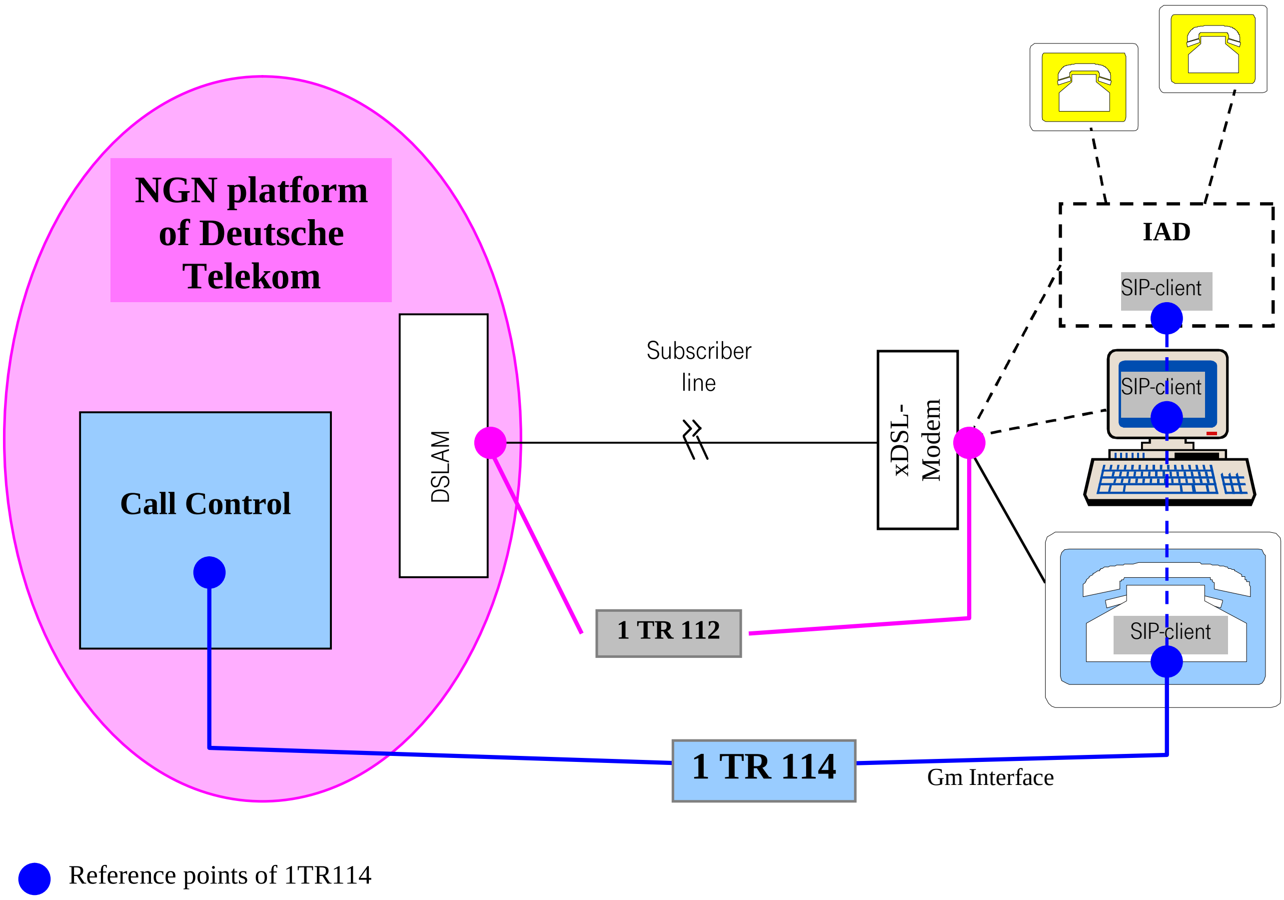


Figure 1-1: Scope of the relevant technical specifications

2 References

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

1. References are either specific (identified by date of publication and/or edition number or version number) or non specific.
   - For a specific reference, subsequent revisions do not apply.
   - For a non-specific reference, the latest version including amendments, errata and corrigenda applies.
2. Date of publication in square brackets [ ] refer just to the last known version while this document was in revision.

[1] AGB: Allgemeine Geschäftsbedingungen der Deutschen Telekom
(see: www.telekom.de/agb)

[2] T-Home 1TR112: Technical Specification of the U-Interfaces of xDSL Systems in the network of T-Home

[3] DT 1TR126: Technical Specification for SIP User Equipments (UE) providing IMS simulation services via analogue (POTS) interfaces (POTS/SIP interworking) using the NGN platform of Deutsche Telekom

[4] DT 1TR127: Technical Specification for SIP User Equipments (UE) providing IMS simulation services via ISDN (DSS1) interfaces (ISDN/SIP interworking) using the NGN platform of Deutsche Telekom

[5] ETSI TS 102 144 V1.1.1 (2003-05): Services and Protocols for Advanced Networks (SPAN);
MTP/SCCP/SSCOP and SIGTRAN (Transport of SS7 over IP);
Stream Control Transmission Protocol (SCTP)
[Endorsement of RFC 2960 and RFC 3309, modified]

[6] 3GPP TS 24.606: "Message Waiting Indication (MWI) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification"(Release 11).

[7] 3GPP TS 24.607: "Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification"(Release 11).

[8] 3GPP TS 24.608: "Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification"(Release 11).

[9] 3GPP TS 24.611: "Anonymous Communication Rejection (ACR) and Communication Barring (CB) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification"(Release 11).

[10] 3GPP TS 24.647: "Messaging using the IP Multimedia (IM) Core Network (CN) subsystem;
Stage 3"(Release 11).

[11] Void

[12] 3GPP TS 24.604: "Communication Diversion (CDIV); Protocol specification using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification "(Release 11).

[13] 3GPP TS 24.608: "Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification"(Release 11).

[14] 3GPP TS 24. 616 V11.1.0 (2012-12): 3rd Generation Partnership Project;Technical Specification Group Core Network and Terminals;Malicious Communication Identification (MCID) using IP Multimedia (IM) Core Network (CN) subsystem;Protocol specification (Release 11)

[15] 3GPP TS 24.629: "Explicit Communication Transfer (ECT) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification"(Release 11).

[16] 3GPP TS 24.605: "Conference (CONF) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification"(Release 11).

[17] 3GPP TS 24. 628 V11.2.0 (2012-12): Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Common Basic Communication procedures using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (Release 11)

[18] 3GPP TS 23.060 V11.5.0 (2013-03): Technical Specification Group Services and System Aspects; General Packet Radio Service (GPRS); Service description; Stage 2 (Release 11)

[19] 3GPP TS 23.228 V11.6.0 (2012-12): Technical Specification Group Services and System Aspects; IP Multimedia Subsystem (IMS); Stage 2 (Release 1)

[20] 3GPP TS 23.401 V11.5.0 (2013-03): Technical Specification Group Services and System Aspects; General Packet Radio Service (GPRS) enhancements for Evolved Universal Terrestrial Radio Access Network (E-UTRAN) access (Release 11)

[21] 3GPP TS 24.229 V11.6.0 (2012-12): 3rd Generation Partnership Project; Technical Specification Group Core Network and Terminals; IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 (Release 11)

[22] 3GPP TS 24.615: 3rd Generation Partnership Project; Technical Specification Group Core Network and Terminals; Communication Waiting (CW) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification; (Release 11)

[23] 3GPP TS 24.642 : 3rd Generation Partnership Project; Technical Specification Group Core Network and Terminals; Completion of Communications to Busy Subscriber (CCBS) Completion of Communications by No Reply (CCNR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification; (Release 11)

[24] 3GPP TS 29.163 v11.5.0 (2012-12): 3rd Generation Partnership Project; Technical Specification Group Core Network and Terminals; Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (Release 11)

[25] ITU-T E.164 (02/2005): OVERALL NETWORK OPERATION, TELEPHONE SERVICE, SERVICE OPERATION AND HUMAN FACTORS; International operation – Numbering plan of the international telephone service; The international public telecommunication numbering plan

[26] ITU-T G.711 (11/1988): GENERAL ASPECTS OF DIGITAL TRANSMISSION SYSTEMS, TERMINAL EQUIPMENT, Pulse code modulation (PCM) of voice frequencies

[27] ITU-T G.722 (1993): GENERAL ASPECTS OF DIGITAL TRANSMISSION SYSTEMS, TERMINAL EQUIPMENT, 7 kHz AUDIO – CODING WITHIN 64 KBIT/S

[28] ITU-T H.261 (03/93): LINE TRANSMISSION OF NON-TELEPHONE SIGNALS; VIDEO CODEC FOR AUDIOVISUAL SERVICES AT p x 64 kbits

[29] ITU-T H.263 (01/2005): AUDIOVISUAL AND MULTIMEDIA SYSTEMS; Infrastructure of audiovisual services – Coding of moving video ; Video coding for low bit rate communication

[30] ITU-T H.264 (11/2007): AUDIOVISUAL AND MULTIMEDIA SYSTEMS; Infrastructure of audiovisual services – Coding of moving video; Advanced video coding for generic audiovisual services

[31] ITU-T T.30 (09/2005): TERMINALS FOR TELEMATIC SERVICES; Procedures for document facsimile transmission in the general switched telephone network

[32] ITU-T T.38 (04/2007): TERMINALS FOR TELEMATIC SERVICES; Facsimile – Group 3 protocols; Procedures for real-time Group 3 facsimile communication over IP networks

[33] ITU-T V.152 (01/2005): DATA COMMUNICATION OVER THE TELEPHONE NETWORK; Interworking with other networks; Procedures for supporting voice-band data over IP networks

[34] IETF RFC 0768: User Datagram Protocol; 28 August 1980

[35] IETF RFC 0791: INTERNET PROTOCOL DARPA INTERNET PROGRAM PROTOCOL SPECIFICATION; September 1981

[36] IETF RFC 0792: INTERNET CONTROL MESSAGE PROTOCOL; September 1981

[37] IETF RFC 0793: TRANSMISSION CONTROL PROTOCOL; DARPA INTERNET PROGRAM; PROTOCOL SPECIFICATION; September 1981

[38] IETF RFC 1035: DOMAIN NAMES - IMPLEMENTATION AND SPECIFICATION; November 1987

[39] IETF RFC 2032: RTP Payload Format for H.261 Video Streams; October 1996

[40] IETF RFC 2046: Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types; November 1996

[41] IETF RFC 2190: RTP Payload Format for H.263 Video Streams; September 1997

[42] IETF RFC 2246: The TLS Protocol; Version 1.0; January 1999

[43] IETF RFC 2327: SDP: Session Description Protocol; April 1998

[44] IETF RFC 2411: IP Security Document Roadmap; November 1998

[45] IETF RFC 2429: RTP Payload Format for the 1998 Version of ITU-T Rec. H.263 Video (H.263+); October 1998

[46] IETF RFC 2460: Internet Protocol, Version 6 (IPv6) Specification; December 1998

[47] IETF RFC 2782: A DNS RR for specifying the location of services (DNS SRV); February 2000

[48] IETF RFC 2915: The Naming Authority Pointer (NAPTR) DNS Resource Record; September 2000

[49] IETF RFC 3041: Privacy Extensions for Stateless Address Autoconfiguration in IPv6; January 2001

[50] IETF RFC 3263: Session Initiation Protocol (SIP): Locating SIP Servers; June 2002

[51] IETF RFC 3316: Internet Protocol Version 6 (IPv6) for Some Second and Third Generation Cellular Hosts; April 2003

[52] IETF RFC 3362: Real-time Facsimile (T.38) - image/t38; MIME Sub-type Registration; August 2002

[53] IETF RFC 3550: RTP: A Transport Protocol for Real-Time Applications; July 2003

[54] IETF RFC 3596: DNS Extensions to Support IP Version 6; October 2003

[55] IETF RFC 3842: A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP); August 2004

[56] IETF RFC 3858: An Extensible Markup Language (XML) Based Format for Watcher Information; August 2004

[57] IETF RFC 3863: Presence Information Data Format (PIDF); August 2004

[58] IETF RFC 3890: A Transport Independent Bandwidth Modifier for the Session Description Protocol (SDP); September 2004

[59] IETF RFC 3966: The tel URI for Telephone Numbers; December 2004

[60] IETF RFC 3984: RTP Payload Format for H.264 Video; February 2005

[61] IETF RFC 4028: Session Timers in the Session Initiation Protocol (SIP); April 2005

[62] IETF RFC 4040: RTP Payload Format for a 64 kbit/s Transparent Call; April 2005

[63] IETF RFC 4122: A Universally Unique IDentifier (UUID) URN Namespace; July 2005

[64] IETF RFC 4443: Internet Control Message Protocol (ICMPv6) for the Internet Protocol Version 6 (IPv6) Specification; March 2006

[65] IETF RFC 4733: RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals; December 2006

[66] IETF RFC 4734: Definition of Events for Modem, Fax, and Text Telephony Signals; December 2006

[67] IETF RFC 5244: Definition of Events for Channel-Oriented Telephony Signalling; June 2008

[68] IETF RFC 5626: Managing Client-Initiated Connections in the Session Initiation Protocol (SIP); October 2009

[69] IETF RFC 5627: Obtaining and Using Globally Routable User Agent URIs (GRUUs) in the Session Initiation Protocol (SIP); October 2009

[70] IETF RFC4884: Extended ICMP to Support Multi-Part Messages, April 2007

[71] IETF RFC3261: "SIP: Session Initiation Protocol".

[72] Broadband Forum TR-069 "CPE WAN Management Protocol"

[73] Broadband Forum TR-104 "DSLHome Provisioning Parameters for VoIP CPE"

[74] Broadband Forum TR-181 "Device Data model for TR-069",

[75] 3GPP TS 23.003: "Numbering, addressing and identification".

3 Abbreviations, Definitions and Symbols

Abbreviations, definitions and symbols, not listed hereafter, are defined in the reference documents in clause 2.

3.1 Abbreviations

For the purposes of the present document, the following abbreviations apply:

-1-	
3GPP	Third Generation Partnership Project
3GPP TS	3GPP Technical Specification (normative)
3PCC	Three Party Call Control
3PTY	Three party conference
-A-	
AAA	Authorization Authentication Accounting
ACR	Anonymous Communication Rejection
ack.	Acknowledge (tone or announcement)
AS	Application Server
-B-	
-C-	
CC	Call Control
CC	Country Code
CCBS	Completion of Communications to Busy Subscriber
CD	Communication session Deflection
CDIV	Communication Diversion Services
CFB	Communication session Forwarding Busy
CFN	Call Forwarding Number, e.g. <CFN>
CFNR	Communication session Forwarding No Reply
CFU	Communication session Forwarding Unconditional
CLID	Calling Line Identification
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
CN	Calling Number (Calling Party Number), e.g. <CN>

COLP	Connected Line Identification Presentation
COLR	Connected Line Identification Restriction
CONF	Conference (Konferenz)
CW	Call Waiting
-D-	
DDI	Direct Dial In
DIV	Digitales Vermittlungssystem
DN	Destination Number, e.g. <DN>
DNS	Domain Name System
DSLAM	Digital Subscriber Line Access Multiplexer
DT	Deutsche Telekom
-E-	
ES	European Standard
ETSI	European Telecommunication Standardisation Institute
ETSI ES	ETSI Standard (normative)
ETSI TR	ETSI Technical Report (informative)
ETSI TS	ETSI Technical Specification (normative)
ETSI/TISPAN	Merger of ETSI Project Tiphon and ETSI Technical Body SPAN
-F-	
FCR	Forwarded Communication Restriction
-G-	
G-ISUP	Gateway ISUP
GRUU	Globally Routable User Agent URI
-H-	
HTTP	Hypertext Transfer Protocol
-I-	
IAD	Integrated Access Device
IAF	Internet-aware fax device
IARI	IMS Application Reference Identifier
ICSI	IMS Communication Service Identifier
ICMP	Internet Control Message Protocol
IETF	Internet Engineering Task Force
INTIME	Independent Timer

IP	Internet Protocol
IPv4	Internet Protocol Version 4
IPv6	Internet Protocol Version 6
ISDN	Integrated Services Digital Network
ISIM	IM Subscriber Identity Module
ISUP	ISDN User Part
ITU-T	International Telecommunication Union, Telecommunication Branch
IWU	Interworking Unit
-J-	
-K-	
-L-	
-M-	
MCID	Malicious Call Identification
MEGACO	Media Gateway Control Protocol
MG	Media Gateway
MGC	Media Gateway Controller
MIME	Multipurpose Internet Mail Extensions
MMS	Multimedia Message Service
MSN	Multiple Subscriber Number
MTP	Message Transfer Part
MWI	Message Waiting Indicator
-N-	
NAPT	Network Address Port Translation
NAS	Network Access Server
NASS	Network Attachment Subsystem
NAT	Network Address Translation
NGN	Next Generation Networks
-O-	
OIP	Originating Identification Presentation
OIR	Originating Identification Restriction
ON	Originating Number, e.g. <ON>

-P-

PCM	Pulse Code Modulation
PDU	Protocol Data Unit
PIN	Personal Identification Number
PINnew	new PIN (4 digits)
PINold	old PIN (4 digits)
PPP	Point to Point Protocol
PPPoE	Point to Point Protocol over Ethernet
PSTN	Public Switched Telephone Network
PUID	Public User Identity

-Q-

QoS	Quality of Service

-R-

RADIUS	Remote Authentication Dial-In User Service
RFC	Request for Comments
RTCP	Real Time Control Protocol
RTP	Real Time Transport Protocol

-S-

SCC	Service Code Command
SCTP	Stream Control Transmission Protocol
SDES	Source Description RTCP Packets
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SIP I	SIP ISUP (SIP with encapsulated ISUP MIME)
SIP T	SIP Telephony
SMS	Short Message Service
SN	Subscriber Number
SOC	Switching Order Command
SS	Supplementary Service
STUN	Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs); also: Session Traversal Utilities for NAT

-T-

TAS	Telephony Application Service

TCP	Transmission Control Protocol
TCP/IP	Transmission Control Protocol / Internet Protocol
TDM	Time Division Multiplex
TIP	Terminating Identification Presentation
TIR	Terminating Identification Presentation Restriction
T-ISUP	Telekom ISUP
TR	Technical Recommendation
TR	Technical Report [ETSI]
TS	Technical Specification
TS	Technical Specification [ETSI]
-U-	
UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
UDP	User Datagram Protocol
UE	User Equipment
UICC	Universal Integrated Circuit Card
UNI	User Network Interface
URI	Universal Resource Identifier
URL	Uniform Resource Locator
-V-	
-W-	
-X-	
xDSL	x Digital Subscriber Line (x stands for various kinds of bit rates)
XML	Extended Markup Language
-Y-	
-Z-	

3.2 Definitions

For the purposes of the present document, the following terms and definitions apply:

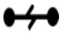
Term	**Definition / Remark**
User Equipment	Any SIP device (terminal) at the subscriber premises used by an end user to communicate. It can be e.g. an IAD or telephone set, or any other telecommunication device.
User Agent	A client application used with a particular network protocol, such as Session Initiation Protocol (SIP); it refers to both end points of a phone call, server and client.
Call Control	In telephony, call control refers to the software within a telephone switch that supplies its central function. Call control decodes addressing information and routes telephone calls from one end point to another. It also creates the features that can be used to adapt standard switch operation to the needs of users. Call control software, because of its central place in the operation of the telephone network, is marked by both complexity and reliability.
NGN platform (of Deutsche Telekom)	The entirety of central servers and gateways, as well as software within a telephone network which provides telephony services.
VoIP line	A VoIP line is equivalent to a MSN in ISDN; Multiple VoIP lines can be assigned to a VoIP account of the NGN platform of Deutsche Telekom.
IP	Considering the expected parallel availability of IPv4 and IPv6 the term “IP” in this document is related to both internet protocol versions.
Shall	As usual within standards and documents of Deutsche Telekom, 3GPP, ETSI and ITU-T the word shall is used to indicate a procedure or requirement as mandatory. (Note)
Should	As usual within standards and documents of Deutsche Telekom, 3GPP, ETSI and ITU-T the word should is used to indicate a procedure or requirement as optional (Note)

Note: The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119.

3.3 Symbols

For the purposes of the present document, the following symbols apply:

Symbol	**Definition**

Symbol	Definition
*	Star sign
#	Hash sign
	Pick-up the receiver (Off-hook)
	Hang-up the receiver (On-hook)
	Hook Flash (flash hook function) (procedure to invoke “register recall”); In case of a SIP UE it shall invoke an equivalent SIP procedure according to the service specifications as described in Annex C.
< >	Information in pointed brackets is an obliged variable input (signs or figures)

4 General Description

4.1 Introduction

The existing PSTN/ISDN technology of Deutsche Telekom will be replaced by an IP-based Next Generation Network (NGN) using the SIP protocol.

The subscriber needs a suitable terminal to be able to use the SIP-based telephony services via the NGN platform of Deutsche Telekom. These terminals can be either hard phones, adaptors or soft phones.

- A hard phone is a conventional telephone but with an interface for connecting to an IP network.
- In addition, routers (with an integrated DSL modem) and adaptors are commercially available which allow conventional telephones be used for Internet telephony by means of a special interface (IAD). The NGN platform of Deutsche Telekom also supports these integrated adapters. Legacy TDM and ISDN-terminals can still be operated in the customer-area via such adapters.
- A soft phone is the name given to application software which allows a PC to perform SIP-based telephony over the Internet.

4.2 Capabilities

4.2.1 SIP capabilities

The request of Preconditions (indication of SUPPORT/REQUIRED within an initial INVITE) are not part of this specification. Only passive support is required.

Requests received indicating support of Preconditions must be supported. This requirement is needed to fulfil FMC when Mobile devices are calling the UE. Further information pleases see in ANNEX B of this document TS 24.229 5.1.4.1

Sending of preconditions Supported or required with the initial INVITE SHALL NOT be done.

Reliable Provisional Responses are mandatory.

SIP URIs shall be supported in SIP header fields.

The use of the P-Early-Media header is mandatory i.e. each INVITE has to contain a P-Early-Media Header set to supported.

For SIP UE supporting SIP-Analogue and SIP-ISDN interworking (e.g. IAD) the subscription of the "ua-profile" is **implicit.** This overrules the procedures stated within 1TR126 ANNEX A Section “A 5.3.1.2 Subscription for profile delivery” and 1TR127 ANNEX B Section “A 5.3.1.2 Subscription for profile delivery”

Note: The sending of the SUBSCRIBE Method for the “ua-profile” is not needed.

4.2.2 Telephony

Voice over IP (VoIP) is performed in accordance with the SIP-Protocol. The specifications to be fulfilled for control of a communication are presented in section 7.

For the Media-Stream the Codecs G.711a [26] (A-Law) and G.722 [27] shall be used. For VGW (IAD) supporting ISDN accesses RFC 4040 [62] (Clearmode) shall be supported.

4.2.3 Fax and Modem

For Fax and Modem transmission over IP, ITU-T Rec. V.152 [33] (based on G.711a [26]) shall be used. If the adjacent endpoint does not support ITU-T Rec. V.152 [33], Fax- and Modem connections shall be set up using G.711a (ITU-T Rec. T.30 [31]).

Currently, ITU-T Rec. T.38 [32] is not supported by the NGN platform of Deutsche Telekom.

4.2.4 Video Telephony

The call control is performed in accordance with the SIP-Protocol as for Telephony, however specific options for multimedia communication are used.

For the Media-Stream the Codecs H.263 [29] and H.264 [30] shall be used (see section 6 Codecs).

4.2.5 DTMF

For DTMF events RFC 4733 [65] and RFC 5244 [67] shall be supported.

For support of DTMF RTP Out-of-Band in binary format RFC 4733 [65] and RFC 4734 [66] shall be supported.

Note: In cases where the remote Endpoint does not support RFC4733 it shall be possible to sent DTMF inband.

4.2.6 Early Media

For early media RFC 5009 MUST be supported. Due to the fact that not all functionalities will support RFC5009 for early media further procedures for identifying early media needs to be supported. In addition not in each case where an SDP is received within a provisional response early media apply.

Therefore the following procedures to identify if early media is received shall apply in the following rowing:

1. If a provisional response includes a P-Early-Media Header with "sendonly" and a require header with 100rel. The procedures shall apply with 3GPP TS 24.628 [17].

2. If a provisional response contains SDP and preconditions are not used.

3. Identifying if an RTP stream is received by the UE.

4.2.7 Locating of P-CSCF (Proxy) in case of (re-) Registration and change of P-CSCF priority due to Maintenance

For P-CSCF discovery and Registration Procedures the Specifications TS 24.229 [21], RFC3261 [71], RFC 2782 [47] and RFC 3263 [50] are valid.

The following procedures shall give a hint for End device vendors how to implement these procedures to fulfil the requirements of Deutsche Telekom.

Due to maintenance and failure situations the prioritization of P-CSCF can change. Therefore the destination must be determined by applying the DNS procedures described within RFC3261 [71], RFC 2782 [47], RFC 3263 [50] and ANNEX B of this document.

A DNS query to request the actual SRV record set shall be done before sending a REGISTER or re-REGISTER request.

NOTE: This is valid in cases where the registration timer expires, or a network initiated deregister was sent or in cases where final responses where received pointing to a failure situation where the target cannot be reached (e.G. 503 Response) or a redirect (305 response) was received.

As described within RFC 2782 [47] a client MUST attempt to contact the target host (P-CSCF) with the lowest-numbered priority it can reach; target hosts with the same priority SHOULD be tried in an order defined by the weight field. Within Deutsche Telekom network normally only the priority field is used.

TTL expiry shall be taken into consideration when starting Register and re-Register procedures.

4.2.8 Preconditions

This section should help to the settings of precondition parameter and their interpretation.

Generally as indicated in Section 4.2.1 the request of Preconditions (indication of SUPPORT/REQUIRED within an initial INVITE) are not part of this specification. Due to the face of mobile convergence mobile requests from devices requesting the precondition mechanism may be received. Therefore the "passive" support of preconditions is required.

The "integration of resource management and SIP" extension is hereafter in this subclause referred to as "the precondition mechanism" is defined in RFC 3312 as updated by RFC 4032.

By "passive" support is meant that a UE receiving:

- a supported header field containing the precondition tag shall, if the configuration requirement of Deutsche Telekom requires preconditions, proceed with the precondition procedures, or
- a supported header field containing the precondition tag shall, if the configuration requirement of Deutsche Telekom do NOT require preconditions, NOT proceed with the precondition procedures, or
- a required header field containing the precondition tag shall proceed with the precondition procedures

The support of the precondition mechanism in case of the setting of the supported header to preconditions shall be configurable.

The procedures described in ANNEX B (Section 5.1.4.1) are valid in case of receiving an initial INVITE with requesting preconditions as shown above.

4.2.9 Auto configuration

Auto configuration procedures are described within the Broadband Forum Technical Reports *TR-069 [72], TR-181 [74] and TR104 [72] and shall be considered for the implementation.*

5 SIP Service functionality requirements

5.1 General

The SIP service functionality requirements are defined in Annex B of the present document.

Further specific service requirements are described in the following.

The Services Communication HOLD (Section C.2.6), Communicating Waiting (CW) (Section C.2.8), TOGGEL and CONF (Section C.2.2) shall be implemented as a End Client Service feature and as described within this specification.

The network centric feature logic for HOLD, CW, TOGGEL and CONF are not available, therefore these features must be additionally implemented locally on the SIP Client. This must be configured as default.

Based on the VGW (IAD) configuration it must be possible to activate a local/terminal based CW on a busy line if an INVITE without a CW indication is received by the VGW.

5.2 Direct Dial In (DDI)

DDI is for further study.

6 Codecs

Codecs listed in the following Table6-1 and Table6-2 are supported by the NGN platform of Deutsche Telekom.

NOTE: If no transcoding rules or other restrictions (e.g. RACS) contradict, any audio and video codecs will be transparently conveyed through the NGN platform of Deutsche Telekom.

Specification	Title	Reference
G.711	Pulse code modulation (PCM) of voice frequencies; (A-law).	[26]
G.722	7 kHz Audio – Coding within 64 kBit/s	[27]

Table6-1: Audio Codecs

Specification	Title	Reference
H.261	Video codec for audiovisual services at p x 64 kbit/s.	[28]
H.263	Coding of moving video; Video coding for low bit rate communication	[29]
H.264	Advanced video coding for generic audiovisual services.	[30]

Table6-2: Video Codecs

7 Protocol (Profiles)

This section profiles the Gm interface for SIP UE intended to be connected to the NGN platform of Deutsche Telekom based on 3GPP TS 24.229 Release 11 [21] (endorsements).

<u>Markings general used within the TEXT:</u>

Text modified due to Deutsche Telekom requirements that is added or deleted compared to 3GPP TS 24.229 Release 11 [21] is shown as cursive and underlined (*example for added text*) or cursive and stricken (~~*example for stricken text*~~).

For information: As usual within 3GPP Standards notes in Tables are mandatory and have to be implemented

7.1 Void

7.1.1 Void

7.2 Modifications to 3GPP TS 24.229 (endorsement)

The relevant modifications to 3GPP TS 24.229 [21] for SIP UE (Gm interface) intended to be connected to the NGN platform of Deutsche Telekom are provided in Annex B of the present document.

7.2.1 Global modifications to 3GPP TS 24.229 Release 11

- *A SIP UE supporting the services provided by the NGN platform of Deutsche Telekom shall use the procedures with the relevant Service Command Codes (SCC) (e.g. *21#). These service procedures (incl. SCC) are described in Annex D of this document. Either the SCC can be directly dialled via the key pad on the SIP Phone or via specific service menu buttons (e.g.: Press button "xyz" for invoking CCBS) which initiates the regarding SCC. The SCC shall be sent in the format of a SIP URI: SCC@hostportion within an initial INVITE. If SIP equivalent procedures are available and supported by the network these shall be preferred.*

- *Particular services provided by the NGN platform of Deutsche Telekom require specific procedures using Switching Order Commands (SOC). A SIP UE supporting these services shall use the procedures (incl. SOC) which are described in Annex D of this document and 1TR126 [3].*

- *The Hook-Flash handling and the invocation of services are described in 1TR126 [3]. The implementation of the procedures for the Hook-Flash handling, if it is a real Hook-Flash or only a menu button to invoke the service; is a matter of the vendor and out of the scope of this document.*

- *Request URI = SIP URI with user=phone.*

- *For future network improvements the capabilities of registering and sending SIP URI as defined for Public User Identities in 3GPP TS 23.003 [75] SHOULD exist. Currently the only Format used is SIP URI's representing a E.164 Number in the host portion. Default is SIP URI with user=phone*

- *Header fields received my contain tel URI or alias URI as defined in 3GPP TS 23.003 [75]*

- *All URI (Request, From etc.) should be presented within global number format.*

- *Request URI = SIP URI with user=phone is used for SCC like *21%23@hostportion.*

- *# (hash) in URIs must be escaped.*

- *The Re-Ringing procedure for the SIP UE shall apply according to 1TR126 [3].*
- *The Protocol stacks shall work with IPv4 and IPv6.*
- *Neither UICC nor ISIM is applicable for this document.*
- *ICSI and IARI are currently not used, but nevertheless elements included within SIP messages shall be passed on in compliance with the current specification.*
- *The Call-Id shall not include the own IP-address of the UA.*
- *Network initiated De-Registration is part of this specification and must be supported.*
- *Authentication shall be possible via HTTP Digest and without HTTP Digest (NASS bundled) based on the line/IP-Address.*
- *Support of session timers regarding RFC 4028 [61] is mandatory.*
- *For tones and announcements the procedures described in 3GPP TS 2. 628, Annex D [17] shall apply. The bidirectional early media shall be used.*
- *Any final response either 200 OK or final error response (e.g. 4xx) shall close all existing early dialogs for the regarding Call-ID.*
- *To avoid problems with a wide spread of existing clients 3PCC procedures shall only send INVITE without SDP and with 100rel supported so that the UAS can decide in sending reliable or unreliable provisional responses.*
- *The AS shall send a 199 for the release of early dialogs. A further 18x response (e.g. 180 in case of CCNR activation rejection) may be sent afterwards. For UE 199 is mandatory to understand.*
- *The challenge mechanism shall be supported.*
- *To avoid too many challenge cycles the nonce shall be included within each request during its validity.*
- *DNS SRV capabilities (including TTL) shall be supported (e.g. see chapter 4.2 in Annex B).*
- *HEX digits as defined within RFC 3966 [58] to be sent or received on the Gm interface in SIP URI user=phone are not allowed.*
- *UE shall minimise or avoid REGISTER procedures for identifying fetch binding's. The avoidance of this procedure is preferred to minimise the network load.*

- *The restoration procedures as described within ANNEX B shall be supported.*

De-Registration

In cases where UE's are booting, there is no knowledge if the UE is already registered or not. Therefore De-Register with "*" in the contact header field is forbidden.

Implicit Registration shall be supported with the following procedures:

For each stored Public URI's (IMPU) received via TR-069 [72], TR-181 [74] and TR104 [72] protocol via the initial configuration the following registration procedure shall apply.

During the initial configuration with the TR-069 [72], TR-181 [74] and TR104 [72] protocol each received IMPU shall be stored.

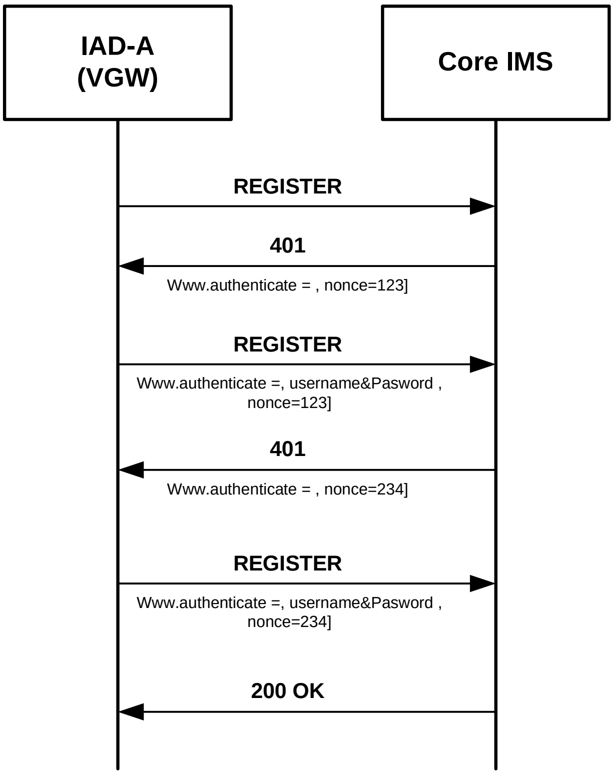
For each IMPU an explicit registration shall apply with the following exception.

In cases where a P-Associated-URI with additional registered URI's (implicit registration set) is received the IAD shall compare these URI's with the stored URI's received via TR-069 [72], TR-181 [74] and TR104 [72].

For IMPU's implicit registered an explicit registration shall not apply in addition.

The early Challenge Procedure shall be supported as follows:

If the early challenge procedure is requested due to local policy it must be supported



Note: The early challenge procedure may appear in some circumstances and is depended on local IMS policy

General procedure for REGISTER Message answered with a 403

General a 403 is an Indication that the user is not provisioned within the HSS. Nevertheless if 403 (Forbidden) has been received as a response to a REGISTER request, a further registration attempts shall be done after 15 sec. In case further 403 responses received with the same URI in the Contact header field REGISTER requests are allowed with a random delay of 30- 60 minutes.

7.3 UE (Gm) interface; Profile tables based on 3GPP TS 24.229

In the following section the actual numbering of the endorsement document is kept with a leading "§" sign, if applicable. If not explicit noted, the references mentioned within the following tables apply to Annex B i.e the numbering of reference [xyz] is the same as used within TS 24.229. The Reference section is in ANNEX B Section 2

7.3.1 Table description

Table 7-1: Table Type 1 Example

Item	PDU or Header	Sending			Receiving		
		Ref.	Profile status UE	UNI (Gm)	Ref.	Profile status UE	UNI (Gm)
1	2	3	4	5	6	7	8

Legend:

Column 1:	Item numbering
Column 2:	Identification of PDU (Method) or SIP Header
Column 3/4/5:	Sending = from UE to P-CSCF
Column 6/7/8:	Receiving = from P-CSCF to UE
Column 3/6:	Reference to section 2 "References" in Annex B .e the numbering of reference [xyz] is the same as used within TS 24.229. The Reference section is in ANNEX B Section 2
Column 4/7:	Profile Status of capabilities the UE has to support
Column 5/8:	Profile Status of the IMS Gm Interface

7.3.2 PDUs (SIP Methods)

The following methods shall be supported on the Gm Interface and the UE.

NOTE: A Method Request (e.g. INVITE request) is the Method itself.

NOTE: A Method Response is the Response (e.g. 1xx) sent as a result of a Request.

The following table is based on Table §A.5: “Supported methods” of Annex B and profiled for this document.

Table 7-2: Supported methods

Item	PDU	Sending			Receiving		
		Ref.	Profile status UE	UNI (Gm)	Ref.	Profile status UE	UNI (Gm)
1	ACK request	[26] 13	m	m	[26] 13	m	m
2	BYE request	[26] 15.1	m	m	[26] 15.1	m	m
3	BYE response	[26] 15.1	m	m	[26] 15.1	m	m
4	CANCEL request	[26] 9	m	m	[26] 9	m	m
5	CANCEL response	[26] 9	m	m	[26] 9	m	m
6	INFO request	[26] 13	c6	m	[26] 13	c5	m
7	INFO response	[26] 13	c6	m	[26] 13	c6	m
8	INVITE request	[26] 13	m	m	[26] 13	m	m
9	INVITE response	[26] 13	m	m	[26] 13	m	m
9A	MESSAGE request	[50] 4	c11	m	[50] 7	c10	m
9B	MESSAGE response	[50] 4	c11	m	[50] 7	c11	m
10	NOTIFY request	[28] 8.1.2	c8	m	[28] 8.1.2	c9	m
11	NOTIFY response	[28] 8.1.2	c8	m	[28] 8.1.2	c8	m
12	OPTIONS request	[26] 11	o	m	[26] 11	m (Note 1)	m
13	OPTIONS response	[26] 11	o	m	[26] 11	o	m
14	PRACK request	[27] 6	c7	m	[27] 6	c7	m
15	PRACK response	[27] 6	c7	m	[27] 6	c7	m
15A	PUBLISH request	[70] 3	c15	c15	[70] 3	c14	c14
15B	PUBLISH response	[70] 3	c15	c15	[70] 3	c15	c15
16	REFER request	[36] 3	c13	m	[36] 3	c12	m
17	REFER response	[36] 3	c13	m	[36] 3	c13	m
18	REGISTER request	[26] 10	c16	c16	[26] 10	n/a	n/a
19	REGISTER response	[26] 10	n/a	n/a	[26] 10	c16	c16
20	SUBSCRIBE request	[28] 8.1.1	m	m	[28] 8.1.1	o	m
21	SUBSCRIBE response	[28] 8.1.1	m	m	[28] 8.1.1	o	m
22	UPDATE request	[30] 6.1	c17	c17	[30] 6.2	c17	c17

Item	PDU	Sending			Receiving		
		Ref.	Profile status UE	UNI (Gm)	Ref.	Profile status UE	UNI (Gm)
23	UPDATE response	[30] 6.2	c17	c17	[30] 6.1	c17	c17
24	other requests		n/a	n/a		n/a	c3
25	other response		n/a	c2		n/a	n/a

Conditions for Table 7-3:

c1: IF received reject with 405.

c2: IF received then response shall be ignored.

c3: IF received reject with 501.

c4: Void.

c5: IF AoC or other Deutsche Telekom applications using INFO Then m ELSE c1.

c6: IF AoC or other Deutsche Telekom applications using INFO Then m ELSE n/a.

c7: IF preconditions or Tones and Announcements (early media) with 18x THEN m ELSE o.

c8: IF CCBS or Dial tone Package or other Deutsche Telekom applications using NOTIFY THEN m ELSE n/a.

c9: IF CCBS or Dial tone Package or other Deutsche Telekom applications using NOTIFY THEN m ELSE c1.

c10: needed for future services (please note that MWI is not longer supported within Deutsche Telekom Core Network).

c11: needed for future services (please note that MWI is not longer supported within Deutsche Telekom Core Network).

c12: IF ECT or other application using REFER message THEN m Else c1.

c13: IF ECT or other application using REFER message THEN m Else n/a.

c14: IF Presence or other application using Publish THEN m ELSE c1.

c15: IF Presence or other application using Publish THEN m ELSE n/a.

c16: IF A.4/1 in ANNEX B THEN m ELSE n/a - - client behaviour for registration.

c17 IF A.4/17 in ANNEX B THEN m ELSE n/a - - the SIP UPDATE method?

NOTE 1: The OPTION method is used to check reliability between UE and P-CSCF.

7.3.3 Supported status-codes on the Gm -Interface

The following status-codes shall be supported on the Gm-Interface:

Table 7-4: Supported status-codes

Item	Header	Sending			Receiving		
		Ref.	RFC status	Profile status UNI (Gm)	Ref.	RFC status	Profile status UNI (Gm)
1	100 (Trying)	[26] 21.1.1	c21	m	[26] 21.1.1	c11	m
101	1xx response	[26] 21.1	p21	m	[26] 21.1	p21	m
101A	18x response	[26] 21.1	p21	m	[26] 21.1	p21	m
2	180 (Ringing)	[26] 21.1.2	c2	m	[26] 21.1.2	c1	m
3	181 (Call Is Being Forwarded)	[26] 21.1.3	c2	m	[26] 21.1.3	c1	m
4	182 (Queued)	[26] 21.1.4	c2	m	[26] 21.1.4	c1	m
5	183 (Session Progress)	[26] 21.1.5	c1	m	[26] 21.1.5	c1	m
5A	199 (Early Dialog Terminated)	[142] 8	c32	m	[142] 8	c32	m
102	2xx response	[26] 21.2	p22	m	[26] 21.1	p22	m
6	200 (OK)	[26] 21.2.1	m	m	[26] 21.2.1	m	m
7	202 (Accepted) Depending on the use of the SUBSCRIB/NOTIFY, REFER.	[28] 8.3.1	c3	o	[28] 8.3.1	c3	o
103	3xx response	[26] 21.3	p23		[26] 21.1	p23	
8	300 (Multiple Choices)	[26] 21.3.1	o	n/a (Note 3)	[26] 21.3.1	o	n/a (Note 4)
9	301 (Moved Permanently)	[26] 21.3.2	o	n/a (Note 3)	[26] 21.3.2	o	n/a (Note 4)
10	302 (Moved Temporarily)	[26] 21.3.3	o	c33	[26] 21.3.3	o	n/a (Note 4)
11	305 (Use Proxy)	[26] 21.3.4	o	n/a	[26] 21.3.4	o	m
12	380 (Alternative Service)	[26] 21.3.5	o	n/a	[26] 21.3.5	o	n/a (Note 4)
104	4xx response	[26] 21.4			[26] 21.4		
13	400 (Bad Request)	[26] 21.4.1	m	m	[26] 21.4.1	m	m

Item	Header	Sending			Receiving		
		Ref.	**RFC status**	**Profile status UNI (Gm)**	**Ref.**	**RFC status**	**Profile status UNI (Gm)**
14	401 (Unauthorized) OPEN due to the registration mechanism used.	[26] 21.4.2	o	n/a	[26] 21.4.2	m	m (Note 2)
15	402 (Payment Required)	[26] 21.4.3	n/a	n/a	[26] 21.4.3	n/a	n/a
16	403 (Forbidden)	[26] 21.4.4	m	m	[26] 21.4.4	m	m (Note 5)
17	404 (Not Found)	[26] 21.4.5	m	m	[26] 21.4.5	m	m
18	405 (Method Not Allowed)	[26] 21.4.6	m	m	[26] 21.4.6	m	m
19	406 (Not Acceptable)	[26] 21.4.7	m	m	[26] 21.4.7	m	m
20	407 (Proxy Authentication Required)	[26] 21.4.8	o	n/a	[26] 21.4.8	m	o
21	408 (Request Timeout)	[26] 21.4.9	c2	m	[26] 21.4.9	m	m
22	410 (Gone)	[26] 21.4.10	m	o	[26] 21.4.10	m	m
22A	412 (Conditional Request Failed)	[70] 11.2.1	c20	o	[70] 11.2.1	c20	m
23	413 (Request Entity Too Large)	[26] 21.4.11	m	o	[26] 21.4.11	m	m
24	414 (Request-URI Too Large)	[26] 21.4.12	m	m	[26] 21.4.12	m	m
25	415 (Unsupported Media Type)	[26] 21.4.13	m	m	[26] 21.4.13	m	m
26	416 (Unsupported URI Scheme)	[26] 21.4.14	m	m	[26] 21.4.14	m	m
26A	417 (Unknown Resource Priority)	[116] 4.6.2	n/a	n/a	[116] 4.6.2	c24	c24
27	420 (Bad Extension)	[26] 21.4.15	m	m	[26] 21.4.15	m	m
28	421 (Extension Required)	[26] 21.4.16	o		[26] 21.4.16	i	
28A	422 (Session Interval Too Small)	[58] 6	c7	m	[58] 6	c7	m
29	423 (Interval Too Brief)	[26] 21.4.17	c4	m	[26] 21.4.17	m	m
29A	424 (Bad Location Information)	[89] 3.3	n/a	n/a	[89] 3.3	c22	c22
29B	429 (Provide Referrer Identity)	[59] 5	c8	n/a	[59] 5	c9	n/a

Item	Header	Sending			Receiving		
		Ref.	RFC status	Profile status UNI (Gm)	Ref.	RFC status	Profile status UNI (Gm)
29C	430 (Flow Failed)	[92] 11	c14	C34	[92] 11	c14	m
29D	433 (Anonymity Disallowed)	[67] 4	c14	C34	[67] 4	c14	m
29E	439 (First Hop Lacks Outbound Support)	[92] 11	c28	n/a	[26] 21.4.18	c29	o
29F	440 (Max Breadth Exceeded)	[117] 5	n/a	n/a	[26] 21.4.18	c31	o
29G	469 (Bad INFO Package)	[25] 4.4	c33	n/a	[25] 4.4	c33	o
29H	470 (Consent Needed)	[125] 5.9.2	c26	n/a	[26] 21.4.18	c27	o
30	480 (Temporarily Unavailable)	[26] 21.4.18	m	m	[26] 21.4.18	m	m
31	481 (Call/Transaction Does Not Exist)	[26] 21.4.19	m	m	[26] 21.4.19	m	m
32	482 (Loop Detected)	[26] 21.4.20	m	m	[26] 21.4.20	m	m
33	483 (Too Many Hops)	[26] 21.4.21	m	m	[26] 21.4.21	m	m
34	484 (Address Incomplete)	[26] 21.4.22	o	m	[26] 21.4.22	m	m
35	485 (Ambiguous)	[26] 21.4.23	o	m	[26] 21.4.23	m	m
36	486 (Busy Here)	[26] 21.4.24	m	m	[26] 21.4.24	m	m
37	487 (Request Terminated)	[26] 21.4.25	m	m	[26] 21.4.25	m	m
38	488 (Not Acceptable Here)	[26] 21.4.26	m	m	[26] 21.4.26	m	m
39	489 (Bad Event)	[28] 7.3.2	c3	m	[28] 7.3.2	c3	m
40	491 (Request Pending)	[26] 21.4.27	m	m	[26] 21.4.27	m	m
41	493 (Undecipherable)	[26] 21.4.28	m	m	[26] 21.4.28	m	m
41A	494 (Security Agreement Required)	[48] 2	c5	n/a.	[48] 2	c6	c23
105	5xx response	[26] 21.5	p25		[26] 21.5	p25	
42	500 (Internal Server Error)	[26] 21.5.1	m	m	[26] 21.5.1	m	m
43	501 (Not Implemented)	[26] 21.5.2	m	m	[26] 21.5.2	m	m
44	502 (Bad Gateway)	[26] 21.5.3	o	n/a	[26] 21.5.3	m	n/a

Item	Header	Sending			Receiving		
		Ref.	RFC status	Profile status UNI (Gm)	Ref.	RFC status	Profile status UNI (Gm)
45	503 (Service Unavailable)	[26] 21.5.4	m	m	[26] 21.5.4	m	m
46	504 (Server Time-out)	[26] 21.5.5	m	m	[26] 21.5.5	m	m
47	505 (Version not supported)	[26] 21.5.6	m	m	[26] 21.5.6	m	m
48	513 (Message Too Large)	[26] 21.5.7	m	m	[26] 21.5.7	m	m
49	580 (Precondition Failure)	[30] 8	C35	m	[30] 8	C35	m
106	6xx response	[26] 21.6	p26		[26] 21.6	p26	
50	600 (Busy Everywhere)	[26] 21.6.1	m	m	[26] 21.6.1	m	m
51	603 (Decline)	[26] 21.6.2	c10	m	[26] 21.6.2	m	m
52	604 (Does Not Exist Anywhere)	[26] 21.6.3	m	m	[26] 21.6.3	m	m
53	606 (Not Acceptable)	[26] 21.6.4	m	m	[26] 21.6.4	m	m

Conditions for Table 7-5:

c1: IF A.5/9 THEN m ELSE n/a - - INVITE response.

c2: IF A.5/9 THEN o ELSE n/a - - INVITE response.

c3: IF A.4/20 THEN m ELSE n/a - - SIP specific event notification extension.

c4: IF A.5/19 OR A.5/21 THEN m ELSE n/a - - REGISTER response or SUBSCRIBE response.

c5: IF A.4/37 AND A.4/2 THEN m ELSE n/a - - security mechanism agreement for the session initiation protocol and registrar.

c6: IF A.4/37 THEN m ELSE n/a - - security mechanism agreement for the session initiation protocol.

c7: IF A.4/42 AND (A.5/9 OR A.5/23) THEN m ELSE n/a - - the SIP session timer AND (INVITE response OR UPDATE response).

c8: IF A.4/43 AND A.5/17 THEN o ELSE n/a - - the SIP Referred-By mechanism and REFER response.

c9: IF A.4/43 AND A.5/17 THEN m ELSE n/a - - the SIP Referred-By mechanism and REFER response.

c10: IF A.4/44 THEN m ELSE o - - the Session Initiation Protocol (SIP) "Replaces" header.

c11: IF A.5/9 THE m ELSE n/a - - INVITE response.

c12: IF A.3/4 THEN m ELSE o - - S-CSCF.

c14: IF ACR THEN m ELSE o - - rejecting anonymous requests in the session initiation protocol.

c20: IF A.4/41 THEN m ELSE n/a - - an event state publication extension to the session initiation protocol.

c21: IF A.5/9 OR A.5/9B or A.5/13 OR A.5/15B OR A.5/17 OR A.5/19 OR A.5/21 THEN o ELSE n/a - - INVITE response or MESSAGE response or OPTIONS response or PUBLISH response or REFER response or REGISTER response or SUBSCRIBE response.

c22: IF A.4/57 THEN m ELSE n/a - - managing client initiated connections in SIP.

c23: IF A.4/60 THEN m ELSE n/a - - SIP location conveyance.

c24: IF CDIV THEN m ELSE n/a - -

c25: UE may use it if internal forwarding apply.

c26: IF A.4/75B THEN m ELSE n/a - - a recipient within the framework for consent-based communications in SIP.

c27: IF A.4/75A THEN m ELSE n/a - - a relay within the framework for consent-based communications in SIP.

c28: IF A.4/2 AND A.4/57 THEN m ELSE n/a - - registrar, managing client initiated connections in SIP.

c29: IF A.4/1 AND A.4/57 THEN m ELSE n/a - - client behaviour for registration, managing client initiated connections in SIP.

c30: IF A.4/71 AND (A.3/9B OR A.3/9C OR A.3/13B OR A.3/13C) THEN m ELSE n/a - - addressing an amplification vulnerability in session initiation protocol forking proxies, IBCF (IMS-ALG), IBCF (Screening of SIP signalling), ISC gateway function (IMS-ALG), ISC gateway function (Screening of SIP signalling).
c31: IF A.4/71 THEN m ELSE n/a - - addressing an amplification vulnerability in session initiation protocol forking proxies.
c32: IF A.5/9 AND A.4/81 THEN m ELSE n/a - - INVITE response and 199 (Early Dialog Terminated) response.
c33: IF A.4/13 THEN m ELSE n/a - - SIP INFO method and package framework.
c34: IF A.4/16 OR A.3/6 THEN m ELSE IF A.5/9 THEN o ELSE n/a - - initiating a session which require local and/or remote resource reservation, MGCF, INVITE response.
c35: IF A.4/16 THEN m ELSE n/a - - integration of resource management and SIP.
p21: A.6/2 OR A.6/3 OR A.6/4 OR A.6/5 - - 1xx response.

p22: A.6/6 OR A.6/7 - - 2xx response.

p23: A.6/8 OR A.6/9 OR A.6/10 OR A.6/11 OR A.6/12 OR A.6/13 - - 3xx response.

p24: A.6/14 OR A.6/15 OR A.6/16 OR A.6/17 OR A.6/18 OR A.6/19 OR A.6/20 OR A.6/21 OR A.6/22 OR A.6/22A OR A.6/23 OR A.6/24 OR A.6/25 OR A.6/26 OR A.6/27 OR A.6/28 OR A.6/28A OR A.6/29 OR A.6/29H OR A.629A OR A.6/29B OR A.6/30 OR A.6/31 OR A.6/32 OR A.6/33 OR A.6/34 OR A.6/35 OR A.6/36 OR A.6/436 OR A.6/38 OR A.6/39 OR A.6/40 OR A.6/41 OR A.6/41A. - 4xx response.

p25: A.6/42 OR A.6/43 OR A.6/44 OR A.6/45 OR A.6/46 OR A.6/47 OR A.6/48 OR A.6/49 - - 5xx response.

p26: A.6/50 OR A.6/51 OR A.6/52 OR A.6/53 - - 6xx response.

NOTE: Conditions c1-c21 and p21-p26 are taken over from Annex B.
Note 1: This Response is within SIP for future use defined.
Note 2: These Responses are sent in cases for Registration. Registration in another domain than the home domain is not allowed. Therefore a re INVITE can not be expected.
Note 3: IF send by an UE the NGN may ignore the Response.
Note 4: Normally not send by UE.
Note 5: General a 403 is a Indication that the user is not provisioned within the HSS. Nevertheless if 403 (Forbidden) has been received as a response to a REGISTER request, a further registration attempts shall be done

after 15 sec. In case further 403 response received a with the same URI in the Contact header field Register requests are allowed with a random delay of 30- 60 minutes. . [1].

[1] *This response indicates that the authentication for this contact is not possible because there is no entry stored within the HSS.*

7.3.4 Support of SIP Headers on the UNI (Gm) -Interface

The following SIP Headers shall be supported on the UNI (Gm) -Interface:

Table 7-6: Supported headers

Item	Header	Sending (UE to P-CSCF)			Receiving (P-CSCF to UE)		
		Ref.	Profile status UNI (Gm)	Profile status UE	Ref.	Profile status UNI (Gm)	Profile status UE
1	Accept	[26] 20.1	m	m	[26] 20.1	m	m
2	Accept-Contact	[56B] 9.2	o	o	[56B] 9.2	o	o
3	Accept-Encoding	[26] 20.2	o	o	[26] 20.2	o	o
4	Accept-Language	[26] 20.3	o	o	[26] 20.3	o	o
5	Alert-Info	[26] 20.4	o	m	[26] 20.4	o	m
6	Allow	[26] 20.5, [26] 5.1	m	m	[26] 20.5, [26] 5.1	m	m
7	Allow-Events	[28] 7.2.2	o	m	[28] 7.2.2	o	m
7b	Answer-Mode	[158]	o	o	[158]	o	o
8	Authentication-Info	[26] 20.6	o	m	[26] 20.6	o	m
9	Authorization	[26] 20.7	m	m	[26] 20.7	m	m
10	Call-ID	[26] 20.8	m	m	[26] 20.8	m	m
11	Call-Info	[26] 20.9	o	n/a	[26] 20.9	o	m
12	Contact	[26] 20.10	m	m	[26] 20.10	m	m
13	Content-Disposition	[26] 20.11	o	m	[26] 20.11	o	m
14	Content-Encoding	[26] 20.12	o	m	[26] 20.12	o	m
15	Content-Language	[26] 20.13	o	m	[26] 20.13	o	m
16	Content-Length	[26] 20.14	m	m	[26] 20.14	m	m
17	Content-Type	[26] 20.15	m	m	[26] 20.15	m	m
18	Cseq	[26] 20.16	m	m	[26] 20.16	m	m
19	Date	[26] 20.17	o	m	[26] 20.17	o	m
20	Error-Info	[26] 20.18	o	o	[26] 20.18	o	m
21	Event	[28] 8.2.1	o	m	[28] 8.2.1	o	m
22	Expires	[26] 20.19	o	m	[26] 20.19	o	m
23	From	[26] 20.20	m	m	[26] 20.20	m	m
23A	Geolocation	[89] 3.2	n/a	n/a	[89] 3.2	n/a	n/a

Item	Header	Sending (UE to P-CSCF)			Receiving (P-CSCF to UE)		
		Ref.	Profile status UNI (Gm)	Profile status UE	Ref.	Profile status UNI (Gm)	Profile status UE
23B	Geolocation-Routing	[89] 4.2	n/a	n/a	[89] 4.2	n/a	n/a
24	History-Info	[66] 4.1	n/a	n/a	[66] 4.1	o	m
25	In-Reply-To	[26] 20.21	o	o	[26] 20.21	o	o
26	Join	[61] 7.1	o	o	[61] 7.1	o	o
26b	Max-Breadth	[117]	o	o	[117]	n/a	n/a
27	Max-Forwards	[26] 20.22	m	m	[26] 20.22	m	m
28	MIME-Version	[26] 20.24	o	m	[26] 20.24	o	m
29	Min-Expires	[26] 20.23, [70] 5, 6	o	m	[26] 20.23, [70] 5, 6	o	m
30	Min-SE	[58] 5	o	o	[58] 5	m	m
31	Organization	[26] 20.25	o	o	[26] 20.25	o	o
32	P-Access-Network-Info	[52] 4.4	o	o	[52] 4.4	o	m
32a	P-Answer-State (Note3)	[34] 9.1	n/a	n/a	[34] 9.1	n/a	n/a
33	P-Asserted-Identity	[34] 9.1	n/a	n/a	[34] 9.1	o	m
33a	P-Asserted-Service	[121]	n/a	n/a	[121]	n/a	c1
33b	P-Associated-URI	[52] 4.1	n/a	n/a	[52] 4.1	c9	m
34	P-Called-Party-ID	[52] 4.2	n/a	n/a	[52] 4.2	n/a	c1
35	P-Charging-Function-Addresses	[52] 4.5	n/a	n/a	[52] 4.5	n/a	c1
36	P-Charging-Vector	[52] 4.6	n/a	n/a	[52] 4.6	n/a	c1
36b	P-Early-Media	[109] 8	o	m	[109] 8	o	m
38	P-Media-Authorization	[31] 6.1	o	o	[31] 6.1	o	o
39	P-Preferred-Identity	[34] 9.2	m	m	[34] 9.2	n/a	n/a
39a	P-Preferred-Service	[121] 4.2	n/a	n/a	[121] 4.2	n/a	c1
39b	P-Profile-Key	[97] 5	n/a	n/a	[97] 5	n/a	c1
39c	P-User-Database	[82] 4	n/a	n/a	[82] 4	n/a	c1
40	P-Visited-Network-ID	[52] 4.3	n/a	n/a	[52] 4.3	n/a	c1
40a	Path	[35] 4.2	o	o	[35] 4.2	m	m
41	Priority	[26] 20.26	n/a	n/a	[26] 20.26	n/a	c1

Item	Header	Sending (UE to P-CSCF)			Receiving (P-CSCF to UE)		
		Ref.	Profile status UNI (Gm)	Profile status UE	Ref.	Profile status UNI (Gm)	Profile status UE
41a	Priv-Answer-Mode	[158]	o	o	[158]	o	o
42	Privacy	[33] 4.2	o	m	[33] 4.2	m	m
43	Proxy-Authenticate	[26] 20.27	n/a	n/a	[26] 20.27	m	m
44	Proxy-Authorization	[26] 20.28	m	m	[26] 20.28	n/a	c1
45	Proxy-Require	[26] 20.29	n/a	n/a	[26] 20.29	n/a	c1
46	Rack	[27] 7.2	m	m	[27] 7.2	m	m
47	Reason	[34A] 2	o	n/a	[34A] 2	m	m
48	Record-Route	[26] 20.30	m	m	[26] 20.30	m	m
48A	Recv-Info	[25] 9.2.3	o	o	[25] 5.2.3	o	o
49	Referred-By	[59] 3	o	m	[59] 3	o	m
49b	Refer-Sub	[95]	n/a	n/a	[95]	n/a	C1
49c	Refer-to	[36] 3	c8	c8	[36] 3	c8	c8
50	Reject-Contact	[56B] 9.2	n/a	n/a	[56B] 9.2	n/a	c1
51	Replaces	[60] 6.1	o	o	[60] 6.1	o	o
52	Reply-To	[26] 20.31	o	o	[26] 20.31	o	m
53	Request-Disposition	[56B] 9.1	o	o	[56B] 9.1	o	o
54	Require	[26] 20.32	o	m	[26] 20.32	o	m
55	Retry-After	[26] 20.33	o	o	[26] 20.33	o	m
56	Route	[26] 20.34	m	m	[26] 20.34	n/a	n/a
57	Rseq	[27] 7.1	m	m	[27] 7.1	m	m
58	Security-Client	[48] 2.3.1	n/a	n/a	[48] 2.3.1	n/a	c1
59	Security-Verify	[48] 2.3.1	n/a	n/a	[48] 2.3.1	n/a	c1
60	Server	[26] 20.35	o	o	[26] 20.35	o	o
60b	Service-Route	[38] 5	n/a	n/a	[58] 4	o	m
61	Session-Expires	[58] 4	o	m	[58] 4	o	m
61b	Session-ID	See [Ref_dt1], Note 5	c5	c5	See [Ref_dt1], Note 5	c5	c5
62	SIP-Etag	[70] 11.3.1	o	n/a	[70] 11.3.1	o	o
63	SIP-If-Match	[70] 11.3.2	o	n/a	[70] 11.3.2	o	o

Item	Header	Sending (UE to P-CSCF)			Receiving (P-CSCF to UE)		
		Ref.	Profile status UNI (Gm)	Profile status UE	Ref.	Profile status UNI (Gm)	Profile status UE
64	Subject	[26] 20.36	o	o	[26] 20.36	o	m
65	Subscription-State	[28] 8.2.3	o	m	[28] 8.2.3	o	m
66	Supported	[26] 20.37	o	m	[26] 20.37	o	m
67	Timestamp	[26] 20.38	o	m	[26] 20.38	o	m
68	To	[26] 20.39	m	m	[26] 20.39	m	m
69	Unsupported	[26] 20.40	o	o	[26] 20.40	o	m
70	User-Agent	[26] 20.41	m	m	[26] 20.41	o	m
70a	User-to-User	[126]	o	o	[126]	o	o
71	Via	[26] 20.42	m	m	[26] 20.42	m	m
72	Warning	[26] 20.43	o	o	[26] 20.43	o	m
73	WWW-Authenticate	[26] 20.44	n/a	n/a	[26] 20.44	o	m

Conditions for Table 7-7:

c1: IF received discard Header.

c2: Void.

c3: This Header is only received within a SUBSCRIBE; see table A.5.

c4: IF CCBS THEN m ELSE o.

c5: IF 3PTY (INVITE) OR ECT (REFER) THEN m OR IF end to end correlation (all succeeding SIP messages following Initial Request within the Dialog) THEN o ;
Session ID must contain the hashed call id value.

c6: void.

c7: void.

c8: IF REFER THEN m ELSE n/a.

c9: If Registration THEN m.

Note 1: The use is only foreseen for the Deutsche Telekom domain.
Note 2: void.
Note 3: P-Answer-State header extension to the session initiation protocol for the open mobile alliance push to talk over cellular.
Note 4: Void.
Note 5: This Reference is shown within Section 2 of this document, because this is a requirement of Deutsche Telekom to align Calls all over the network.

7.3.5 MIME Types

The following MIME Types shall be supported:

Table 7-8: Supported MIME Types

Item	MIME Type	Sending (UE to P-CSCF)			Receiving (P-CSCF to UE)		
		Ref.	Profile status UE	Profile status UNI (Gm)	Ref.	Profile status UE	Profile status UNI (Gm)
1	application/vnd.etsi.pstn+xml	Note 1	c1	o	Note 1	c1	o
2	application/x-session-info	3GPP TS 29.163 [24]	o	o	3GPP TS 29.163 [24]	o	o
3	application/vnd.etsi.aoc+xml	3GPP TS 24.647 [10]	m	o	3GPP TS 24.647 [10]	m	o
4	application/simservs+xml	1TR126 [3]	m	o	1TR126 [3]	m	o
5	application/vnd.3gpp.cw+xml	3GPP TS 24.610 [8]	m	o	3GPP TS 24.610 [8]	m	o
6	application/sdp	RFC 2327 [43]	m	m	RFC 2327 [43]	m	m
7	application/pidf+xml	RFC 3863 [57]	n/a	n/a	RFC 3863 [57]	n/a	n/a
8	multipart/mixed		m	m		m	m
9	application/rlmi+xml		o	o		o	o
10	application/watcherinfo+xml	RFC 3858 [56]	n/a	n/a	RFC 3858 [56]	n/a	n/a
11	text/plain	RFC 2046 [40]	n/a	n/a	RFC 2046 [40]	n/a	n/a
12	image/t.38	RFC 3362 [52]	o	o	RFC 3362 [52]	o	o
13	application/simple-message-summary	RFC 3842 [55]	c2	o	RFC 3842 [55]	c2	o
14	other MIME types (Note 2)		n/a	n/a		n/a	n/a
15	encrypted MIME TYPE		n/a	n/a		n/a	n/a

NOTE: The references in this table are listed in the present document (see clause 2).
Conditions:

c1: IF ISDN is supported THEN m ELSE n/a.

c2: IF MWI is supported THEN m ELSE n/a.

Note 1: 3GPP the definition is within 3GPP TS 29.163 [24].

Note 2: Other MIME Types can be received and must be discarded in case where no content disposition header is present the MIME is not known.

7.3.6 SDP Types

Table §A.318 modified as follows:

§A.3.2.2 SDP types

Table A.318: SDP types

Item	Type	Sending			Receiving		
		Ref.	RFC status	Profile status	Ref.	RFC status	Profile status
	Session level description						
1	v= (protocol version)	[39] 5.1	m	m	[39] 5.1	m	m
2	o= (owner/creator and session identifier)	[39] 5.2	m	m	[39] 5.2	m	m
3	s= (session name)	[39] 5.3	m	m	[39] 5.3	m	m
4	i= (session information)	[39] 5.4	o	c2	[39] 5.4	m	c3
5	u= (URI of description)	[39] 5.5	o	c4	[39] 5.5	o	n/a
6	e= (email address)	[39] 5.6	o	c4	[39] 5.6	o	n/a
7	p= (phone number)	[39] 5.6	o	c4	[39] 5.6	o	n/a
8	c= (connection information)	[39] 5.7	c5	c5	[39] 5.7	m	m
9	b= (bandwidth information)	[39] 5.8	o	o (NOTE 1)	[39] 5.8	m	m
	Time description (one or more per description)						
10	t= (time the session is active)	[39] 5.9	m	m	[39] 5.9	m	m
11	r= (zero or more repeat times)	[39] 5.10	o	c4	[39] 5.10	o	n/a
	Session level description (continued)						
12	z= (time zone adjustments)	[39] 5.11	o	n/a	[39] 5.11	o	n/a

13	k= (encryption key)	[39] 5.12	x	x	[39] 5.12	n/a	n/a
14	a= (zero or more session attribute lines)	[39] 5.13	o	o	[39] 5.13	m	m
	Media description (zero or more per description)						
15	m= (media name and transport address)	[39] 5.14	o	o	[39] 5.14	m	m
16	i= (media title)	[39] 5.4	o	c2	[39] 5.4	o	c3
17	c= (connection information)	[39] 5.7	c1	c1	[39] 5.7	c1	c1
18	b= (bandwidth information)	[39] 5.8	o	o (NOTE 1)	[39] 5.8		
19	k= (encryption key)	[39] 5.12	x	x	[39] 5.12	n/a	n/a
20	a= (zero or more media attribute lines)	[39] 5.13	o	o	[39] 5.13	m	m

Conditions for Table A.318:

c1: IF (A.318/15 AND NOT A.318/8) THEN m ELSE (IF (A.318/15 AND A.318/8) THEN o ELSE n/a - - “c=” contained in session level description and SDP contains media descriptions.

c2: IF A.3A/6 THEN x ELSE o - - MGCF.

c3: IF A.3A/6 THEN n/a ELSE m - - MGCF.

c4: IF A.3A/6 THEN x ELSE n/a - - MGCF.

c5: IF A.318/17 THEN o ELSE m - - "c=" contained in all media description.

NOTE 1: The UE may use b=TIAS and b=AS as described in RFC 3890 [152]. For "video" and "audio" media types that utilize RTP/RTCP, and if the UE is configured to request an RTCP bandwidth level different than the default RTCP bandwidth as specified in RFC 3556 [56], then the UE shall include the “b=” media descriptors with the bandwidth modifiers “RS” and “RR”. For other media types, the UE may include the “b=” media descriptor with the bandwidth modifiers “RS” and “RR”.

Prerequisite A.318/14 OR A.318/20 - - a= (zero or more session/media attribute lines)

Table A.319: zero or more session / media attribute lines (a=)

Item	Field	Sending			Receiving		
		Ref.	RFC status	Profile status	Ref.	RFC status	Profile status
1	category (a=cat)	[39] 6	c8	c8	[39] 6	c9	c9
2	keywords (a=keywds)	[39] 6	c8	c8	[39] 6	c9	c9
3	name and version of tool (a=tool)	[39] 6	c8	c8	[39] 6	c9	c9
4	packet time (a=ptime)	[39] 6	c10	c10	[39] 6	c11	c11
5	maximum packet time (a=maxptime)	[39] 6 (NOTE 1)	c10	c10	[39] 6 (NOTE 1)	c11	c11
6	receive-only mode (a=recvonly)	[39] 6	o	o	[39] 6	m	m

7	send and receive mode (a=sendrecv)	[39] 6	o	o	[39] 6	m	m
8	send-only mode (a=sendonly)	[39] 6	o	o	[39] 6	m	m
8A	Inactive mode (a=inactive)	[39] 6	o	o	[39] 6	m	m
9	whiteboard orientation (a=orient)	[39] 6	c10	c10	[39] 6	c11	c11
10	conference type (a=type)	[39] 6	c8	c8	[39] 6	c9	c9
11	character set (a=charset)	[39] 6	c8	c8	[39] 6	c9	c9
12	language tag (a=sdplang)	[39] 6	o	o	[39] 6	m	m
13	language tag (a=lang)	[39] 6	o	o	[39] 6	m	m
14	frame rate (a=framerate)	[39] 6	c10	c10	[39] 6	c11	c11
15	quality (a=quality)	[39] 6	c10	c10	[39] 6	c11	c11
16	format specific parameters (a=fmtp)	[39] 6	c10	c10	[39] 6	c11	c11
17	rtpmap attribute (a=rtpmap)	[39] 6	c10	c10	[39] 6	c11	c11
18	current-status attribute (a=curr)	[30] 5	c1	c1	[30] 5	c2	c2
19	desired-status attribute (a=des)	[30] 5	c1	c1	[30] 5	c2	c2
20	confirm-status attribute (a=conf)	[30] 5	c1	c1	[30] 5	c2	c2
21	media stream identification attribute (a=mid)	[53] 3	c3	c3	[53] 3	c4	c4
22	group attribute (a=group)	[53] 4	c5	c5	[53] 3	c6	c6
23	setup attribute (a=setup)	[83] 4	c7	c7	[83] 4	c7	c7
24	connection attribute (a=connection)	[83] 5	c7	c7	[83] 5	c7	c7
25	candidate IP addresses (a=candidate)	[99]	c12	c12	[99]	c13	c13
26	floor control server determination (a=floorctrl)	[108] 4	c14	c14	[108] 4	c14	c14
27	conference id (a=confid)	[108] 5	c14	c14	[108] 5	c14	c14
28	user id (a=userid)	[108] 5	c14	c14	[108] 5	c14	c14
29	association between streams and floors (a=floorid)	[108] 6	c14	c14	[108] 6	c14	c14
30	RTCP feedback capability attribute (a=rtcp-fb)	[135] 4.2	c15	c15	[135] 4.2	c15	c15
31	extension of the rtcp-fb attribute (a=rtcp-fb)	[136] 7.1	c15	c15	[136] 7.1	c15	c15
32	supported capability negotiation extensions (a=csup)	[137] 6.1	c16	c16	[137] 6.1	c16	c16
33	required capability negotiation extensions (a=creq)	[137] 6.1	c16	c16	[137] 6.1	c16	c16
34	attribute capability (a=acap)	[137] 6.1	c16	c16	[137] 6.1	c16	c16
35	transport protocol capability (a=tcap)	[137] 6.1	c16	c16	[137] 6.1	c16	c16
36	potential configuration (a=pcfg)	[137] 6.1	c16	c16	[137] 6.1	c16	c16
37	actual configuration (a=acfg)	[137] 6.1	c16	c16	[137] 6.1	c16	c16

38	connection data capability (a=ccap)	[156] 5.1	c17	c17	[156] 5.1	c18	c18

Conditions for Table A.319:

c1: IF A.317/22 AND A.318/20 THEN o ELSE n/a - - integration of resource management and SIP, media level attribute name "a=".

c2: IF A.317/22 AND A.318/20 THEN m ELSE n/a - - integration of resource management and SIP, media level attribute name "a=".

c3: IF A.317/23 AND A.318/20 THEN o ELSE n/a - - grouping of media lines, media level attribute name "a=".

c4: IF A.317/23 AND A.318/20 THEN m ELSE n/a - - grouping of media lines, media level attribute name "a=".

c5: IF A.317/23 AND A.318/14 THEN o ELSE n/a - - grouping of media lines, session level attribute name "a=".

c6: IF A.317/23 AND A.318/14 THEN m ELSE n/a - - grouping of media lines, session level attribute name "a=".

c7: IF A.317/26 AND A.318/20 THEN m ELSE n/a - - TCP-based media transport in the dession description protocol, media level attribute name "a=".

c8: IF A.318/14 THEN o ELSE x - - session level attribute name "a=".

c9: IF A.318/14 THEN m ELSE n/a - - session level attribute name "a=".

c10: IF A.318/20 THEN o ELSE x - - media level attribute name "a=".

c11: IF A.318/20 THEN m ELSE n/a - - media level attribute name "a=".

c12: IF A.317/27 AND A.318/20 THEN o ELSE n/a - - candidate IP addresses, media level attribute name "a=".

c13: IF A.317/27 AND A.318/20 THEN m ELSE n/a - - candidate IP addresses, media level attribute name "a=".

c14: IF A.317/28 AND A.318/20 THEN m ELSE n/a - - session description protocol format for binary floor control protocol streams, media level attribute name "a=".

c15: IF (A.317/29 AND A.318/20) THEN m ELSE n/a - - extended RTP profile for real-time transport control protocol (RTCP)-based feedback (RTP/AVPF), media level attribute name "a=".

c16: IF A.317/30 AND A.318/20 THEN m ELSE n/a - - SDP capability negotiation, media level attribute name "a=".

c17: IF A.317/32 AND A.318/20 THEN o ELSE n/a - - miscellaneous capabilities negotiation in the Session Description Protocol (SDP), media level attribute name "a=".

c18: IF A.317/32 AND A.318/20 THEN m ELSE n/a - - miscellaneous capabilities negotiation in the Session Description Protocol (SDP), media level attribute name "a=".

NOTE 1: Further specification of the usage of this attribute is defined by specifications relating to individual codecs.

7.4 SIP User Agent (UA)

NOTE: The references in the following tables are listed in the present document (see clause 2).

7.4.1 Supported SIP Signalling Transport Protocols in UA

The following SIP Signalling Transport Protocols shall be supported:

Table7-9: Supported SIP Signalling Transport Protocols in UA

Protocol (NOTE)	Specification	Ref.	Support
UDP	RFC 0768/STD006	[34]	m
TCP	RFC 0793/STD007	[37]	m
TLS	RFC 2246	[42]	o
SCTP	ETSI TS 102 144	[5]	o
IPSec	RFC 2411	[44]	o
NOTE: The following combinations shall be possible to configure: ■ SIP over UDP ■ SIP over TCP without TLS ■ SIP over TCP with TLS			

7.4.2 Support of IPv4 und IPv6

Table 7-10: RFC for support of IPv4 and IPv6

Specification	Title	Ref.	Support
RFC 0791	Internet Protocol, Version 4	[35]	m
RFC 0792	Internet Control Message Protocol	[36]	m
RFC 1035	Domain names implementation and specification	[38]	m
RFC 2460	Internet Protocol, Version 6	[46]	m
RFC 2782	A DNS RR for specifying the location of services (DNS SRV)	[47]	m
RFC 2915	The Naming Authority Pointer (NAPTR) DNS Resource Record	[48]	o
RFC 3596	DNS Extensions to Support IP Version 6	[54]	m
RFC 4443	Internet Control Message Protocol (ICMPv6) for the Internet Protocol Version 6 (IPv6) Specification; March 2006	[64]	m

RFC4884	Extended ICMP to Support Multi-Part Messages, April 2007	[70]	m

Table 7-11: DNS Records

Procedure	Specification	Ref.	Support
DNS SRV-record	RFC 3263	[50]	m
DNS NAPTR-record	RFC 3263	[50]	o

Table 7-12: Procedures for SIP-Server Localisation

Protocol	Remarks	Support
Static Routing	Preconfigured SIP server ip address in UA.	m
DNS A-record	Preconfigured SIP server fully qualified domain name in UA.	m
DNS AAAA-record	Preconfigured SIP server fully qualified domain name in UA.	m

7.4.3 Video Codec Transport Procedures

Table7-13: Specifications Video Codec Transport Procedures

Specification	Title	Ref.	Support
RFC 2190	RTP Payload Format for H.263 Video Streams	[41]	m
RFC 2429	RTP Payload Format for the 1998 Version of ITU-T Rec. H.263 Video (H.263+)	[45]	m
RFC 3984	RTP Payload Format for H.264 Video	[60]	o
RFC 2032	RTP Payload Format for H.261 Video Streams	[39]	o

7.4.4 Real-time Transport Procedures

Table7-14: Specifications Real-time Transport Procedures

Specification	Title	Ref.	Support
RFC 3550	RTP: A Transport Protocol for Real-Time Applications; July 2003	[53]	M
RFC 4040	RTP Payload Format for a 64 kbit/s Transparent Call; April 2005 (see Note)	[62]	c1
Conditions: c1: If ISDN interworking then m else o. NOTE1: This protocol is applicable to carry 64 kbit/s channel data transparently in RTP packets, using a pseudo-codec called "Clearmode" and is used in case of ISDN accesses via IADs, only. NOTE2: Fragmented IP packets are not supported by the NGN platform of Deutsche Telekom. If the UA chooses to send RTCP/SDES packets it shall not send the UA’s public IP address.			

8 SIP terminals

SIP terminals can be connected to the NGN platform of Deutsche Telekom in two ways:

- By connecting directly to the Internet, e.g. via a Network Access Server (NAS)
- As part of a local network whose local router (NAT Router) is connected to the Internet by means of a Network Access Server

SIP terminals can use IPv4 or IPv6 to communicate with the call control. The communication (Signalling and Media) of an UA shall be based either on IPv4 or IPv6. If both protocols available, IPv6 shall be preferred.

8.1 Direct connection

The SIP Client is either a hard phone (physical IP telephone) or a soft phone (software client). This type of connection is characterized by the fact that the SIP Client can be reached via the IP address assigned by the NAS or via a fixed IP address.

8.2 Local network

One or more SIP terminals can also be part of a local network. In this environment, every SIP Client has its own private IPv4 address in the local network. Starting from the NGN platform of Deutsche Telekom, both SIP Clients are talked to with the same external address. However, the NGN platform of Deutsche Telekom uses different ports here for the individual clients here. A local router is responsible for converting the external IPv4 address to the internal IPv4 address and transport protocol port number. This conversion is known as Network Address (Port) Translation (NAPT). The rules for NAPT shall be stored in the local router.

Where necessary, a SIP Client enters its internal IP address in the SIP messages. Taking the rules described above, the NGN platform of Deutsche Telekom translates the internal IP addresses to the external IP addresses.

In order to eliminate the necessity of IPv6 packet fragmentation for IPv4 packets that have to be translated by the Call Control into IPv6 packets, a SIP client shall not send RTP messages with maximum transmission units (MTU) larger than 1260 bytes. When choosing 1260 bytes as MTU for IPv4 packets the IPv4 to IPv6 translation within the Call Control will lead to in maximum 1280 bytes large IPv6 packets - a MTU size that can be handled by all IPv6 capable link types without IPv6 packet fragmentation.

Over and above that an adoption of the IPv4 MTU of 576 bytes for IPv4 and IPv6 RTP packets is strongly recommended for all SIP clients. In order to minimize delay, jitter and packet loss the fragmentation of IP packets shall be be avoided. In the IPv4 header of an RTP packet the value of the DF bit in the Flag field shall be "1" (Don't Fragment).
When receiving an ICMPv4 'Destination Unreachable' message with error code '4' (fragmentation needed and DF set) the SIP client shall use the 'Next-Hop MTU' info (RFC4884) [70] of this ICMP error packet to limit its MTU to the given value."

When the SIP terminal uses IPv6, the IPv6 address is a global address. NAPT shall not be used.

"When using IPv6 the SIP client shall limit its IPv6 MTU size for RTP packetsto 1280 bytes. Neverthless a MTU size of 576 bytes is also recommended for IPv6 based RTP packets."

The local router can be one of the following devices, for example:

- DSL router

- WLAN router

8.3 Support of IPv6 by the UE

The set of IPv6 functionality a 3GPP UE will require is dependent on the services (IMS, Packet Streaming etc.) it will use. As a minimum, a 3GPP UE shall comply with the Basic IP group of specifications as defined in RFC 3316 [51] (chapter 2). This IPv6 functionality is sufficient to provide compatibility towards IPv6 entities external to 3GPP. To ensure network performance the UE shall limit the sending packet size of media data to 1280 octets; to ensure this and the requirements of RFC 3316 [51], the UE has to support RFC 4443 [64] (chapter 3). Neverthless a MTU size of 576 bytes is also recommended for IPv6 based RTP packets."

According to the procedures defined in TS 23.060 [18] and in TS 23.401 [20], when a UE is assigned an IPv6 prefix, it can change the global IPv6 address it is currently using via the mechanism defined in RFC 3041 [49], or similar means, without updating the PS domain. Any application that requires full IP address knowledge shall provide a mechanism to get the latest IPv6 address when the IPv6 address in the UE has been changed.

An example of such means is defined in TS 23.228 [19].

NOTE: RFC 3316 [51] does not make any recommendations on preferred transition and interoperability mechanisms between IPv4 and IPv6.

8.4 Network Access

Description of minimum technical requirements for potential QoS transport (up-/downstream).

8.4.1 General User Equipment (UE) requirements

The packetization size has to be equal or more than 20 ms to ensure two high quality media streams using the codec G.711a or G.722 over the guaranteed bandwidth on the SIP (Gm) interface of Deutsche Telekom.

Due to this requirement, the UE shall support the following features:

- The packetization size shall be increased to 30 ms if necessary;
- The switching between 10 ms, 20 ms or 30 ms packetization size might happen automatically according to the available bitrate.

8.4.2 Traffic Classes in Layer 3

The UE uses the following traffic classes at Layer 3 (according to the Architecture of T-Home)

- Voice Control Class 6 (DSCP 110 000)
- Voice Bearer Class 5 (DSCP 101 110)
- PPP/PPPoE Control Traffic Class 6
- Best Effort Class 0 (DSCP 000 000)

Prioritisation & marking should always be processed internally at the UE, even when the network connection does not support VLAN tags for prioritisation.

The traffic classes used shall be the same, independently whether IPv4 or IPv6 is used.

Authorized traffic & signalling should be marked, tagged and prioritized at the UE towards to the T-Home platform. All other traffic should be marked and scheduled as Best Effort (assumption is that LAN bandwidth is larger than towards T-Home platform).

8.4.3 Service Creation

Note that this chapter gives only a functional description of possible interaction with the platform. The QoS agreement respectively the service creation depends on the Service Level Agreement. Deutsche Telekom gives no service guarantee neither for the received traffic class or for the transparent transport of code points nor for VLANs or other protocol types.

8.5 Number handling by the UE

Generally, concerning UEs there are no requirements for any kind of number handling.

The UE shall not alter the dialled number by the user when sending to the NGN platform of Deutsche Telekom.

All phone numbers beginning with “11” (short codes, e.g. 112 or 110 or 11833; see also national number plan of Germany) shall not be manipulated by any UE; these numbers shall be sent out without neither any Country Code (CC) nor any National Destination Code (NDC).

NOTE: Each connection to a 11xyz destination including a CC and/or a NDC has to be rejected in the NGN platform of Deutsche Telekom according to regulation requirements.

8.6 Support of NAT traversal by the UE

In general endorsement of 3GPP TS 24.229 Annex F and Annex K is valid with the following extensions.

For keeping the NAT-Pinholes open, and depending on the transport protocol for SIP signaling, the following packet type shall be sent by the UE to the next hop SIP port:

- Basic keepalive,
- Empty UDP messages,
- TCP single CRLF.

For keeping the NAT-Pinholes open for media, one of the following packet types shall be sent by the UE to the next hop session media ports[1]:

- Empty RTP packets, as follows

  <u>To keep NAT bindings and firewall pinholes open with uni-directional RTP traffic and enable the C-BGF to perform address latching, the UE shall send keep alive messages for each media stream. These messages shall be sent regardless of whether the media stream is currently inactive, send only, recvonly or sendrecv. It is recommended that the keepalive message be an empty (no payload) RTP packet with a payload type of 20</u>

[1] Per session, keepalive packets shall be sent for each media stream and media stream component, i.e. RTP and RTCP, if used.

as long as the other end has not negotiated the use of this value. If this value has already been negotiated, then some other unused static payload type from Table 5 of RFC 3551 [89] shall be used.

- STUN BINDING Indication packets, according to 3GPP TS 24.229 Annex K.2.1, which relates to UE usage of Interactive Connectivity Establishment (ICE).

Re-Register messages shall not be used for maintaining NAT Pinholes.

9 Interworking requirements for SIP user equipment (UE)

The interworking requirements for SIP user equipment (e.g. IAD) are specified in separate documents. The referenced documents can be interpreted as recommendations for SIP terminal developers and vendors.

9.1 Analogue (POTS) – SIP basic interworking requirements

The Analogue /(POTS) – SIP basic interworking requirements are contained in the technical specification 1 TR 126 [3].

9.2 DSS1 – SIP basic interworking requirements

The DSS1 – SIP basic interworking requirements are contained in the technical specification 1 TR 127 [4].

Annex A Void

Annex B 3GPP TS 24.229 V11.6.0 (2012-12): 3rd Generation Partnership Project; Technical Specification Group Core Network and Terminals; IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 (Release 11) Modified version for SIP (Gm) interfaces provided by Deutsche Telekom only !

The protocol specification of the SIP (Gm) interface for SIP UE intended to be connected to xDSL accesses of Deutsche Telekom are described in the specifically modified 3GPP Standard TS 24.229 11.6.0 (2129-12) [21]. The modifications in this 3GPP standard assure the compatibility with the NGN platform of Deutsche Telekom.

The modified specification is available as a PDF file with the following file name:

- 1TR114 Annex B_V020000_TS24229-b60.pdf

NOTE: The modified text that is added or deleted is shown as cursive and underlined (*example for added text*) or cursive and stricken (~~*example for stricken text*~~).

Annex C Service functionality requirements

This annex C describes the service functionality requirements as a recommendation for the behaviour of SIP user equipments connected to the NGN platform of Deutsche Telekom.

The following services are based on referenced ETSI or 3GPP standards. Deviations to these standards are explicitly defined in the present document. These services are defined as “Simulation Services” for the NGN platform of Deutsche Telekom.

Only the relevant clauses with the used options of these standards are provided in this section.

The actual numbering of the referenced documents are kept in the following with a leading “§” sign. Further on the references given in these clauses refer to the according document.

Annex C.1 Global definitions

For all services the “Ut” interface (XML) is currently not used by the NGN platform of Deutsche Telekom.

A later implementation is not generally excluded.

Note 1: The relevant service code commands (SCC) for provision/withdrawal, registration/erasure, activation/de-activation, interrogation and invocation are provided in Annex D.

Note 2: The activation/de-activation/ registration/erasure and interrogation of most of the following services are also possible via other methods (e.g. Web interface).

Annex C.2 Simulation services

The Services Communication HOLD (Section C.2.6), Communicating Waiting (CW) (Section C.2.8), TOGGEL and CONF (Section C.2.2) shall be implemented as a End Client Service feature and as described within this specification.

The network centric feature logic for HOLD, CW, TOGGEL and CONF is currently not available; therefore these features must be implemented locally on the SIP Client.

C.2.1 CDIV (Communication Diversion)

The relevant standard for the CDIV services is 3GPP TS 24.604 [12]. The determinations of options and deviations to this standard are defined in the following sections.

C.2.1.1 Currently supported CDIV services

Various methods of communication diversion are implemented in the NGN platform of Deutsche Telekom.

The following CDIV services are currently supported by Deutsche Telekom:

- CFU (Communication Forwarding Unconditional)1,
- CFB (Communication Forwarding Busy),
- CFNR (Communication Forwarding No Reply),
- CFNL (Communication Forwarding on Not Logged-in),
- CFNRc (Communication Forwarding on Subscriber Not Reachable) [2]

C.2.1.2 Currently not supported CDIV services

The following CDIV services are currently not supported in the NGN platform of Deutsche Telekom:

- CDIVN (Communication Diversion Notification)

C.2.2 CONF (Conference)

The relevant standard for the CONF service is 3GPP TS 24.605 [16]. The current implementation does not support this Specification.

The 3PTY CONF MUST be supported by the VGW itself.

The context (focus) building of the RTP session of Party A, B and C has to apply within the VGW.

C.2.3 MWI (Message Waiting Indication)

The relevant standard for the MWI service is 3GPP TS 24.606 **Fehler! Verweisquelle konnte nicht gefunden werden.**.

Deutsche Telekom network does not support MWI. If MWI is implemented as an Option it shall be deactivated per default.

C.2.4 OIP/OIR (Originating Identification Presentation and Originating Identification Restriction)

The relevant standard for the OIP/OIR service is 3GPP TS 24.607 [7]. The determinations of options and deviations to this standard are defined in the following sections.

§4.5.2.1 Actions at the originating UE

As part of basic communication, the originating UE may insert a P-Preferred-Identity header field in any initial SIP request for a dialog or in any SIP request for a standalone transaction as a hint for creation of a public user identity as described in 3GPP TS 24.229 [3].

NOTE 1: According 3GPP TS 24.229 [3], the UE can include any of the following in the P-Preferred-Identity header field:

a public user identity which has been registered by the user;

---

[1] While CFU services are activated the special dial tone shall be played in off-hook state (e.g. after picking up the receiver) instead of normal dial tone to remember the user that no incoming calls will be received.

[2] Communication Forwarding on Subscriber Not Reachable shall be seen as “Anrufweiterschaltung sofort im Störungsfall” (CDIV:CFU Stö). In this case the communication is forwarded to an announcement.

a public user identity returned in a registration-state event package of a NOTIFY request as a result of an implicit registration that was not subsequently deregistered or has expired; or

any other public user identity which the user has assumed by mechanisms outside the scope of 3GPP TS 24.229 [3] to have a current registration.

If the originating user wishes to override the default setting of "presentation not restricted" of the OIR service in temporary mode:

- ~~*The originating UE shall include an "anonymous" From header field. The convention for configuring an anonymous From header field described in IETF RFC 3323 [6] should be followed; i.e. From: "Anonymous" <sip:anonymous@anonymous.invalid>;tag= xxxxxxx.*~~

  For the NGN platform of Deutsche Telekom the sending of From: "Anonymous" <sip:anonymous@anonymous.invalid>;tag=xxxxxxx is not allowed.

- If only the P-Asserted-Identity needs to be restricted the originating UE shall include a Privacy header field [6] set to "id" in accordance with IETF RFC 3325 [7].

  This Option is not used within the NGN platform of Deutsche Telekom. At least the privacy header must be identified by the AS and must be completed with the other privacy values "user" and "header".

- If all headers containing private, that the UE cannot anonymize itself, information need to be restricted the originating UE shall include a Privacy header field set to "header" in accordance with IETF RFC 3323 [6].

NOTE 2: The P-Asserted-Identity header field is removed by the Privacy header field value "header". The From header field is not anonymized by the Privacy header field value "header".

NOTE 3: In each case the From header shall include a PUID.

*The following section is only valid for **CLIR 1** Service (See Section D.2):*

If the originating user wishes to override the default setting of "presentation restricted" of the OIR service in temporary mode:

- The originating UE shall include a Privacy header field of privacy type "none" in accordance with 3GPP TS 24.229 [3] (IETF RFC 3323 [6]) *and CLIR 1*.

*In normal presentation cases (**CLIP permanent**) the setting of "none is "forbidden.*

If the display name is used then it shall have the same content as the accompanying uri. The device MUST NOT allow a user defined "display name".

C.2.5 TIP/TIR (Terminating Identification Presentation and Terminating Identification Restriction)

The relevant standard for the TIP/TIR service is 3GPP TS 24.608 [13]. The determinations of options and deviations to this standard are defined in the following sections.

§4.5.2.1 Actions at the originating UE

A UE that supports the TIP service signalling procedures shall support the receipt, in SIP responses to SIP requests initiating a dialog or for standalone transactions, one or more P-Asserted-Identity headers, each one containing a network-provided identity information of the terminating user.

If no P-Asserted-Identity header fields are present, but a Privacy header field set to "id" was present, then the network-provided identity information was withheld due to presentation restriction.

If neither P-Asserted-Identity header fields nor a Privacy header fields set to "id" are present, then the network-provided identity information was not available (due, for example, to interworking with other networks).

Once a 2xx response is received, the P-Asserted-Identity header field of the first 2xx response is used, e.g. when presenting the identity to the user.

NOTE 1: Any P-Asserted-Identity received in a provisional response is outside the scope of this service.

If the originating user is subscribed to the TIP services and wants to receive changes of the terminating identity in the dialog the UE shall add the option tag "from-change" to the Supported header field in the initial request.

NOTE 2: This option tag is used to indicate that a UA supports changes to URIs in From and To header fields during a dialog. Not setting this indication shows that the UE is not supporting this procedure.

C.2.6 HOLD (Communication HOLD)

The relevant standard for the HOLD service is 3GPP TS 24.610 [8].

The actions in Section "4.5.2.1 Actions at the invoking UE" shall apply with the following general modification:

- Only re-INVITE shall be used.
- The bandwidth indicated by the bandwidth attribute according to RFC 3890 [58] shall be used.

C.2.7 ACR and CB (Anonymous Communication Rejection and Communication Barring)

The relevant standard for the ACR and CB service is 3GPP TS 24.611 [9].

C.2.8 CW (Communication Waiting)

Based on the VGW (IAD) configuration and it must be possible to activate a local/terminal based CW on a busy line if an INVITE without a CW indication is received by the VGW. The default for local/terminal based CW is activated.

The relevant standard for the CW service is 3GPP TS 24.615 [22].

The sections "4.5.5.1 Actions at the UE of user C" and "4.5.5.3 Actions at the UE of user B" in 3GPP TS 24.615 [22] shall apply.

The actions described within section "4.5.5.2 Actions at the AS of user B" of 3GPP TS 24.615 [22] are currently not executed within the IMS of DT. Nevertheless future network configuration could change, thus UE SHOULD be prepared to receive a Content-Type header field set to "application/vnd.3gpp.cw+xml" and apply the regarding procedures.

To avoid loss in QoS, a maximum of 2 active and 2 waiting communications shall apply.

Note: To avoid "ghost ringing", an UE while in idle state should not signal an incoming communication (by any acoustical or visual signal, e.g. ringing signal or display information) based on a SIP request INVITE with XML: call-waiting-indication.

C.2.9 MCID (Malicious Communication Identification)

The relevant standard for the MCID service is 3GPP TS 24.616 [14]. The determinations of options and deviations to this standard are defined in the following sections:

§4.5.2.12.1 Subscriber has a temporary subscription

In case of invoking the MCID service the UE shall sent a ~~*Re-INVITE*~~ *initial INVITE including a Service Code Command*.

Note: The Service Code Commands for invocation are defined in Annex D.
~~*As a network operator option including a XML-MIME with XML mcid body with MCID XML Request schema containing a McidRequestIndicator set to 1 could be sent.*~~

C.2.10 ECT (Explicit Communication Transfer)

The relevant standard for the ECT service is 3GPP TS 24.629 [15]. The determinations of options and deviations to this standard are defined in the following sections.

§4.4 Coding requirements

A user agent that wishes to use the ECT service (to act as a transferor):

- Shall support the REFER method as a client as specified in RFC 3515 [2].
- Shall support the Referred-By header as specified in RFC 3892 [3].

A user agent that is the transferred party in a communication transfer (acts as the transferee):

- ~~*Shall support the REFER method as a server as specified in RFC 3515 [2].*~~
- ~~*Shall*~~ *May* support the Referred-By header as specified in RFC 3892 [3].
- ~~*Shall*~~ *May* support Replaces header field as a client as specified in RFC 3891 [4].

A user agent that is the transfer target in a communication transfer:

- May support the Referred-By header as a server as specified in RFC 3892 [3].
- May support the Replaces header as a server as specified in RFC 3891 [4].

§4.5.2.1 Actions at the transferor UE

A UE that initiates a transfer operation shall:

- Issue a REFER request in the original communications dialog, where:
  - The request URI shall contain the SIP URI of the transferee as received in the Contact header field.
  - The Refer-To header field shall indicate the public address of the transfer Target.
  - ~~*If the transferor UE has a consultation communication with the transfer Target, a*~~ *A* Replaces header field parameter shall be added to the Refer-To URI together with a Require=replaces header field parameter.

NOTE: The NGN platform of Deutsche Telekom supports only consultative ECT for confirmed communications.

- The Referred-By header field may indicate the identity of the transferor.
- The REFER shall contain a Session-ID header field.

After the REFER request is accepted by the other end with a 202 (Accepted) response, the transferor UE should get notifications of how the transferee's communication setup towards the transfer Target is progressing.

When a NOTIFY request is received on the REFER dialog that indicates that the transferee and the transfer Target have successfully setup a communication, the transferor UE shall~~may~~ terminate the original communication with the transferee UE, by sending a BYE message on the original dialog.

§4.5.2.5 Actions at the transferee UE

Basic communication procedures according to 3GPP TS 24.229 [2] shall apply.

~~When a REFER request is received in the context of a call transfer scenario (see clause 4.5.2.4.1), the transferee UE shall perform the following steps:~~

~~1) apply the procedure for holding the active communication with the transferor as described in TS 183 010 [8] clause 4.5.2.1; and~~

~~2) apply normal REFER handling procedures according to ES 283 003 [1].~~

§A.2 Consultative transfer

Note: The HOLD call flow as described in Annex A, A.2 shall not apply; the call flow as defined in Annex B of 1TR126 [3] shall apply accordingly.

C.2.11 CCBS/CCNR (Completion of Communications to Busy Subscriber/ Completion of Communications by No Reply)

The relevant standard for the CCBS service is 3GPP TS 24.642 [23]. The determinations of options and deviations to this standard are defined in the following sections.

C.2.11.1 Global modification

The NGN platform of Deutsche Telekom sends a 199 for the release of early dialogs. A further 18x response (e.g. 180 in case of CCNR activation rejection) may be sent afterwards.

§4.5.4.1 Actions at the originating UE

Basic call procedures ~~and in case of a call-completion recall initiated by a REFER request, normal REFER method handling procedures~~ according to 3GPP TS 24.229 [2] shall apply.

For invoking and revoking of the call completion services, announcement procedures according to 3GPP TS 24.628 [3] and inband-interaction procedures shall~~should~~ be used.

C.2.12 AOC (Advice Of Charge)

The relevant standard for the AOC service is 3GPP TS 24.647 [10].

Advice of charge (AOC) information shall be treated according [10], Annex D „AOC XML Schema“.

During a communication different AOC types (e.g. AOC-D) can be used to convey tariff and/or cost information. All types of AOC described in [10] are supported, depending on the requested service options on the subscriber line.

AOC tariff information will only be conveyed in monetary format (currency), but not in non-monetary format (units).

If two communications have been established via the UE (e.g. in case of HOLD, Toggle or 3PTY), the UE receives separate AOC XML's for each communication.

C.2.12.1 AOC-S

The AOC-S service provides tariff information at the beginning of a communication and when tariff changes occur during a communication.

AOC-S information can be conveyed in the following SIP messages:

- reliable 18x message
- 200OK (INVITE)
- INFO

The AOC-S tariff-information can be expressed as:

- Monetary price per time unit and time unit (basic-type->price-time);
- Monetary flat rate (communication-setup->flat-rate).

C.2.12.2 AOC-D

The AOC-D service provides tariff information about the recorded charges during the active phase of a communication.

Only accumulated charges in monetary format are transmitted.

AOC-D information can be conveyed in the following SIP messages:

- INFO
- BYE
- Final response (BYE)

C.2.12.3 AOC-E

The AOC-E service provides tariff information about the recorded charges for a communication when it is terminated.

Only accumulated charges in monetary format are transmitted.

AOC-E information can be conveyed in the following SIP messages:

- BYE
- Final response (BYE)

Annex D Service code commands (SCC) and Service Order Commands (SOC)

Annex D gives an overview of the supplementary services provided by the NGN platform of Deutsche Telekom and the corresponding Service Code Commands (SCC) as well as the Service Order Commands (SOC). These SCC/SOC allow the control of service features by the subscriber via the UE.

Which services actually are provided by the NGN platform of Deutsche Telekom must be taken from the AGB [1] of Deutsche Telekom.

NOTE: Further on, the instruction manual of a particular service has to taken into account.

A Service Code Commands can be made up by "*" (star), "#" (hash) and/or figures; a Service Order Commands can be made up by a Hook Flash function and a figure (usually one digit).

Depending on the service, a SCC can be used to activate, deactivate, interrogate or invoke a specific service feature. For some of these services a PIN is needed for service modifications.

NOTE: The "Hook Flash" function initiated by the user on an UE has to be mapped to SIP messages in the relevant SIP client, accordingly.

Syntax:

The following syntax rules are used for the SCC/SOC:

< > Information in pointed brackets is an obliged input (signs or figures)

NOTE: See also clause 3.3 Symbols.

Annex D.0 Overview of the supplementary services

The following table gives an overview of the currently supported supplementary services in the NGN platform of Deutsche Telekom.

Clause	Service / Supplementary service	Abbr.
D.1	Calling Line Identification Presentation / Originating Identification Presentation [Anzeige der Rufnummer des Anrufers]	CLIP/OIP
D.2	Calling Line Identification Restriction / Originating Identification Restriction [Unterdrückung der Rufnummernanzeige des Anrufers]	CLIR/OIR
D.3	Connected Line Presentation / Terminating Identification Presentation [Anzeige der Rufnummer des Angerufenen]	COLP/TIP
D.4	Connected Line Restriction/Terminating Identification Restriction [Unterdrückung der Rufnummernanzeige des Angerufenen]	COLR/TIR
D.5	Call Waiting / Communication Waiting [Anklopfen]	CW
D.6	Hold / Toggle [Rückfrage/Makeln]	HOLD/TOGGLE
D.7	Three Party Conference / Conference [Dreierkonferenz]	3PTY/CONF
D.8	Call Forwarding Unconditional / Communication Forwarding Unconditional [Anrufweiterschaltung sofort]	CDIV:CFU
D.9	Call Forwarding Busy / Communication Forwarding Busy [Anrufweiterschaltung bei Besetzt]	CDIV:CFB
D.10	Call Forwarding No Reply / Communication Forwarding No Reply [Anrufweiterschaltung bei Nichtmelden]	CDIV:CFNR
D.11	Selective Call Forwarding Unconditional / Selective Communication Forwarding Unconditional [Anrufweiterschaltung selektiv sofort]	CDIV:S-CFU
D.12	Selective Call Forwarding Busy/Selective Communication Forwarding Busy [Anrufweiterschaltung selektiv bei Besetzt]	CDIV:S-CFB
D.13	Selective Call Forwarding No Reply / Selective Communication Forwarding No Reply [Anrufweiterschaltung selektiv bei Nichtmelden]	CDIV:S-CFNR
D.14	Selective Call Forwarding / Selective Communication Forwarding – Deletion/Reset [Anrufweiterschaltung selektiv - aufheben/löschen]	-
D.15	Communication Forwarding on Not Logged-in [Anrufweiterschaltung bei offline/nicht registriert]	CFNL
D.16	Communication Barring - Incoming Communication Barring [Sperren kommend – Alle eingehenden Anrufe abweisen]	ICB

D.17	Communication Barring - Forwarded Communication Barring [Sperren kommend – Alle weitergeleiteter Verbindungen abweisen]	-
D.18	Communication Barring - Anonymous Communication Rejection [Sperren kommend – Alle unbekannten Anrufer abweisen]	ACR
D.19	Communication Barring – Incoming Communication Barring: White list [Sperren kommend – Nur Anrufer aus Positivliste zulassen]	-
D.20	Communication Barring – Incoming Communication Barring: Black list [Sperren kommend – Alle Anrufer aus Negativliste abweisen]	-
D.21	Communication Barring –Incoming Communication Barring: Virtual Black list / Kick Out [Sperren kommend – virtuelle Negativliste (Blacklist – Kick Out)]	-
D.22	Communication Barring - Outgoing Communication Barring [Sperren gehend – alle und selektiv (Wahlsperren)]	OCB
D.23	Communication Barring –Outgoing Communication Barring: White list [Sperren gehend – Nur Anrufe auf Positivliste zulassen (Whitelist)]	-
D.24	Communication Barring – Outgoing Communication Barring: Black list [Sperren gehend – Alle Anrufe auf Negativliste abweisen (Blacklist)]	-
D.25	Completion of Call to Busy Subscriber /Completion of Communications to Busy Subscriber [Rückruf bei Besetzt]	CCBS
D.26	Completion of Call on No Reply/Completion of Communications by No Reply [Rückruf bei Nichtmelden]	CCNR
D.27	Completion of Call on Not Logged-in [Rückruf bei Nichtregistriert]	CCNL
D.28	Explicit Call Transfer/Explicit Communication Transfer [Zusammenschalten von Verbindungen]	ECT
D.29	PIN modification [PIN Modifizierung]	-
D.30	Reset [Zurücksetzen von LM auf Defaultwerte]	-
D.31	Malicious Call Identification/Malicious Communication Identification [Identifizieren/Fangen]	MCID

Table Annex D0-1: Overview of the supplementary services

NOTE 1: For some services no abbreviations are still defined.

NOTE 2: Some official abbreviations are modified for certain services used within the NGN platform of Deutsche Telekom.

Annex D.1 Calling Line Identification Presentation /Originating Identification Presentation (CLIP/OIP)

D.1.1 Procedures

D.1.1.1 Activation

(wait for dial tone) *30*<PIN># (wait for ack.)

D.1.1.2 Deactivation

(wait for dial tone) #30*<PIN># (wait for ack.)

D.1.1.3 Interrogation

(wait for dial tone) *#30# (wait for ack.)

Annex D.2 Calling Line Identification Restriction /Originating Identification Restriction (CLIR/OIR)

D.2.0 Description

CLIR 1: While calling line identification restriction is set to permanent mode, the subscriber can deactivate the calling line identification restriction on per communication basis. If this service is deactivated for an outgoing communication, the calling line identification is presented to the called party for this communication; after this communication the permanent CLIR is automatically activated, again.

CLIR 2: While calling line identification presentation is set to permanent mode, the subscriber can activate the calling line identification restriction on per communication basis. If this service is activated for an outgoing communication, the calling line identification is restricted for this communication; after this communication the CLIR is automatically deactivated, again.

CLIR 3: Permanent calling line identification restriction: This service do NOT allow the subscriber to activate or deactivate the CLIR service permanently.

CLIR: Permanent calling line identification restriction: This service allows the subscriber to activate or deactivate the CLIR service permanently. The selected CLIR mode may be temporarily changed by using CLIR1 or CLIR2.

NOTE: Control of all CLIR supplementary services is only possible if the specific feature CLIR 3 (permanent CLIR for all lines) is not activated/booked.

D.2.1 Procedures

D.2.1.1 Activation

D.2.1.1.1 CLIR1/OIR1

(wait for dial tone) #31#<DN>

D.2.1.1.2 CLIR2/OIR2

(wait for dial tone) *31#<DN>

D.2.1.1.3 CLIR/OIR

(wait for dial tone) *32*<PIN># (wait for ack.)

D.2.1.2. Deactivation

D.2.1.2.1 CLIR1/OIR1

(automatically)

D.2.1.2.2 CLIR2/OIR2

(automatically)

D.2.1.2.3 CLIR/OIR

(wait for dial tone) #32*<PIN># (wait for ack.)

D.2.1.3 Interrogation

D.2.1.3.1 CLIR1/OIR1

Not applicable.

D.2.1.3.2 CLIR2/OIR2

Not applicable.

D.2.1.3.3 CLIR/OIR

(wait for dial tone) *#32# (wait for ack.)

Annex D.3 Connected Line Identification Presentation / Terminating Indication Presentation (COLP/TIP)

D.3.1 Procedures

D.3.1.1 Activation

(wait for dial tone) *40*<PIN># (wait for ack.)

(wait for dial tone) *40*<PIN>*0# (wait for ack.) (for all VoIP lines)

D.3.1.2 Deactivation

(wait for dial tone) #40*<PIN># (wait for ack.)

D.3.1.3 Interrogation

(wait for dial tone) *#40# (wait for ack.)

Annex D.4 Connected Line Identification Restriction / Terminating Indication Restriction (COLR/TIR)

D.4.1 Procedures

D.4.1.1 Activation

(wait for dial tone) *42*<PIN># (wait for ack.)

(wait for dial tone) *42*<PIN>*0# (wait for ack.) (for all VoIP lines)

D.4.1.2. Deactivation

(wait for dial tone) #42*<PIN># (wait for ack.)

D.4.1.3 Interrogation

(wait for dial tone) *#42# (wait for ack.)

Annex D.5 Call Waiting/Communication Waiting (CW)

D.5.1 Procedures

D.5.1.0 General

All following described procedures for Activation, Deactivation, Interrogation and Invocation must be handled internally by the VGW/UE to process the Service. It shall not be sent towards the network. For a VGW it is recommended to implement those functions as follows.

D.5.1.1 Activation

(wait for dial tone) *43# (wait for ack.)

(wait for dial tone) *43*0# (wait for ack.) (for all VoIP lines)

D.5.1.2. Deactivation

(wait for dial tone) #43# (wait for ack.)

(wait for dial tone) #43*0# (wait for ack.) (for all VoIP lines)

D.5.1.3 Interrogation

(wait for dial tone) *#43# (wait for ack.)

D.5.1.4 Invocation

D.5.1.4.1 Acceptance of an incoming communication (with or without authorisation of 3PTY service)

(hang-up and wait for ringing signal)

D.5.1.4.2 Acceptance of an incoming communication (with authorisation of 3PTY service)

a) (wait for special dial tone) **1** (the current communication will be released)

b) (wait for special dial tone) **2** (the current communication is put on HOLD)

D.5.1.4.3 Rejection of an incoming communication

(wait for special dial tone) **0** (the incoming communication will be rejected)

Annex D.6 HOLD / TOGGLE

D.6.1 Procedures

D.6.1.0 General

All following described procedures for Activation, Deactivation, Interrogation and Invocation must be handled internally by the VGW/UE to process the Service. It shall not be sent towards the network. For a VGW it is recommended to implement those functions as follows.

D.6.1.1 Invocation (…)

(wait for special dial tone and dial third party number) <DN> (0…9)

D.6.1.1.1 Worst case (communication could not be established)

a) (wait for special dial tone) **1**

b) (hang-up and wait for ringing signal)

D.6.1.2 Invocation (change to the party on HOLD –TOGGLE-)

(wait for special dial tone) **2** (the current communication is put on HOLD)

D.6.1.3 Invocation (release a communication during HOLD)

D.6.1.3.1 Invocation (release the current communication)

(wait for special dial tone) **1** (the communication on HOLD becomes active)

D.6.1.3.2 Invocation (release the communication on HOLD)

(wait for special dial tone) **0** (the communication on HOLD will be released)

D.6.1.3 Invocation (Release initiated by the current party)

Congestion tone provided

(wait for special dial tone) **1** or **2**

or

(wait for ringing tone)

Annex D.7 Three Party Conference/Conference (3PTY/CONF)

D.7.1 Procedures

D.7.1.0 General

All following described procedures for Activation, Deactivation, Interrogation and Invocation must be handled internally by the VGW/UE to process the Service. It shall not be sent towards the network. For a VGW it is recommended to implement those functions as follows.

D.7.1.1 Invocation (3PTY/CONF initiation)

Prerequisite: Initiator has one communication in an active state and a second communication on hold.

(wait for special dial tone) **3**

D.7.1.2 Invocation (change from 3PTY/CONF to HOLD/TOGGLE)

(wait for special dial tone) **2**

Annex D.8 Communication Diversion: Call Forwarding Unconditional / Communication Forwarding Unconditional (CDIV:CFU)

D.8.1 Procedures

D.8.1.1 Activation [1]

D.8.1.1.1 Activation without Call Forwarded Number (CFN)

(wait for dial tone) *21# (wait for ack.) [2]

D.8.1.1.2 Activation with CFN

(wait for dial tone) *21*<CFN># (wait for ack.)

---

[1] While CFU is activated the special dial tone shall be played after picking up the receiver instead of normal dial tone to remember the user that no incoming calls will be received.

[2] CFN has already been registered before.

D.8.1.2 Deactivation

(wait for dial tone) #21# (wait for ack.)

D.8.1.3 Delete / Reset

(wait for dial tone) ##21# (wait for ack.) [1]

D.8.1.4 Interrogation

(wait for dial tone) *#21# (wait for ack.)

Annex D.9 Communication Diversion: Call Forwarding Busy / Communication Forwarding Busy (CDIV:CFB)

D.9.1 Procedures

D.9.1.1 Activation

D.9.1.1.1 Activation without Call Forwarded Number (CFN)

(wait for dial tone) *67# (wait for ack.) [2]

D.9.1.1.2 Activation with CFN

(wait for dial tone) *67*<CFN># (wait for ack.)

D.9.1.2 Deactivation

(wait for dial tone) #67# (wait for ack.)

D.9.1.3 Delete / Reset

(wait for dial tone) ##67# (wait for ack.) [2]

D.9.1.4 Interrogation

(wait for dial tone) *#67# (wait for ack.)

---

[1] After Delete/Reset the current CFU is deactivated and the default CFU is set automatically according to the initial configuration of this VoIP line.

[2] After Delete/Reset the current CFB/CFNR is deactivated and the default CFB/CFNR is set automatically according to the initial configuration of this VoIP line; In case of CFNR the timer value is set to 20 seconds (default).

Annex D.10 Communication Diversion: Call Forwarding No Reply/ Communication Forwarding No Reply (CDIV:CFNR)

D.10.1 Procedures

D.10.1.1 Activation

D.10.1.1.1 Activation without Call Forwarded Number (CFN) and without timer value

(wait for dial tone) *61# (wait for ack.) [1] [2]

D.10.1.1.2 Activation with CFN; and without timer value

(wait for dial tone) *61*<CFN># (wait for ack.)

D.10.1.1.3 Activation with CFN and with timer value [3]

(wait for dial tone) *61*<CFN>*<timer value># (wait for ack.)

D.10.1.1.4 Activation and modification of timer value (without CFN) [2]

(wait for dial tone) *61**<timer value># (wait for ack.) [3]

D.10.1.2 Deactivation

(wait for dial tone) #61# (wait for ack.)

D.10.1.3 Delete / Reset

(wait for dial tone) ##61# (wait for ack.) [2]

D.10.1.4 Interrogation

(wait for dial tone) *#61# (wait for ack.)

---

[1] CFN has already been registered before.

[2] Default timer value or timer value has already been registered before.

[3] The timer value has a valid range from 5 to 60 (seconds) in steps of 1 (second); default value is 20 seconds; the timer value item may consist of 1 or 2 digits; in case of an invalid entry the current timer value is kept unchanged.

Annex D.11 Communication Diversion: Selective Call Forwarding Unconditional/ Selective Communication Forwarding Unconditional (CDIV:S-CFU)

D.11.1 Procedures

D.11.1.1 Activation [1]

(wait for dial tone) *212*<CFN>*<ON[2]># (wait for ack.)

D.11.1.2 Deactivation

(wait for dial tone) #212# (wait for ack.)

or

(wait for dial tone) #212*<ON># (wait for ack.)

D.11.1.3 Delete / Reset

(wait for dial tone) ##212# (wait for ack.) [3]

D.11.1.4 Interrogation

(wait for dial tone) *#212# (wait for ack.)

---

[1] While S-CFU is activated the special dial tone shall be played after picking up the receiver instead of normal dial tone to remember the user that no incoming calls will be received.

[2] ON= Originating Number

[3] After Delete/Reset the current S-CFU is deactivated and the default S-CFU is set automatically according to the initial configuration of this VoIP line.

Annex D.12 Communication Diversion: Selective Call Forwarding Busy/ Selective Communication Forwarding Busy (CDIV:S-CFB)

D.12.1 Procedures

D.12.1.1 Activation

(wait for dial tone) *214*<CFN>*<ON># (wait for ack.)

D.12.1.2 Deactivation

(wait for dial tone) #214# (wait for ack.)

or

(wait for dial tone) #214*<ON># (wait for ack.)

D.12.1.3 Delete / Reset

(wait for dial tone) ##214# (wait for ack.) [1]

D.12.1.4 Interrogation

(wait for dial tone) *#214# (wait for ack.)

[1] After Delete/Reset the current S-CFB/S-CFNR is deactivated and the default S-CFB/S-CFNR is set automatically according to the initial configuration of this VoIP line; In case of S-CFNR the timer value is set to 20 seconds (default).

Annex D.13 Communication Diversion: Selective Call Forwarding No Reply/ Selective Communication Forwarding No Reply (CDIV:S-CFNR)

D.13.1 Procedures

D.13.1.1 Activation

D.13.1.1.1 Activation with CFN and without timer value

(wait for dial tone) *213*<CFN>*<ON># (wait for ack.) [1]

D.13.1.1.2 Activation with CFN and with timer value [2]

(wait for dial tone) *213*<CFN>*<ON>*<timer value># (wait for ack.)

D.13.1.2 Deactivation

(wait for dial tone) #213# (wait for ack.)

or

(wait for dial tone) #213*<ON># (wait for ack.)

D.13.1.3 Delete / Reset

(wait for dial tone) ##213# (wait for ack.) [1]

D.13.1.4 Interrogation

(wait for dial tone) *#213# (wait for ack.)

---

[1] Default timer value or timer value has already been registered before.

[2] The timer value has a valid range from 5 to 60 (seconds) in steps of 1 (second); default value is 20 seconds; the timer value item may consist of 1 or 2 digits; in case of an invalid entry the current timer value is kept unchanged.

Annex D.14 Selective Call Forwarding/ Selective Communication Forwarding – Deletion/Reset

D.14.1 Procedures

D.14.1.1 Activation

Not applicable.

D.14.1.2 Deactivation

Not applicable.

D.14.1.3 Delete/Reset

(wait for dial tone) #211# (wait for ack.) [1]

or

(wait for dial tone) #211*0# (wait for ack.) (for all VoIP lines) [1]

D.14.1.4 Interrogation

Not applicable.

Annex D.15 Communication Forwarding on Not Logged-in (CFNL)

D.15.1 Procedures

D.15.1.1 Activation

D.15.1.1.1 Activation without Call Forwarding Number (CFN)

(wait for dial tone) *62# (wait for ack.) [2]

---

[1] After Delete/Reset all current S-CF features are deactivated and all default S-CF features are set automatically according to the initial configuration of this VoIP line; The timer value (S-CFNR) is set to 20 seconds (default).

[2] CFN has already been registered before.

D.15.1.1.2 Activation with CFN

(wait for dial tone) *62*<CFN># (wait for ack.)

D.15.1.2 Deactivation

(wait for dial tone) #62# (wait for ack.)

D.15.1.3 Delete / Reset

(wait for dial tone) ##62# (wait for ack.) [1]

D.15.1.4 Interrogation

(wait for dial tone) *#62# (wait for ack.)

Annex D.16 Communication Barring - Incoming Communication Barring (ICB)

D.16.1 Procedures

D.16.1.1 Activation

(wait for dial tone) *335*<PIN># (wait for ack.)

(wait for dial tone) *335*<PIN>*0# (wait for ack.) (for all VoIP lines)

D.16.1.2 Deactivation

(wait for dial tone) #335*<PIN># (wait for ack.)

(wait for dial tone) #335*<PIN>*0# (wait for ack.) (for all VoIP lines)

D.16.1.3 Interrogation

(wait for dial tone) *#335# (wait for ack.)

[1] After Delete/Reset the current CFNL is deactivated and the default CFNL is set automatically according to the initial configuration of this VoIP line.

Annex D.17 Communication Barring - Forwarded Communication Barring

D.17.1 Procedures

D.17.1.1 Activation

(wait for dial tone) *337*<PIN># (wait for ack.)

(wait for dial tone) *337*<PIN>*0# (wait for ack.) (for all VoIP lines)

D.17.1.2 Deactivation

(wait for dial tone) #337*<PIN># (wait for ack.)

(wait for dial tone) #337*<PIN>*0# (wait for ack.) (for all VoIP lines)

D.17.1.3 Interrogation

(wait for dial tone) *#337# (wait for ack.)

Annex D.18 Communication Barring - Anonymous Communication Rejection (ACR)

D.18.1 Procedures

D.18.1.1 Activation

(wait for dial tone) *336*<PIN># (wait for ack.)

(wait for dial tone) *336*<PIN>*0# (wait for ack.) (for all VoIP lines)

D.18.1.2 Deactivation

(wait for dial tone) #336*<PIN># (wait for ack.)

(wait for dial tone) #336*<PIN>*0# (wait for ack.) (for all VoIP lines)

D.18.1.3 Interrogation

(wait for dial tone) *#336# (wait for ack.)

Annex D.19 Communication Barring – Incoming Communication Barring: White list

D.19.1 Procedures

D.19.1.1 Activation “White List” without Calling Number (CN)

(wait for dial tone) *339*<PIN># (wait for ack.)

(wait for dial tone) *339*<PIN>**0# (wait for ack.) (for all VoIP lines)

D.19.1.2 Activation “White List” with Calling Number (CN)

(wait for dial tone) *339*<PIN>*<CN># (wait for ack.)

(wait for dial tone) *339*<PIN>*<CN>*0# (wait for ack.) (for all VoIP lines)

D.19.1.3 Deactivation

(wait for dial tone) #339*<PIN># (wait for ack.)

(wait for dial tone) #339*<PIN>*0# (wait for ack.) (for all VoIP lines)

D.19.1.4 Delete White list

(wait for dial tone) ##339*<PIN># (wait for ack.)

(wait for dial tone) ##339*<PIN>*0# (wait for ack.) (for all VoIP lines)

D.19.1.5 Interrogation

(wait for dial tone) *#339# (wait for ack.)

Annex D.20 Communication Barring – Incoming Communication Barring: Black list

D.20.1 Procedures

D.20.1.1 Activation “Black List” without Calling Number (CN)

(wait for dial tone) *338*<PIN># (wait for ack.)

(wait for dial tone) *338*<PIN>**0# (wait for ack.) (for all VoIP lines)

D.20.1.2 Activation “Black List” with Calling Number (CN)

(wait for dial tone) *338*<PIN>*<CN># (wait for ack.)

(wait for dial tone) *338*<PIN>*<CN>*0# (wait for ack.) (for all VoIP lines)

D.20.1.3 Deactivation

(wait for dial tone) #338*<PIN># (wait for ack.)

(wait for dial tone) #338*<PIN>*0# (wait for ack.) (for all VoIP lines)

D.20.1.4 Delete Black list

(wait for dial tone) ##338*<PIN># (wait for ack.)

(wait for dial tone) ##338*<PIN>*0# (wait for ack.) (for all VoIP lines)

D.20.1.5 Interrogation

(wait for dial tone) *#338# (wait for ack.)

Annex D.21 Communication Barring –Incoming Communication Barring: Virtual Black list / Kick Out

D.21.1 Procedures

D.21.1.1 Activation

(wait for dial tone) *934*<PIN># (wait for ack.)

D.21.1.2 Activation/Deactivation (toggle mode)

(wait for dial tone) #934*<PIN># (wait for ack.)

D.21.1.3 Delete virtual Black list

(wait for dial tone) ##934*<PIN># (wait for ack.)

D.21.1.4 Interrogation

(wait for dial tone) *#934# (wait for ack.)

Annex D.22 Communication Barring - Outgoing Communication Barring (OCB)

D.22.1 Procedures

D.22.1.1 Activation

D.22.1.1.1 Outgoing Communications - All

(wait for dial tone) *03*<PIN># (wait for ack.)

D.22.1.1.2 Outgoing Communications - Selective

(wait for dial tone) *05*<PIN># (wait for ack.) (for all activated classes)

D.22.1.1.2.1 Selective Outgoing Communications to 0900 …

(wait for dial tone) *051*<PIN># (wait for ack.)

D.22.1.1.2.2 Selective Outgoing Communications to 0137 …

(wait for dial tone) *052*<PIN># (wait for ack.)

D.22.1.1.2.3 Selective Outgoing Communications to 0180 …

(wait for dial tone) *053*<PIN># (wait for ack.)

D.22.1.1.2.4 Selective Outgoing Communications to International numbers (00)

(wait for dial tone) *054*<PIN># (wait for ack.)

D.22.1.1.2.5 Selective Outgoing Communications to Intercontinental numbers (0011-0019, 002, 005-009)

(wait for dial tone) *055*<PIN># (wait for ack.)

D.22.1.2. Deactivation

D.22.1.2.1 Outgoing Communications - All

(wait for dial tone) #03*<PIN># (wait for ack.)

D.22.1.2.2 Outgoing Communications - Selective

(wait for dial tone) #05*<PIN># (wait for ack.) (for all activated classes)

D.22.1.2.2.1 Selective Outgoing Communications to 0900 …

(wait for dial tone) #051*<PIN># (wait for ack.)

D.22.1.2.2.2 Selective Outgoing Communications to 0137 …

(wait for dial tone) #052*<PIN># (wait for ack.)

D.22.1.2.2.3 Selective Outgoing Communications to 0180 …

(wait for dial tone) #053*<PIN># (wait for ack.)

D.22.1.2.2.4 Selective Outgoing Communications to International numbers (00)

(wait for dial tone) #054*<PIN># (wait for ack.)

D.22.1.2.2.5 Selective Outgoing Communications to Intercontinental numbers (0011-0019, 002, 005-009)

(wait for dial tone) #055*<PIN> # (wait for ack.)

D.22.1.3 Interrogation

D.22.1.3.1 All Outgoing Communications

(wait for dial tone) *#03# (wait for ack.)

D.22.1.3.2 Selective Outgoing Communications

(wait for dial tone) *#05# (wait for ack.)

Annex D.23 Communication Barring –Outgoing Communication Barring: White list

D.23.1 Procedures

D.23.1.1 Activation “White List” without Originating Number (ON)

(wait for dial tone) *04*<PIN># (wait for ack.)

D.23.1.2 Activation “White List” with Originating Number (ON)

(wait for dial tone) *04*<PIN>*<ON># (wait for ack.)

(wait for dial tone) *04*<PIN>*<ON>*0# (wait for ack.) (for all VoIP lines)

D.23.1.3 Deactivation

(wait for dial tone) #04*<PIN># (wait for ack.)

D.23.1.4 Delete White list

(wait for dial tone) ##04*<PIN># (wait for ack.)

(wait for dial tone) ##04*<PIN>*0# (wait for ack.) (for all VoIP lines)

D.23.1.5 Interrogation

(wait for dial tone) *#04# (wait for ack.)

Annex D.24 Communication Barring – Outgoing Communication Barring: Black list

D.24.1 Procedures

D.24.1.1 Activation “Black List” without Originating Number (ON)

(wait for dial tone) *059*<PIN># (wait for ack.)

D.24.1.2 Activation “Black List” with Originating Number (ON)

(wait for dial tone) *059*<PIN>*<ON># (wait for ack.)

(wait for dial tone) *059*<PIN>*<ON>*0# (wait for ack.) (for all VoIP lines)

D.24.1.3 Deactivation

(wait for dial tone) #059*<PIN># (wait for ack.)

D.24.1.4 Delete Blacklist

(wait for dial tone) ##059*<PIN># (wait for ack.)

(wait for dial tone) ##059*<PIN>*0# (wait for ack.) (for all VoIP lines)

D.24.1.5 Interrogation

(wait for dial tone) *#059# (wait for ack.)

Annex D.25 Completion of Communication to Busy Subscriber (CCBS)

D.25.1 Procedures

D.25.1.1 Activation

(subscriber busy; announcement CCBS possible) *37# (wait for ack.)

D.25.1.2. Deactivation of all activated recalls

(wait for dial tone) #37# (wait for ack.)

D.25.1.3 Interrogation

(wait for dial tone) *#37# (wait for ack.)

Annex D.26 Completion of Communication on No Reply (CCNR)

D.26.1 Procedures

D.26.1.1 Activation

(announcement CCNR possible) *38# (wait for ack.)

D.26.1.2. Deactivation of all activated recalls

(wait for dial tone) #38# (wait for ack.)

D.26.1.3 Interrogation

(wait for dial tone) *#38# (wait for ack.)

Annex D.27 Completion of Communication on Not Logged-in (CCNL)

D.27.1 Procedures

D.27.1.1 Activation

(no answer; announcement CCNL possible) *36# (wait for ack.)

D.27.1.2. Deactivation

(wait for dial tone) #36# (wait for ack.)

D.27.1.3 Interrogation

(wait for dial tone) *#36# (wait for ack.)

Annex D.28 Explicit Call Transfer (ECT)

D.28.1 Procedures

D.28.1.1 Activation

Activation of ECT is only possible during the communication state.

D.28.1.1.1 Controlled by subscriber B (after an incoming call)

3. Active communication between subscriber A and subscriber B,
   - Subscriber B initiates Hook Flash ( ),
   - Subscriber A is on HOLD (see Annex D.6 HOLD / TOGGLE),
   - Subscriber B calls subscriber C,
   - Active communication between subscriber B and subscriber C,
   - Subscriber A is still on HOLD,
4. Subscriber B initiates ECT with

(wait for special dial tone) **4** (wait for ack.)

5. Subscriber A is connected to subscriber C,
   - Subscriber B has been released,

6. New active communication between subscriber A and subscriber C;

D.28.1.1.2 Controlled by subscriber A (after an outgoing call)

7. Active communication between subscriber A and subscriber B,
   - Subscriber A initiates Hook Flash ( ),
   - Subscriber B is on HOLD (see Annex D.6 HOLD / TOGGLE),
   - Subscriber A calls subscriber C,
   - Active communication between subscriber A and subscriber C,
   - Subscriber B is still on HOLD,
8. Subscriber A initiates ECT with

(wait for special dial tone) **4** (wait for ack.)

9. Subscriber B is connected to subscriber C,
   - Subscriber A has been released,
10. New active communication between subscriber B and subscriber C.

D.28.1.2. Deactivation

Disconnection of all active ECT communications initiated on this subscriber number (initiator).

(wait for dial tone) #96*<PIN># (wait for ack.)

D.28.1.3 Interrogation

Control procedure for the number and the status of active ECT communications.

(wait for dial tone) *#96# (wait for ack.)

Annex D.29 PIN Modification

D.29.1 Procedures

D.29.1.1 PIN modification

(wait for dial tone) *99*<PINold>*<PINnew>*<PINnew># (wait for ack.)

Annex D.30 Reset All Services to Default Values

D.30.1 Procedures

D.30.1.1 Activation

D.30.1.1.1 Activation without deletion of list entries

(wait for dial tone) *001*<PIN># (wait for ack.)

or

(wait for dial tone) *001*<PIN>*0# (wait for ack.) (for all VoIP lines)

D.30.1.1.2 Activation with deletion of list entries

(wait for dial tone) *000*<PIN># (wait for ack.)

or

(wait for dial tone) *000*<PIN>*0# (wait for ack.) (for all VoIP lines)

D.30.1.2. Deactivation

Not applicable.

D.30.1.3 Interrogation

Not applicable.

Annex D.31 Malicious Communication Identification

D.31.1 Procedures

D.31.1.1 Activation

(wait for dial tone) *39# (wait for ack.) (flag the last call)

(wait for dial tone) *392# (wait for ack.) (flag the penultimate call)

D.31.1.2. Deactivation

Not applicable.

D.31.1.3 Interrogation

Not applicable.

Annex E (informative) Supervisory tones

Annex E.0 General

This annex E provides user equipment information (e.g. for SIP phones) with the purpose to optimise the user interface by generating the well known supervisory tones in the UE.

The following tables provide the respective audible tones according to the relevant cause:

Cause	Audible tone
Off-hook; as prescribed in user profile: name= „dial-tone-pattern“ (default = „standard-dial-tone“).	Dial tone
Like „Dial tone“; as prescribed in user profile: name= „dial-tone-pattern“, value = „special-condition-tone“.	Special dial tone
SIP INVITE request with CW indication during communication state.	Communication waiting tone
Receipt of a SIP response “180 Ringing” without P-Early-Media header.	Ringing tone
Receipt of a SIP response “486 Busy Here”	Busy tone
After receipt of any 4xx, 5xx or 6xx responses as well as on receipt of a BYE or CANCEL request in off-hook or communication state.	Congestion tone

Responses with Reason-Header, cause = [1]		Audible tone
16	„Normal call clearing“	Congestion tone
17	„User busy“	Busy tone
18	„No user responding“	Busy tone
19	„No answer from user (user alerted)”	Busy tone
28	„Address incomplete“	Congestion tone
31	„Normal, unspecified“	Congestion tone
34	„No circuit/ channel available“	Congestion tone
38	„Network out of order“	Congestion tone
41	„Temporary failure“	Congestion tone
44	„Requested circuit/channel not available”	Congestion tone
47	„Resources unavailable, unspecified”	Congestion tone
58	„Bearer capability not implemented“	Congestion tone
91	„Invalid transit selection“	Congestion tone
102	„Recovery on timer expiry“	Congestion tone

[1] Causes according to Q.850. On deviant cause values the “Congestion tone” shall be played.
If the Reason header is missing, the audible tone regarding the SIP response/request (upper part of Table Annex E 1) shall be played.

Responses with Reason-Header, cause = 1	Audible tone
127 „Interworking unspecified“	Congestion tone

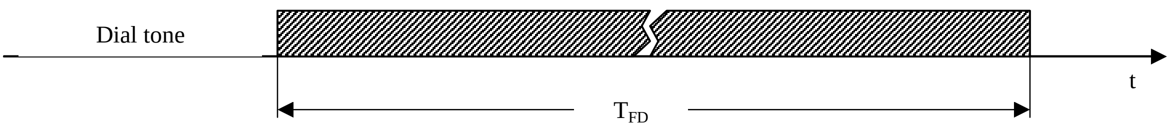
Table Annex E 1: Audible tones

The levels for audible tones shall be properly defined. The following signals should be provided by an UE.

Annex E.1 Dial tone

The dial tone should be applied in the off-hook state (receiver picked up or loudspeaker activated) until the first digit has been dialled. It serves as information "ready for dialling" for the subscriber or specific terminal devices.

The dial tone should be a permanent tone with a frequency range of 425 Hz ± 7 Hz and a distortion factor of k ≤ 5 % as depicted in the figure below.

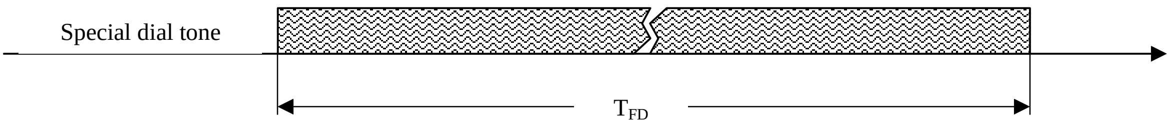


Annex E.2 Special dial tone

The special dial tone should be applied in the off-hook state (receiver picked up or loudspeaker activated) in conjunction with a particular case (e.g. CFU is activated) until the first digit has been dialled. It serves as information "ready for dialling" for the subscriber or specific terminal devices.

In addition, the special dial tone gives the subscriber a hint about a specific situation (e.g. another party is on hold).

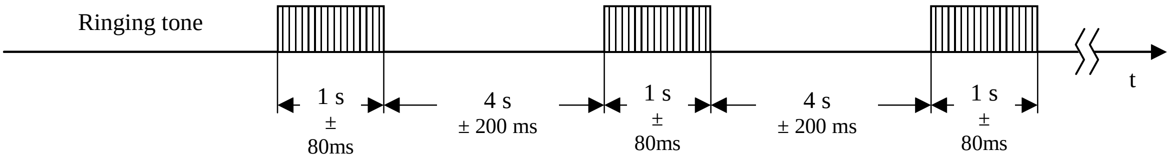
The special dial tone should be a permanent tone with a frequency composition range of f_1 = 400 Hz ± 7 Hz and f_2 = 425 Hz ± 7 Hz (additive); the distortion factor shall be k ≤ 5 % as depicted in the figure below.



Annex E.3 Ringing tone

The ringing tone is provided after a successful communication set-up is finalized and the destination party is idle. The ringing tone serves as information "ringing signal is provided to the destination party" for the subscriber or specific terminal devices.

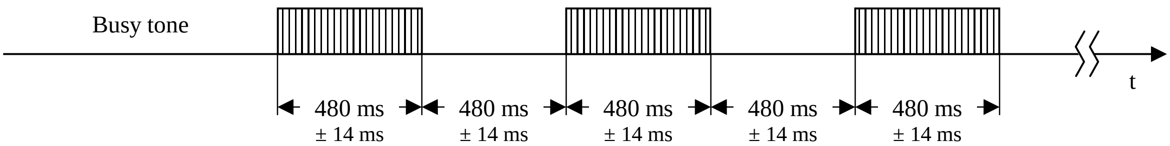
The ringing tone should be a paced tone with a frequency range of 425 Hz ± 7 Hz and a distortion factor of k ≤ 5 % as depicted in the figure below.



Annex E.4 Busy tone

The busy tone is provided after a successful communication set-up is finalized and the destination party is busy. The busy tone serves as information "destination party busy" for the subscriber or specific terminal devices.

The busy tone should be a paced tone with a frequency range of 425 Hz ± 7 Hz and a distortion factor of k ≤ 5 % as depicted in the figure below.



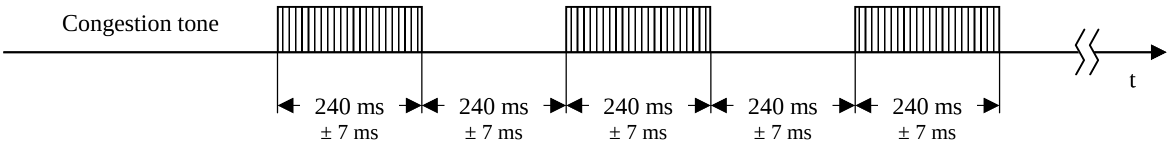
Annex E.5 Congestion tone

The congestion tone is provided in case of an unsuccessful communication set-up or in case of an error. The congestion tone serves as information "unsuccessful communication" or "operating error" (see bullet points below) for the subscriber or specific terminal devices.

The congestion tone should be a paced tone with a frequency range of 425 Hz ± 7 Hz and a distortion factor of k ≤ 5 % as depicted in the figure below.

The congestion tone is played in case the following causes:

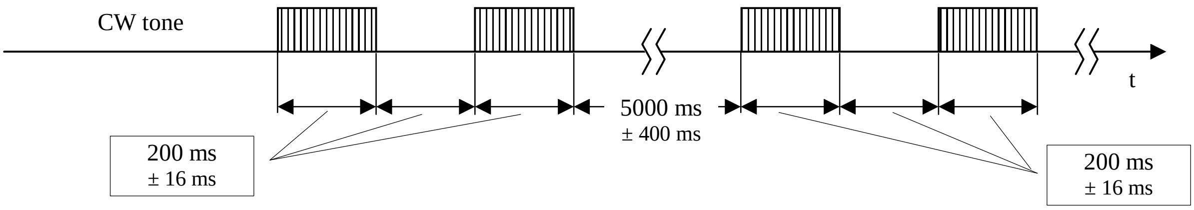
- Communication set-up to the destination party is not possible,
- a time-out has expired,
- a garbage has been detected (e.g. SOC/SCC).



Annex E.6 Communication waiting tone

The communication waiting tone (CW tone) is provided during the communication state while a third party communication is waiting. The CW tone serves as information “another communication is waiting” for the subscriber or specific terminal devices.

The communication waiting tone should be a paced tone with a frequency range of 425 Hz ± 7 Hz and a distortion factor of k ≤ 5 % as depicted in the figure below.



Annex E.7 Acoustic ringing signal

The acoustic ringing signal for incoming calls (external call) is provided during the idle state and should have the following ringing cadence:

Table E.7-1

	Duration in ms				
Ringing signal (pulse)	920	≤	1000	≤	1080

Ringing signal pause	4800	≤	5000	≤	5200

Additional deviated ringing cadences for other kind of calls or features are possible.

History

Version	Published	Remarks
1.0	15.05.2009	First edition of 1 TR 114
1.1	05.11.2009	Updated edition of 1 TR 114 Version 1.0: - General formal changes (editorial); - Foreword updated; - References updated (RFC 2411, RFC 4734 and RFC 5244 added, RFC 2833 deleted); - 3.1 Abbreviations updated; - 3.3 Symbols updated; - Clause 4.2.1 (A-law and µ-law added, DTMF moved to 4.2.4); - Clause 4.2.4 added (DTMF); - Clause 6: Note added; - Clause 7.1.1 updated (also: “TLS is not used” deleted, global number format, challenge mechanism, DNS SRV); - Clause 7.2 updated: §5.1.1.2A added and modified, §5.1.2A.1: P-Preferred-ID, §9.2.1: DNS procedures added, §9.2.1 added and modified, Table A.4 modified (Item: 89, DT1; Condition: c17, c90, c_dt4); - Clause 7.3 updated: Table 7-1 modified; - Clause 7.3.2 updated: Table 7-2 (item numbering and conditions modified; Overlap deleted); - Clause 7.3.3 updated: Table 7-3 (item: 101A added) - Clause 7.3.4 updated: Table 7-4 (item 33b and 36a added, 39, 61b and 70a modified, conditions c6, c7, c9 and Note 4 modified); - Clause 7.3.5 updated: Table 7-5 (condition c1 modified); - Clause 7.3.6 “SDP Types” (new): §A3.2.2 added; - Clause 7.4.1 updated: Table 7-6 (Ref. modified and column “Support” added, IPSec added); - Clause 7.4.2: Table 7-7, Table 7-8 and Table 7-9 updated; - Clause 7.4.3: Table 7-10 updated; - Clause 7.4.4 (new) “Real time Transport Procedures” added: Table 7-11 added; - Clause 8.4 (new) “Network access” added; - Clause 8.5 (new) “Number handling by the UE” added; - Clause 8.6 (new) added: “Support of NAT traversal by the UE”; - A.2.1.1: Footnote 1 added; - A.2.2: Note and footnote added; - A.2.4 (§4.5.2.1): modified (anonymous and display name not allowed); - A.2.6: modified (re-INVITE); - B.2: Updated (CLIR 3 becomes CLIR/OIR) and Note added; - B.8: Updated and footnote added; - B.10: Updated and footnote added; - B.11: Footnote added; - B.13: Updated and footnote added; - B.15: Updated and footnote added;

Version	Published	Remarks
		- B.19: Procedures “336” changed to “339”; - B.20 (B.20.1.2): corrected; - B.21 (B.21.1.2): corrected; - B.23 (B.23.1.2): corrected; - B.24 (B.24.1.2 and B.24.1.4): corrected; - B.25 (B.25.1.2): corrected; - B.26 (B.26.1.1 and B.26.1.2): corrected; - B.27 (B.27.1.1): corrected; - B.28 (ECT) completed (added); - Annex C (Supervisory tones) added.
2.0	30.06.2010	Second edition of 1 TR 114: - General (formal) updates; - Foreword corrected; - Endorsements of ETSI TS124 503 and 3GPP TS 24.229 (modifications) moved to Annex A and Annex B; - Endorsement part (new Annex A and new Annex B) deleted and general text updated; - References updated and V.152 and T.30 added; - Reference [36] added; - Reference to µ-Law deleted in clauses 4.2.2 and 6. (Table 6-1); - Clause 4.2.1 (new): SIP capabilities inserted; - Clause 4.2.3 Fax and Modem: Text modified (T.38 not supported) - Clause 5.2: DDI text deleted; - Clause 7.1.1 : Text (HEX digits) added; - Clause 7.1.1 and C.2.11.1: bidirectional early media clarified; - Clause 7.3.4 (Table 7-4): c5 “(REFER)” added; - Clause 7.4.2: condition c1 deleted and RFC 4443 set to “m”; - Clause 7.4.4: RFC 4040 added; - Clause 7.4.4: NOTE2 added; - Clause 8.4.1: G.711a explicit specified; - Table 7-2: Supported methods (Item 12: “o” changed to “m” and Note 1 added); - Table 7-3 : note 5 added; - Table 7-7: reference to RFC 0792 added; - Table headers corrected; - Priority of Annex A against Annex B clarified; - Annex B: already deleted clause 5.1.1.2.1 reactivated, again; - Clause C.2.10 (§4.5.2.1): bullet point added (Session-ID); - Clause C.2.12: Text (AOC) added; - Clauses C.2.12.1 … C.2.12.3 updated; - Delta list in History added (for V1.1); - T-Home replaced by Deutsche Telekom; - Zentrum Technik Einführung (ZTE) replaced by Fixed Mobile Engineering Deutschland (FMEd); - Document properties updated; - History updated.
2.1	22.07.2010	Updated edition of 1 TR 114 (V2.0): - Formal corrections; - Clause 3.1: Updated;

Version	Published	Remarks
		- Clause 7.4.4: Requirement for RTCP/SDES added; - Clause 7.1.1: Support of GRUU (mandatory) and Instance-ID added; - Annex C.1: First sentence corrected (GRUU is used); - References RFC 4122 and RFC 5626 added; - Annex C.2.6: Text added (bandwidth); - References RFC 3890 added; - References RFC 5627 added; - Clause 7.1.1: Bullet point concerning Call-Id added; - Annex D: Editorial changes; - Annex D.7: Prerequisite for 3PTY added; - Annex D.8.1.3, D.9.1.3 and D.10.1.3: Footnote added; - Annex D.11.1.3, D.12.1.3 and D.13.1.3: Footnote added; - Annex D.14.1.3 and D.14.1.4 swapped; - Annex D.14.1.3, D.15.1.3: Footnote added. - Endorsement References of ETSI TS124 503 and 3GPP TS 24.229 modified to 1TR114 Annex A Version 1.1.0 and 1TR114 Annex B Version 1.1.0;
2.2	18.02.2011	- Formal changes (Version number on cover page); - Reference to ETSI TS 124 505 deleted; - Reference to ETSI TS 124 605 (Release 9) added; - Annex C.2.2: Reference updated; - Table Annex E 1: State P4 deleted and “off-hook state” added; ; - Clause §4.5.2.1: “may” replaced by “shall; - Annex C.2.8 CW: Note added (ghost ringing); ; - Annex C.2.10 ECT: Paragraph §A.2 and note added (call flow); ; - Fetch Binding Requests shall not apply or shall be used minimal.
2.2.1	01.03.2011	- Editorial Change of Date - Note added for DTMF use when not supporting RFC4733
2.3.0	13.04.2011	- Editorial: reinstallation of correct numbering - Marking of features which will be not supported in September 2011 - Reference RFC4884 added. Section 1, Table 7.-7 (needed for IPv6), - IPv6 Procedures added Section 8.2 and 8.3 - Addition of clarification for implicit registration procedure for implicit registration sets in HSS (use of P-Associated-URI header) Section 7.7.1 - Activation option via TR-069 [72], TR-181 [74] AND TR104 [72] for HOLD, CW, TOGGEL and CONF added, Section 5.1 and ANNEX C - Configuration option for local CW Section 5.1 - implicit registration for "ua-profile" included, Section 4.2.1 - Table 7-2 PUBLISH behaviour corrected - Table 7-2 REGISTER behaviour corrected - Table 7-2 UPDATE behaviour corrected
2.4.0	15.12.2011	Main Change is that the procedures for CONF/3PTY, HOLD, CW and TOGGLE where changed from a centric network approach to the implementation within the end device. Foreword, better description to understand the structure of the

Version	Published	Remarks
		document (editorial) Section 4.2.1 editorial write up to understand requirements on preconditions. (editorial) Section 7.1.1 editorial extensions for better understanding, Section7.1.1 deletion of GRUU, Section 7.1.1 Addition of early Challenge procedure Section 7.1.1 Addition of REGISTER procedure in case of the receipt of 403 Page 40, Note5 geändert um weitere Registrierung mit gleichem Contact zuzulassen auch bei 403. Table 7-4 Refer-sub added, Prox-Authentication entry corrected. (editorial) Table 7-3 new Status Code ended 469 (Bad INFO Package) (editorial/default behaviour) 7.4.1 Note in Table 7-6 shifted. (editorial) C.2.2 (CONF) corrected for VGW internal Service execution C.2.6 (HOLD) corrected for VGW internal Service execution C.2.8 (CW) corrected for VGW internal Service execution D.5 (CW) Addition of general procedure for VGW internal Service execution D.6 (HOLD/TOGGLE) Addition of general procedure for VGW internal Service execution D.7 (CONF) Addition of general procedure for VGW internal Service execution Comment included on Option for Implementing *# and SIP Procedures.
3.0.0		-locating P-CSCF and correct prioritization of P-CSCF in case of registration including maintenance procedures. -Preconditions support "passive" better described -Early-Media Header and indication of early media described to avoid misinterpretation. And allow handling of calls initiated by mobile devices. - use of from-change. No default setting - deletion of Annex A - Update of Annex B - Deletion of TS 124.503 - UPDATE to 3GPP Release 11 documents -Correction of *# Procedures using PIN (ECT, OCB, Kick Out, Black List, White List, ACR, CB, ICB) - CLIR 3 included in D.2.0 - Documentation Update TIP/TIR and OIP/OIR -MWI voided - Documentation Update of " 8.6 Support of NAT traversal by the UE" -MIME Type UPDATE Table 7-5 -UPDATE Table 7-4 SIP Headers - add references TR-069, TR-104 and TR-181 - add reference 3GPP TS 23.003 - C.2.8 allow implementations acting on "application/vnd.3gpp.cw+xml" All changes are backward compatible with the procedures described within 1TR114 Version 2.4.0

Version	Published	Remarks