IMS Profile for Voice and SMS

Version 10.0

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1 Introduction

1.1 Overview

The IP Multimedia Subsystem (IMS) Profile for Voice and SMS, documented in this Permanent Reference Document (PRD), defines a profile that identifies a minimum mandatory set of features which are defined in 3GPP specifications that a wireless device (the User Equipment (UE)) and network are required to implement in order to guarantee an interoperable, high quality IMS-based telephony service and IMS-based and SGs-based Short Message Service (SMS) over Long Term Evolution (LTE) radio access. The scope includes the following aspects:

- IMS basic capabilities and supplementary services for telephony [Chapter 2].
- Real-time media negotiation, transport, and codecs [Chapter 3].
- LTE radio and evolved packet core capabilities [Chapter 4].
- Functionality that is relevant across the protocol stack and subsystems [Chapter 5].
- Additional features that need to be implemented for the UEs and networks that wish to support concurrent Circuit Switched (CS) coverage [Annex A].
- Additional features that only a subset of the IMS telephony operators needs to support in certain markets [Annex B].

The UE and network protocol stacks forming the scope of the IMS Profile for Voice and SMS are depicted in figure 1.1 below:

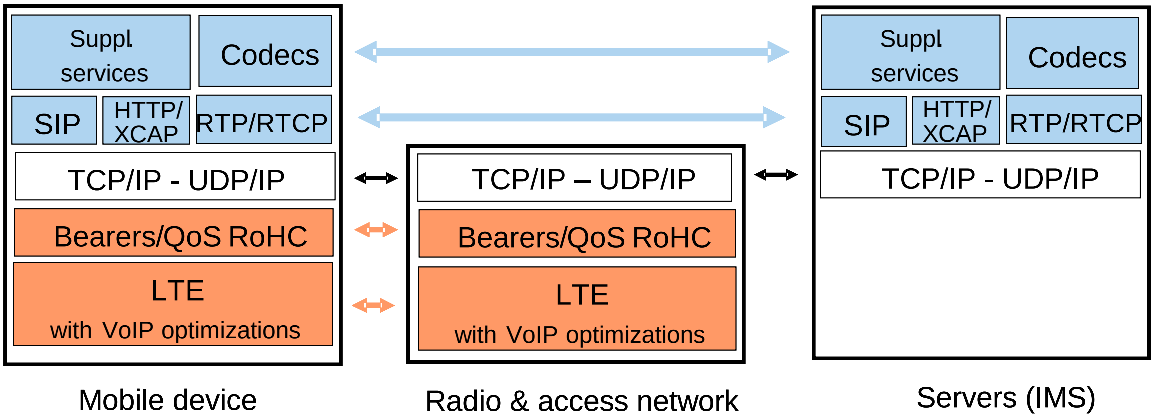

Figure 1.1: Depiction of UE and Network Protocol Stacks in IMS Profile for Voice

The main body of this PRD is applicable for a scenario where IMS telephony is deployed over LTE in a standalone fashion without relying on any legacy infrastructure, packet or circuit switched. In order to be compliant with IMS Profile for Voice and SMS, the UEs and networks must be compliant with all of the normative statements in the main body.

Annex A defines the profile for an alternative approach where IMS telephony is deployed with a certain degree of reliance on an existing 3GPP circuit switched network infrastructure. Whenever there are additional requirements to the main profile, these are explicitly stated. In order to be compliant with the functionality described in Annex A, the UEs and networks must be compliant with all of the normative statements in Annex A as well as to all of the normative statements in the main body of the PRD that are unaltered by Annex A.

1.2 Relationship to existing standards

1.2.1 3GPP specifications

This profile is solely based on the open and published 3GPP specifications as listed in the Section 1.5. 3GPP Release 8, the first release supporting LTE, is taken as a basis. It should be noted, however that not all the features mandatory in 3GPP Release 8 are required for compliance with this profile.

Conversely, some features required for compliance with this profile are based on functionality defined in 3GPP Release 9 or higher releases.

All such exceptions are explicitly mentioned in the following sections along with the relevant Release 8 or higher 3GPP release specifications, respectively.

Unless otherwise stated, the latest version of the referenced specifications for the relevant 3GPP release applies.

1.3 Scope

This document defines a profile for voice over IMS over LTE, and for SMS over IMS and SMS over SGs, by listing a number of Evolved Universal Terrestrial Radio Access Network (E-UTRAN), Evolved Packet Core, IMS core, and UE features that are considered essential to launch interoperable services. The defined profile is compliant with 3GPP specifications. The scope of this profile is the interface between UE and network.

The profile does not limit anybody, by any means, to deploy other standardized features or optional features, in addition to the defined profile.

1.4 Definition of Acronyms and Terms

1.4.1 Acronyms

Acronym	Description
3GPP	3rd Generation Partnership Project
AM	Acknowledged Mode
AMR	Adaptive Multi-Rate
AMR-WB	Adaptive Multi-Rate Wideband
APN	Access Point Name
AVP	Audio Video Profile
AVPF	AVP Feedback Profile
BSF	Bootstrapping Server Function
CB	Communication Barring
CDIV	Communication Diversion
CFNL	Communication Forwarding on Not Logged-in
CFNRc	Communication Forwarding on Not Reachable
CN	Core Network
CS	Circuit Switched

Acronym	Description
CSFB	CS Fallback
CW	Communication Waiting
DRB	Data Radio Bearer
DRX	Discontinuous Reception
DTX	Discontinuous Transmission
ECT	Explicit Communication Transfer
eNB	eNodeB
EPS	Evolved Packet System
E-UTRAN	Evolved Universal Terrestrial Radio Access Network
EVS	Enhanced Voice Services
FDD	Frequency-Division Duplexing
GBR	Guaranteed Bit Rate
GRUU	Globally Routable User agent URI
GSM	Global System for Mobile communications
ICS	IMS Centralized Services
ICSI	IMS Communication Service Identifier
IM	IP Multimedia
IMPU	IP Multimedia Public Identity
IMS	IP Multimedia Subsystem
IMS-AKA	IMS Authentication and Key Agreement
IMSI	International Mobile Subscriber Identity
IP	Internet Protocol
IPv4	Internet Protocol Version 4
IPv6	Internet Protocol Version 6
ISIM	IM Services Identity Module
LTE	Long Term Evolution
MGW	Media Gateway
MMTel	Multimedia Telephony
MO	Managed Object
MRFP	Media Resource Function Processor
MS	Mobile Station
MS-ISDN	Mobile Subscriber ISDN Number
MWI	Message Waiting Indication
NGBR	Non Guaranteed Bit Rate
PCC	Policy and Charging Control
PCRF	Policy and Charging Rules Function
P-CSCF	Proxy - Call Session Control Function
PDN	Packet Data Network

Acronym	Description
PS	Packet Switched
QCI	Quality of Service Class Indicator
RAT	Radio Access Technology
RLC	Radio Link Control
RoHC	Robust Header Compression
RR	Receiver Report
RTCP	RTP Control Protocol
RTP	Real Time Protocol
SCC AS	Service Centralization and Continuity Application Server
SDES	Source Description
SDP	Session Description Protocol
SigComp	Signalling Compression
simservs	MMTel supplementary services XML document
SIP	Session Initiation Protocol
SMSoIP	SMS over IP
SR	Sender Report
SRB	Signalling Radio Bearer
SR-VCC	Single Radio Voice Call Continuity
TAS	Telephony Application Server
TDD	Time-Division Duplexing
TFO	Tandem-Free Operation
TrFO	Transcoder-Free Operation
UDP	User Datagram Protocol
UE	User Equipment
UICC	Universal Integrated Circuit Card
UM	Unacknowledged Mode
URI	Uniform Resource Identifier
USSD	Unstructured Supplementary Service Data
VoIP	Voice Over IP
XCAP	XML Configuration Access Protocol
XML	eXtensible Markup Language
UDUB	User Determined User Busy

1.4.2 Terms

Term	Description
Data Off	A feature, which when activated, prevents transport via PDN connections in 3GPP access of all IP packets except IP packets required by Data Off Enabled Services. Data Off can be activated only when the UE roams or regardless whether the UE roams or not, depending on UE implementation.

Term	Description
Data Off Enabled Service	A service that is enabled (e.g. based on device configuration) regardless of when Data Off is activated.
Region	A part of a country, a country or a set of countries.

1.5 Document Cross-References

Ref	Doc Number	Title
[1]	3GPP TS 22.011	Service Accessibility
[2]	3GPP TS 23.003	Numbering, addressing and identification
[3]	3GPP TS 23.167	IP Multimedia Subsystem (IMS) emergency sessions
[4]	3GPP TS 23.203	Policy and charging control architecture
[5]	3GPP TS 23.216	Single Radio Voice Call Continuity (SRVCC); Stage 2
[6]	3GPP TS 23.221	Architectural requirements
[7]	3GPP TS 23.228	IP Multimedia Subsystem (IMS); Stage 2
[8]	3GPP TS 23.237	IP Multimedia Subsystem (IMS) Service Continuity; Stage 2
[9]	3GPP TS 23.272	Circuit Switched (CS) fallback in Evolved Packet System (EPS); Stage 2
[10]	3GPP TS 23.401	General Packet Radio Service (GPRS) enhancements for Evolved Universal Terrestrial Radio Access Network (E-UTRAN) access
[11]	3GPP TS 24.008	Mobile radio interface layer 3 specification; Core Network protocols; Stage 3
[12]	3GPP TS 24.109	Bootstrapping interface (Ub) and network application function interface (Ua); Protocol details
[13]	3GPP TS 24.147	Conferencing using the IP Multimedia (IM) Core Network (CN) subsystem; Stage 3
[14]	3GPP TS 24.173	IMS Multimedia telephony service and supplementary services; Stage 3
[15]	3GPP TS 24.229	IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3
[16]	3GPP TS 24.237	IP Multimedia Subsystem (IMS) Service Continuity; Stage 3
[17]	3GPP TS 24.301	Non-Access-Stratum (NAS) protocol for Evolved Packet System (EPS); Stage 3
[18]	3GPP TS 24.305	Selective Disabling of 3GPP User Equipment Capabilities (SDoUE) Management Object (MO)
[19]	3GPP TS 24.341	Support of SMS over IP networks; Stage 3
[20]	3GPP TS 24.604	Communication Diversion (CDIV) using IP Multimedia (IM)Core Network (CN) subsystem; Protocol specification
[21]	3GPP TS 24.605	Conference (CONF) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[22]	3GPP TS 24.606	Message Waiting Indication (MWI)using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification

Ref	Doc Number	Title
[23]	3GPP TS 24.607	Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[24]	3GPP TS 24.608	Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[25]	3GPP TS 24.610	Communication HOLD (HOLD) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[26]	3GPP TS 24.611	Anonymous Communication Rejection (ACR) and Communication Barring (CB) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[27]	3GPP TS 24.615	Communication Waiting (CW) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol Specification
[28]	3GPP TS 24.623	Extensible Markup Language (XML) Configuration Access Protocol (XCAP) over the Ut interface for Manipulating Supplementary Services
[29]	3GPP TS 26.071	Mandatory speech CODEC speech processing functions; AMR speech Codec; General description
[30]	3GPP TS 26.073	ANSI C code for the Adaptive Multi Rate (AMR) speech codec
[31]	3GPP TS 26.090	Mandatory Speech Codec speech processing functions; Adaptive Multi-Rate (AMR) speech codec; Transcoding functions
[32]	3GPP TS 26.093	Mandatory speech codec speech processing functions Adaptive Multi-Rate (AMR) speech codec; Source controlled rate operation
[33]	3GPP TS 26.103	Speech codec list for GSM and UMTS
[34]	3GPP TS 26.104	ANSI-C code for the floating-point Adaptive Multi-Rate (AMR) speech codec
[35]	3GPP TS 26.114	IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction
[36]	3GPP TS 26.131	Terminal acoustic characteristics for telephony; Requirements
[37]	3GPP TS 26.132	Speech and video telephony terminal acoustic test specification
[38]	3GPP TS 26.171	Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; General description
[39]	3GPP TS 26.173	ANSI-C code for the Adaptive Multi-Rate - Wideband (AMR-WB) speech codec
[40]	3GPP TS 26.190	Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Transcoding functions
[41]	3GPP TS 26.193	Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Source controlled rate operation
[42]	3GPP TS 26.204	Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; ANSI-C code
[43]	3GPP TS 27.007	AT command set for User Equipment (UE)

Ref	Doc Number	Title
[44]	3GPP TS 31.103	Characteristics of the IP Multimedia Services Identity Module (ISIM) application
[45]	3GPP TS 33.203	3G security; Access security for IP-based services
[46]	3GPP TS 33.222	Generic Authentication Architecture (GAA); Access to network application functions using Hypertext Transfer Protocol over Transport Layer Security (HTTPS)
[47]	3GPP TS 36.101	Evolved Universal Terrestrial Radio Access (E-UTRA); User Equipment (UE) radio transmission and reception
[48]	3GPP TS 36.104	Evolved Universal Terrestrial Radio Access (E-UTRA); Base Station (BS) radio transmission and reception
[49]	3GPP TS 36.300	Evolved Universal Terrestrial Radio Access (E-UTRA) and Evolved Universal Terrestrial Radio Access Network (E-UTRAN); Overall description; Stage 2
[50]	3GPP TS 36.321	Evolved Universal Terrestrial Radio Access (E-UTRA); Medium Access Control (MAC) protocol specification
[51]	3GPP TS 36.323	Evolved Universal Terrestrial Radio Access (E-UTRA); Packet Data Convergence Protocol (PDCP) specification
[52]	3GPP TS 36.331	Evolved Universal Terrestrial Radio Access (E-UTRA);Radio Resource Control (RRC); Protocol specification
[53]	IETF RFC 768	User Datagram Protocol
[54]	IETF RFC 3095	RObust Header Compression (ROHC): Framework and four profiles: RTP, UDP, ESP, and uncompressed
[55]	IETF RFC 3261	SIP: Session Initiation Protocol
[56]	IETF RFC 3550	RTP: A Transport Protocol for Real-Time Applications
[57]	IETF RFC 3551	RTP Profile for Audio and Video Conferences with Minimal Control
[58]	IETF RFC 3556	Session Description Protocol (SDP) Bandwidth Modifiers for RTP Control Protocol (RTCP) Bandwidth
[59]	IETF RFC 3680	A Session Initiation Protocol (SIP) Event Package for Registrations
[60]	IETF RFC 3842	A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
[61]	IETF RFC 4575	A Session Initiation Protocol (SIP) Event Package for Conference State
[62]	IETF RFC 4815	RObust Header Compression (ROHC): Corrections and Clarifications to RFC 3095
[63]	IETF RFC 4867	RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs
[64]	IETF RFC 5939	Session Description Protocol (SDP) Capability Negotiation
[65]	GSMA PRD IR.65	IMS Roaming and Interworking Guidelines
[66]	GSMA PRD IR.67	DNS/ENUM Guidelines for Service Providers and GRX/IPX Providers
[67]	GSMA PRD IR.88	LTE Roaming Guidelines

Ref	Doc Number	Title
[68]	3GPP TS 24.167	3GPP IMS Management Object (MO); Stage 3
[69]	3GPP TS 36.322	Evolved Universal Terrestrial Radio Access (E-UTRA); Radio Link Control (RLC) protocol specification
[70]	ITU-T Recommendation T.140	Protocol for multimedia application text conversation
[71]	3GPP TS 24.628	Common Basic Communication procedures using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[72]	IETF RFC 4961	Symmetric RTP / RTP Control Protocol (RTCP)
[73]	IETF RFC 4745	Common Policy: A Document Format for Expressing Privacy Preferences
[74]	IETF RFC 5009	Private Header (P-Header) Extension to the Session Initiation Protocol
[75]	IETF RFC 4825	The Extensible Markup Language (XML) Configuration Access Protocol (XCAP)
[76]	3GPP TS 26.441	Codec for Enhanced Voice Services (EVS); General overview
[77]	3GPP TS 26.442	Codec for Enhanced Voice Services (EVS); ANSI C code (fixed-point)
[78]	3GPP TS 26.443	Codec for Enhanced Voice Services (EVS); ANSI C code (floating-point)
[79]	3GPP TS 26.445	Codec for Enhanced Voice Services (EVS); Detailed Algorithmic Description
[80]	3GPP TS 26.446	Codec for Enhanced Voice Services (EVS); AMR-WB Backward Compatible Functions
[81]	3GPP TS 26.447	Codec for Enhanced Voice Services (EVS); Error Concealment of Lost Packets
[82]	3GPP TS 26.449	Codec for Enhanced Voice Services (EVS); Comfort Noise Generation (CNG) Aspects
[83]	3GPP TS 26.450	Codec for Enhanced Voice Services (EVS); Discontinuous Transmission (DTX)
[84]	3GPP TS 26.451	Codec for Enhanced Voice Services (EVS); Voice Activity Detection (VAD)
[85]	3GPP TS 22.030	Man-Machine Interface (MMI) of the User Equipment (UE)
[86]	ITEF RFC 3840	Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)
[87]	3GPP TS 24.629	Explicit Communication Transfer (ECT) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification

2 IMS Feature Set

2.1 General

The IMS profile part lists the mandatory capabilities, that are required over the Gm and Ut reference points.

2.2 Support of generic IMS functions

2.2.1 SIP Registration Procedures

The UE and the IMS core network must follow the Session Initiated Protocol (SIP) registration procedures defined in 3GPP TS 24.229 [15]. In addition, when the conditions for performing IMS registration in bullets 2, 3, 4, 5 and 6 in section L.3.1.2 of 3GPP TS 24.229 [15] evaluate to true, then the UE must register with the IMS. Selective Disabling of 3GPP User Equipment Capabilities as defined in 3GPP TS 24.305 [18] is not mandated in this profile, therefore in the case where 3GPP TS 24.305 [18] Managed Object (MO) is not deployed, it is assumed that IMS is enabled in the terminal.

Note 1: UE registering with IMS in other situations is possible.

The UE must include IMS Communication Service Identifier (ICSI) value used to indicate the IMS Multimedia Telephony service, that being urn:urn-7:3gpp-service.ims.icsi.mmtel per 3GPP TS 24.173 [14], using procedures as defined in section 5.1.1.2.1 of 3GPP TS 24.229 [15]. The UE must also include the feature tag used to indicate SMS over IP service (see section 2.5 and A.7), that being +g.3gpp.smsip as defined in section 5.3.2.2 of 3GPP TS 24.341 [19]. If the UE is a Session Continuity UE (SC-UE) (e.g. due to support of SR-VCC as described in Annex A.3), then the UE must include the g.3gpp.accesstype media feature tag as specified in section 6.2.2 of Release 11 version of 3GPP TS 24.237 [16].

The UE must include the audio media feature tag, as defined in IETF RFC 3840 [86], in the Contact header field of the SIP REGISTER request, using procedures of 3GPP TS 24.229 [15].

The UE and the IMS core network must support network-initiated de-registration as defined in 3GPP TS 24.229 [15].

The UE must subscribe to the registration event package as defined in section 5.1.1.3 of 3GPP TS 24.229 [15].

The UE must include an IMEI URN (see 3GPP TS 23.003 [2] section 13.8) in the "+sip.instance" header field parameter (Instance ID) of the Contact address.

All IMS public user identities provided in the implicit registration set used for VoLTE by the IMS core network must be alias user identities and must include a tel URI (Uniform Resource Identifier). The following public user identity must be assigned to the implicit registration set used for VoLTE and it must be used by the UE when registering for VoLTE:

a) When ISIM is used, the public user identity in the first (or only) record in the Elementary File in the ISIM (see 3GPP TS 31.103 [44] section 4.2.4); or
b) The temporary public user identity derived from the IMSI (3GPP TS 24.229 [15]).

No other implicit registration set must be used for VoLTE.

Note 2: According to 3GPP TS 23.228 [7], a public user identity is an alias of another public user identity if both identities belong to the same implicit registration set, are linked to the same service profile and have the same service data configured for each and every service.

The UE must set the URI of the From header field of the REGISTER request for user-initiated reregistration or for user-initiated deregistration to the public user identity which was used in the URI of the From header field of the REGISTER request that created the binding being refreshed or being removed. The UE must set the URI of the To header field of the REGISTER request for user-initiated reregistration and for user-initiated deregistration to the public user identity that was used in the URI of the To header field of the REGISTER request that created the binding being refreshed or being removed.

Note 3: The "tag" header field parameter can differ in the From header field and in the To header field for the different REGISTER requests

For backwards compatibility the network must support all formats of URIs compliant with 3GPP TS 24.229 [15].

The UE must perform a re-registration prior to the expiry time of the existing registration as described in section 5.1.1.4.1 of 3GPP TS 24.229 [15].

If the UE receives a SIP 305 (Use Proxy) response to a re-registration, then the UE must acquire a P-CSCF different from the currently used P-CSCF and initiate a new initial registration as described in section 5.1.1.4.1 of 3GPP TS 24.229 [15]. If the UE receives a SIP 503 (Service Unavailable) response without a Retry-After header field, the SIP 503 (Service Unavailable) response must be treated as a SIP 500 (Server Internal Error) response (as stated in IETF RFC 3261 [55]) and the UE must initiate a new initial registration as described in section 5.1.1.4.1 of 3GPP TS 24.229 [15]. For the new initial registration, the UE must select a different P-CSCF from the P-CSCF list received from last PCO if not all of them have been attempted, otherwise the UE must re-establish a new PDN connection to the IMS well-known APN and get a new list of P-CSCFs (as stated in section 4.4) and choose from one of these P-CSCFs, as specified in section 5.1.1.4.1 of 3GPP TS 24.229 [15].

If the UE receives a SIP 503 (Service Unavailable) response or any other SIP 4xx, 5xx or 6xx response with Retry-After header as a response to an initial SIP REGISTER request, then the UE must re-attempt an initial registration via the same P-CSCF after the amount of time indicated in the Retry-After header field has expired or must immediately re-attempt an initial registration (as described above) when another P-CSCF is used.

Note 4: The above condition assumes that the UE has IP connectivity when the UE re-attempts an initial registration.

2.2.2 Authentication

The UE and the IMS core network must follow the procedures defined in 3GPP TS 24.229 [15] and 3GPP TS 33.203 [45] for authentication with IMS Authentication and Key Agreement (IMS-AKA), Sec-Agree and IPsec. Support of integrity protection is mandatory for both UE and network. Support of confidentiality protection is optional in the network, considering that lower layer security is available.

The IMS core network must support the procedures for IM Services Identity Module (ISIM) based authentication. Support for ISIM based authentication in the UE is mandatory.

The UE and IMS core network must support the procedures for USIM based authentication if there is no ISIM present on the Universal Integrated Circuit Card (UICC) as defined in Annex E.3.1 of 3GPP TS 23.228 [7] and Annex C.2 of 3GPP TS 24.229 [15]. This includes support for the P-Associated-URI header to handle barred IMS Public User Identities (IMPUs).

The UE and the IMS core network must support the procedures for authentication at the Ut reference point as specified in 3GPP TS 24.623 [28]. If the UE supports the Generic Authentication Architecture procedures specified in 3GPP TS 24.623 [28], 3GPP TS 33.222 [46] and 3GPP TS 24.109 [12], then the UE must construct the Bootstrapping Server Function (BSF) address as defined in section 16.2 of 3GPP TS 23.003 [2].

Note 1: It is recommended that the UE supports the Generic Authentication Architecture procedures specified in 3GPP TS 24.623 [28], 3GPP TS 33.222 [46] and 3GPP TS 24.109 [12].

The UE must support receiving an HTTP 2xx response to the HTTP request without being challenged by an HTTP 401 (Unauthorized) response.

Note 2: The above authentication scenario is possible only if the APN used for XCAP traffic (see section 4.3.1) is routed to the home network. The home network is able to authenticate the UE without challenging the UE for Ut authentication.

2.2.3 Addressing

The UE and IMS core network must support Public User Identities as defined in section 13.4 of 3GPP TS 23.003, which includes all of the following types of addresses:

- Alphanumeric SIP-URIs
    - Example: sip:voicemail@example.com
- MSISDN represented as a SIP URI:
    - Example: sip:+447700900123@example.com;user=phone
- MSISDN represented as a tel URI:
    - Example: tel:+447700900123

Note 1: Further requirements for support of Public User Identities in the network are specified in IR.65 [65].

The UE and IMS core network must support the local numbers as defined in Alternative 2 in Sections 5.1.2A.1.3 and 5.1.2A.1.5 in 3GPP TS 24.229 [15]. That is, the UE must set the dial string containing the local number to the user part of SIP URI in the Request URI, and set the "user=phone" parameter, with the “phone-context” tel URI parameter to the user part.

The UE must set the “phone-context” parameter as defined in section 7.2A.10 in 3GPP TS 24.229 [15]. That is, for home local numbers the UE must set the “phone-context” parameter to the home domain name, as it is used to address the SIP REGISTER request. The UE and network must support geo-local numbers. The UE must set the “phone-context” parameter according to section 7.2A.10.3 in 3GPP TS 24.229 [15].

- Example of phone-context for geo-local number: if the visited network has MCC = 234, MNC = 15, and the home network has MCC = 567, MNC = 26, the “phone context” parameter is set to the string “234.15.eps.ims.mnc026.mcc567.3gppnetwork.org”

Note 2: The UE on E-UTRAN knows the access information and hence the “phone-context” can be set accordingly.

The UE and IMS core network must support the P-Called-Party-ID header field; the network must use this header field as defined in 3GPP TS 24.229 [15].

The support of Globally Routable User agent URIs (GRUUs) by UE or network is not required.

2.2.4 Call Establishment and Termination

The UE and the IMS core network must follow 3GPP TS 24.229 [15] for establishment and termination of a call.

The UE and the IMS core network must support reliable provisional responses.

The UE must be able to accept a SIP INVITE request without a Session Description Protocol (SDP) offer and the UE must include an SDP offer with audio media in the first non-failure reliable response to a SIP INVITE request without SDP offer.

Note 1: Other media can be included in the SDP offer in the first non-failure reliable response.

The UE must include the audio media feature tag, as defined in IETF RFC 3840 [86], in the Contact header field of the SIP INVITE request, and in the Contact header field of the SIP response to the SIP INVITE request, as specified in 3GPP TS 24.229 [15].

The UE must indicate in the SDP of the initial INVITE that the audio media is send-receive i.e. either by including the direction attribute “a=sendrecv” or by omitting the direction attributes. If the UE receives an initial INVITE that contains “a=sendrecv” or no direction attribute in the SDP offer, the UE must indicate “a=sendrecv” or no direction attribute in the SDP answer, regardless of the use of SIP preconditions framework or of the resource reservation status.

Note 2: Previous versions of 3GPP TS 24.229 [15] mandated the use of the SDP inactive attribute. 3GPP Release 12 TS 24.229 [15] does not prohibit specific services from using direction attributes to implement their service-specific behaviours.

For the purpose of indicating an IMS communication service to the network, the UE must use an ICSI value in accordance with 3GPP TS 24.229 [15]. The ICSI value used must indicate the IMS Multimedia Telephony service, which is urn:urn-7:3gpp-service.ims.icsi.mmtel, as specified in 3GPP TS 24.173 [14].

The usage of preconditions is discussed in Section 2.4.

If the user rejects an incoming call by invoking User Determined User Busy (UDUB) as described in 3GPP TS 22.030 [85], then the UE must send a SIP 486 (Busy here) response to the network.

Note 3: The appropriate SIP response to reject a call on all devices for a multiple-device scenario and operator and vendor specific services are out-of-scope of this document.

2.2.5 Forking

Forking in the network is outside the scope of the present document. However for inter-operability and forward-compatibility reasons, the UE must be ready to receive responses generated due to a forked request and behave according to the procedures specified in IETF RFC 3261 [55], section 4.2.7.3 of 3GPP TS 23.228 [7], 3GPP TS 24.229 [15] and section 4.7.2.1 of 3GPP Release 13 TS 24.628 [71]. Furthermore, the UE should be able to maintain at least forty (40) parallel early dialogs until receiving the final response on one of them and the UE must support receiving media on one of these early dialogs. If the originating UE needs to release an early dialog, the UE must send a BYE request within the early dialog to be released, in accordance with section 15 in IETF RFC 3261 [55], e.g. when the UE receives the first response that would create an early dialog it cannot maintain, the UE sends a BYE request on that early dialog without saving dialog data.

Note 1: An early dialog that is maintained is one where a SIP 18x response has been received and the early dialogue has not been terminated (e.g. by receipt of a SIP 199 response) prior to receiving a SIP 2xx response.

Note 2: Multiple early dialogs can occur as a result of forking or for other reasons such as announcements or services.

The IMS core network can support sending and the UE must support receiving a SIP CANCEL request including a Reason header field with values of:

- SIP; cause=200; text="Call completed elsewhere"
- SIP; cause=603; text="Declined"

for forked calls as defined in 3GPP Release 12 TS 24.229 [15].

2.2.6 The use of Signalling Compression

The UE must not use SIGCOMP when the initial IMS registration is performed in E-UTRAN access as specified in Release 10 3GPP TS 24.229 [15].

Note: Although this version of the profile focuses on E-UTRAN, if the initial IMS registration occurs in other IP Connectivity Accesses then SIGCOMP can be used by the UE.

2.2.7 Early media and announcements

The UE must behave as specified in section 4.7.2.1 of 3GPP Release 13 TS 24.628 [71].

In addition, he UE must support the P-Early-Media header field as defined in IETF RFC 5009 [74], and must include a P-Early-Media header field with the “supported” parameter to INVITE requests it originates as specified in 3GPP TS 24.229 [15].

The UE must also maintain an early media authorization state per dialog as described in IETF RFC 5009 [74].

As stated in 3GPP TS 24.628 [71], the UE must render locally generated communication progress information, if:

- an early dialog exists where a SIP 18x response to the SIP INVITE request other than 183 (Session Progress) response was received;
- no early dialog exists where the last received P-Early-Media header field as described in IETF RFC 5009 [12] contained "sendrecv" or "sendonly"; and
- in-band information is not received from the network.

2.2.8 SIP Session Timer

The UE must support and use IETF RFC 4028 [86] as follows:

- for an initial SIP INVITE request, the UE must include a Supported header with the option tag “timer” and must either insert Session-Expires header field with the delta-seconds portion set to 1800, or must not include the Session-Expires header field in the initial SIP INVITE request;
- if the UE receives a SIP 422 response to an INVITE request, the UE must follow the procedures of section 7.4 in IETF RFC 4028 [86];
- it is recommended that the UE does not include the "refresher" parameter in the Session-Expires header field of the SIP INVITE request. If the UE includes the "refresher" parameter in the Session-Expires header field of the SIP INVITE request, the UE must set the "refresher" parameter to “uac” ;
- if a received SIP INVITE request indicates support of the "timer" option tag, and does not contain the Session-Expires header field, the UE must include a Session-Expires header field with the delta-seconds portion set to the greater of 1800 or the value contained in the Min-SE header (if present in the received INVITE) and the "refresher" parameter with the value "uac" in SIP 2xx response to the SIP INVITE request;and
- if a received SIP INVITE request indicates support of the "timer" option tag, and contains the Session-Expires header field without "refresher" parameter, the UE must include the "refresher" parameter with the value "uac" in the Session-Expires header field of the SIP 2xx response to the SIP INVITE request, and must set the delta-seconds portion of the Session-Expires header field of the SIP 2xx response to the SIP INVITE request to the value indicated in the delta-seconds portion of the Session-Expires header field of the SIP INVITE request.

Note: The network can choose to influence the session timer negotiation by modifying any of the related header fields or header field parameters within the constraints of IETF RFC 4028 [86].

2.3 Supplementary Services

2.3.1 Supplementary Services Overview

Supplementary services must be supported as defined as part of 3GPP MMTel TS 24.173 [14], with the constraints described in this section.

The UE and the Telephony Application Server (TAS) must support the supplementary services listed in Table 2.1. The provisioning of these supplementary services for a subscriber is optional and is an operator decision.

Note 1: Support of other supplementary services is out of scope of this document.

Supplementary Service
Originating Identification Presentation 3GPP TS 24.607 [23]
Terminating Identification Presentation 3GPP TS 24.608 [24]
Originating Identification Restriction 3GPP TS 24.607 [23] (Note 2)
Terminating Identification Restriction 3GPP TS 24.608 [24] (Note 2)
Communication Forwarding Unconditional 3GPP TS 24.604 [20] (Note 2)
Communication Forwarding on not Logged in 3GPP TS 24.604 [20] (Note 2)
Communication Forwarding on Busy 3GPP TS 24.604 [20] (Note 2)
Communication Forwarding on not Reachable 3GPP TS 24.604 [20] (Note 2)
Communication Forwarding on No Reply 3GPP TS 24.604 [20] (Note 2)
Barring of All Incoming Calls 3GPP TS 24.611 [26] (Note 2)
Barring of All Outgoing Calls 3GPP TS 24.611 [26] (Note 2)
Barring of Outgoing International Calls 3GPP TS 24.611 [26] (Note 3)
Barring of Outgoing International Calls – ex Home Country 3GPP TS 24.611 [26] (Note 3)
Barring of Outgoing International Calls - When Roaming 3GPP TS 24.611 [26] (Note 3)
Barring of Incoming Calls - When Roaming 3GPP TS 24.611 [26] (Note 2)
Communication Hold 3GPP TS 24.610 [25]
Message Waiting Indication 3GPP TS 24.606 [22] (Note 2)
Communication Waiting 3GPP TS 24.615 [27] (Note 2)
Ad-Hoc Multi Party Conference 3GPP TS 24.605 [21] (Note 2)
Explicit Communication Transfer - Consultative 3GPP TS 24.629 [87] (Note 2)

Table 2.1 Supplementary services

Note 2: Recommended options are described in sections 2.3.3 – 2.3.12.

Note 3: Barring of International Calls is a 3GPP Release 9 feature.

2.3.2 Supplementary Service Configuration

For supplementary service configuration, the UE and IMS core network must support XCAP at the Ut reference point as defined in 3GPP TS 24.623 [28].

The home operator can configure the UE with an XCAP root URI as specified in 3GPP TS 24.623 [28]. If the UE has not been configured with an XCAP root URI, then the UE must construct an XCAP root URI as defined in section 13.9 of 3GPP TS 23.003 [2].

As XCAP User Identity (XUI) the UE must use the default public user identity received in P-Associated-URI header in the SIP 200 (OK) response for REGISTER.

When not registered with IMS, the UE must use the default public user identity received during the last successful registration as in Section 2.2.1 in this document.

If the UE receives an HTTP 404 (Not Found) response when attempting to access the entire *simservs* XML document (i.e. a node selector is not included in the Request-URI of the XCAP request), or the UE does not have a stored default public user identity, then:

- if the UE has an ISIM, then the UE must use the public user identity in the first (or only) record in the EFIMPU Elementary File in the ISIM (see section 4.2.4 of 3GPP TS 31.103 [44]) as XUI in further XCAP requests sent until the next successful IMS registration.
- if the UE has a USIM but not an ISIM, then the UE must use the temporary public user identity derived from the IMSI (see section 13.4B of 3GPP TS 23.003 [2]) as XUI in further XCAP requests sent until the next successful IMS registration.

Note 1: If the UE attempts to access a fragment of the simservs XML document (i.e. a node selector is included in the Request-URI of the XCAP request), and the UE receives a HTTP 404 (Not Found) response, the UE is allowed to continue attempting to access the simservs XML document. If the UE continues to receive a HTTP 404 (Not Found) response when attempting to access a fragment of the simservs XML document, the UE can attempt to access the entire simservs XML document to determine if the XUI is valid.

Note 2: If the XUI is derived from the IMPU stored on the ISIM or derived from the temporary IMPU, then the UE does not share such XUI with another UE in order to prevent the revealing of a potentially barred IMPU.

The UE must configure settings of one supplementary service only per XCAP request. If the supplementary service to be configured contains a <ruleset> element with multiple <rule> elements as defined in IETF RFC 4745 [73] (e.g. as for Communication Diversion (CDIV), Communication Barring (CB)), then the UE must modify at most one <rule> element of the supplementary service per XCAP request.

The UE must perform HTTP PUT and HTTP DELETE as conditional operations using the If-Match header field as defined in section 7.11 of IETF RFC 4825 [75].

When modifying a supplementary service, if there is an existing matching <rule> element, the UE must modify the child elements of the existing <rule> element. Otherwise, if no matching <rule> element is found, the UE must consider that the supplementary service is not provisioned for the user and must not insert a new <rule> element with a rule ID different from any existing rule ID in the XML document.

Note 3: For each supplementary service that is provisioned for the user, the home operator needs to provide the matching <rule> element in the initial XML document.

When deactivating a <rule> element for a supplementary service, and if there is a matching <rule> element without <rule-deactivated> condition, the UE must insert the <rule-deactivated> condition in the <conditions> element of the <rule> element.

A <rule> element matches a supplementary service if:

1. the supplementary service requires a <conditions> element, and the conditions (with exception of the <rule-deactivated> condition) included in the <conditions> element of the <rule> element are the same as the conditions of the <conditions> element required by the supplementary service; or
2. the supplementary service does not require a <conditions> element and the <rule> element:

   - does not contain a <conditions> element;
   - contains an empty <conditions> element; or
   - contains a <conditions> element containing solely the <rule-deactivated> condition.

The UE must not remove a <rule> element of a supplementary service profiled in this document.

2.3.3 Ad-Hoc Multi Party Conference

The UE and the IMS core network must support the procedures defined in 3GPP TS 24.605 [21] and clause 5.3.1.3.2 of 3GPP Release 10 TS 24.147 [13], with the clarifications defined in this sub section.

Note 1: As per section 4.2 of 3GPP TS 24.605 [21], the invocation and operation for conferencing is described in 3GPP TS 24.147 [13].

For conference creation, the UE and the IMS core network must support Three Way Session creation as described in section 5.3.1.3.3 of 3GPP TS 24.147 [13]. The UE must apply option 2b) when inviting the remote user to the conference. The UE must support the construction of the “Default Conference Factory URI for MMTel” as specified in clause 13.10 of 3GPP Release 12 TS 23.003 [2].

When inviting other users to a conference, the UE and the IMS core network must support the procedure described in section 5.3.1.5.3 of 3GPP TS 24.147 [13]. The UE must send the SIP REFER method by using the existing dialog for a conference session between the UE and the IMS core network (conference server). The UE must add the Replaces header to the Refer-to header field in the SIP REFER request, as described in section 5.3.1.5.3 of 3GPP TS 24.147 [13].

Note 2: In Three-Way session creation procedures, the UE has an existing session with the SIP REFER target.

The UE can and the IMS core network must support the procedures in 3GPP TS 24.605 [21] for subscription to conference state events. The SIP SUBSCRIBE to conference state events must be sent outside the SIP INVITE dialog between the UE and the conference server. If the SUBSCRIBE request outside the existing INVITE dialog is rejected by a SIP 403 (Forbidden) response, the UE must send a SUBSCRIBE request in the existing INVITE dialog, as specified in clause 5.3.1.2 of 3GPP Release 12 TS 24.147 [13]. The UE is recommended to send the SUBSCRIBE request in the INVITE dialog until the next initial IMS registration.

To ensure compatibility with UEs compliant with older versions of this specification the IMS core network must support SUBSCRIBE requests received within an INVITE dialog. The IMS core network can support all, or a subset of the elements and attributes specified in IETF RFC 4575 [61]. As a minimum, the IMS core network must support the following elements and attributes:

- Conference-info: entity
- Maximum-user-count
- Users
- User: entity
- Display-text
- Endpoint: entity
- Status (supported values: connected, disconnected, on-hold)

The UE and the IMS core network must support audio media for the conference session.

Note 3: Support of other media types is out of scope of the document.

Floor control for conferencing as described in section 8 of 3GPP TS 24.147 [13] is not required.

Consent procedures for list server distribution as described in section 5.3.1.7 of 3GPP TS 24.147 [13] are not required.

2.3.4 Communication Waiting

The UE and the IMS core network must support the terminal based service, as described in 3GPP TS 24.615 [27]. The network-based service is not required. The Communication Waiting (CW) indication as defined in section 4.4.1 of 3GPP TS 24.615 [27] is not required. The UE is required to support Alert-Info, with values as specified in 3GPP TS 24.615 [27].The UE must provide the ability for the user to activate, deactivate and interrogate the terminal based service without using UE-to-network signaling (e.g. XCAP/Ut).

2.3.5 Message Waiting Indication

The UE must and the IMS core network can support the Message Waiting Indication (MWI) event package, as defined in 3GPP TS 24.606 [22] and IETF RFC 3842 [60].

2.3.6 Originating Identification Restriction

The UE and the IMS core network must support the SIP procedures in 3GPP TS 24.607 [23]. Service configuration as described in Section 4.10 of 3GPP TS 24.607 [23], is not required.

2.3.7 Terminating Identification Restriction

The UE and the IMS core network must support the SIP procedures in 3GPP TS 24.608 [24]. Service configuration, as described in section 4.9 of 3GPP TS 24.608 [24], is not required.

2.3.8 Communication Diversion

The UE and the IMS core network must support the SIP procedures in 3GPP TS 24.604 [20] for Communication Diversion (CDIV). For CDIV service activation, deactivation, and interrogation (XCAP operations), the UE and the IMS core network must support the XML rules for Call Forwarding Unconditional and the conditions, actions and elements listed in Table 2.2.

The UE and the IMS core network must support the XML rules as described in section 4.9.1 of 3GPP TS 24.604 [20]. The UE must support the History-Info header for identification of diverting parties at the terminating side and of diverted-to parties at the originating side. At the terminating side, a History-Info entry must be used for the identification of the diverting party only if another History-Info entry exists that has assigned the next index in sequence *and* includes a cause value. At the originating side only History-Info entries including a cause value must be used for presentation of the diverted-to party.

Note 1: Support of subscription options and other conditions and actions are out of scope of the document.

Type	Parameter
Rule Condition	busy
Rule Condition	media (supported media types: audio, audio AND video)
Rule Condition	no-answer
Rule Condition	not-registered
Rule Condition	not-reachable (Note 2)
Rule Condition	rule-deactivated
Rule Action	target
Element	NoReplyTimer

Table 2.2 Supported conditions, actions and elements in CDIV

Note 2: The CS version of Communication Forwarding on Not Reachable (CFNRc) implies diversion when the user is not registered in the CS core or cannot be reached. To mimic this behaviour, it is recommended that an UE activates both the CFNRc (CDIV using condition not-reachable) and the Communication Forwarding on Not Logged-in (CFNL) (CDIV using condition not-registered) to the same target.

In addition to the requirements in section 2.3.2, when configuring settings for the Communication Diversion supplementary service the UE must configure only one of the following in an XCAP request:

- Communication diversion supplementary service activation, no-reply-timer or both.

    - If the <cp:ruleset> element is present, and a <NoReplyTimer> element is to be created, the UE must include the <cp:ruleset> element in the HTTP PUT request.

- For the communication diversion services supported in this PRD, elements of one <rule> element for communication diversion supplementary service only.

Note 3: It is not possible to create a no-reply timer without including the rule set in a document where a rule set already exists since this would create an invalid XML document according to IETF RFC 4825 [75].

If:

- a <rule> element matching a CDIV service exists in the XML document; and
- the <target> child element contains empty string;

then the UE must consider that the CDIV service is not registered for the user otherwise the UE must consider that the CDIV service is registered for the user.

2.3.9 Communication Barring

The UE and the IMS core network must support the SIP procedures in 3GPP TS 24.611 [26]. For service activation, deactivation, and interrogation (XCAP operations), the UE and the IMS core network must support the XML rules for Barring of All Incoming Calls, Barring of All Outgoing Calls and the conditions listed in Table 2.3. The UE and the IMS core network must support the XML rules as described in section 4.9.1.3 of 3GPP TS 24.611 [26].

Note: Support of other conditions is out of scope of the document.

Condition
roaming
international
International-exHC
rule-deactivated

Table 2.3 Supported conditions in CB

In addition to the requirements in section 2.3.2, when configuring settings for the Communication Barring supplementary service the UE must modify only one of the following in an XCAP request:

- Incoming communication barring supplementary service activation
- Outgoing communication barring supplementary service activation
- For the communication barring services supported in this PRD, elements of one <rule> element for communication barring supplementary service only.

2.3.10 Communication Hold

The UE invoking the HOLD service must not send any media to the other party.

2.3.11 Explicit Communication Transfer - Consultative

The UE must and IMS core network can support the procedures for the consultative transfer defined in 3GPP Release 12 TS 24.629 [87], with the clarifications defined in this sub section.

The UE as a Transferee must support the procedures with 3rd Party Call Control (3PCC) as defined in 3GPP Release 11 TS 24.628 [71]. The UE procedure without 3PCC is not required.

The UE and IMS core network must support audio media for the transferred session.

2.3.12 Originating Identification Presentation

The UE and IMS core network must support the SIP procedures in 3GPP Release 13 TS 24.607 [23].

The UE must support the presentation of the originating user identity both from the identity within the P-Asserted-Identity header field and the identity within the From header field.

In the presence of both P-Asserted-Identity and From header fields, the UE must use the identity within the P-Asserted-Identity header field as the originating user identity, unless the home operator preconfigured the UE to use the identity within the From header field as the originating user identity.

In the absence of the P-Asserted-Identity header field, the UE must use the unavailable user identity specified in section 13.7 of 3GPP TS 23.003 [2] as the originating user identity, unless the home operator preconfigured the UE to use the identity within the From header field as the originating user identity.

Note: As, by default, the identity in the From header field may not be network asserted, it is the responsibility of the network to ensure that the From header field contains a reliable identity of the originating user when it is sent to the UE.

2.4 Call Set-up Considerations

2.4.1 SIP Precondition Considerations

Unless preconfigured otherwise by the home operator, the UE must support and use the SIP preconditions framework, as specified in 3GPP Release 13 TS 24.229 [15].

The network may disable the use of preconditions in the network; the means by which this takes place is outside the scope of this document.

The terminating UE implementation must not rely on the use of preconditions by the originating UE.

Upon receiving an INVITE request, when preconditions are not used by the originating UE or preconditions are disabled by the network, and the local resources required at the terminating UE are not available, the terminating UE, according to 3GPP Release 13 TS 24.229 [15], must:

- send a SIP 183 (Session Progress) response containing SDP;
- not alert the user until resources are reserved successfully on the terminating side; and
- not send a SIP 180 (Ringing) response until resources are reserved successfully on the terminating side.

2.4.2 Integration of resource management and SIP

2.4.2.1 Loss of PDN connectivity

If the Packet Data Network (PDN) connectivity between a UE and the network is lost, the network must terminate all ongoing SIP sessions related to this UE, according to the procedures in section 5.2.8 of 3GPP TS 24.229 [15] (e.g. when the P-CSCF receives an abort session request from the Policy and Charging Rules Function (PCRF)).

If the UE discovers (for example during a TAU procedure) that PDN connectivity had been lost, then the UE must attempt to re-establish the PDN connection. This will trigger the network to initiate a new SIP signalling bearer in conjunction with the PDN connection establishment.

Note: The PDN connectivity may also be lost if the UE moves to GERAN/UTRAN, see also GSMA PRD IR.88 [67]. It may not be possible to re-establish the PDN connectivity in GERAN/UTRAN in such deployments.

When the UE regains PDN and IP connectivity, if the IP address has changed or the IMS registration expired during the period of absence of IP connectivity then the UE must perform a new initial registration to IMS.

2.4.2.2 Void

2.4.2.3 Loss of media bearer and Radio Connection

If a Guaranteed Bit Rate (GBR) bearer used for voice fails to get established, or is lost mid-session, then the network must terminate the session associated to the voice stream according to the procedures in section 5.2.8 of 3GPP TS 24.229 [15] (P-CSCF must be informed about loss of bearer by the PCRF).

Note 1: The loss of the GBR bearer may be due to loss of radio connection indicated by an S1 release with cause “Radio Connection With UE Lost” and then followed by the MME Initiated Dedicated Bearer Deactivation procedure for the GBR bearer used for voice. Or, the GBR bearer may be lost or not established, due to the current resource and radio situation. However, termination of the SIP session due to loss of the voice GBR bearer is the only way for the system to stop the IMS level charging (quickly) when the UE loses radio connection.

Note 2: If other media types are used, and a GBR bearer used for another media type fails to get established, or is lost mid-session, then the network, based on its policies, has the option to either allow the SIP session to continue as is, or terminate the SIP session that the GBR bearer is associated with (the network can handle loss of video in a video call in such a way that the session continues as voice-only).

If a SIP session includes media streams, and if a dedicated bearer for any media stream fails to get established, or is lost mid-session, then the UE must, based on its preferences, modify, reject or terminate the SIP session that the dedicated media bearer is associated with, according to section 6.1.1 of 3GPP TS 24.229 [15]. The UE can act differently per media type.

Note 3: If a voice bearer is lost or fails to get established, the network will, in normal cases, release the session as described in the beginning of this section. As a complement to this, the UE must have internal logic to react to the detection of loss of bearer/radio connection to handle its internal state. For a multimedia communication, if the radio connection is not lost, but a bearer not used for voice is lost, then the UE must decide if the session should be maintained as is, should be modified, or should be released.

If the UE loses radio connectivity and the IMS registration expires prior to regaining radio connectivity, then upon regaining radio connectivity the UE must perform a new initial registration to IMS.

2.4.3 Voice Media Considerations

The SDP offer/answer for voice media must be formatted as specified in section 6.2.2 of 3GPP TS 26.114 [35], with the restrictions included in the present document. If the Enhanced Voice Services (EVS) codec is included, then the offer/answer for voice media must be formatted as specified in section 6.2.2 of 3GPP Release 12 TS 26.114 [35], with the restrictions included in the present document.

If multiple audio bandwidths are offered by the UE for speech communication, then the codec preference order must be as specified in clauses 5.2.1.5 and 5.2.1.6 of 3GPP Release 12 TS 26.114 [35].

Unless otherwise preconfigured by the home operator, if a dedicated bearer for the media does not exist, the UE must consider itself not having local resources. If the UE has no local resources, the UE must not send media. See also section L.2.2.5.1B in 3GPP TS 24.229 [15].

Note: The existence of a dedicated bearer does not grant by itself the UE authority to send media. Other conditions need to be fulfilled.

2.4.4 Multimedia Considerations

UEs using the full set of media functions can send SDP offers containing multiple “m=” lines to indicate the wish to establish a more advanced multimedia session than this profile defines.

If one of these “m=” lines indicates the wish of establishing an audio (voice) session (using a compatible codec), then the UE following this profile must accept the offer and allow the use of whatever media streams it supports. The UE must set the port number to 0 (zero) for the media streams it does not support.

Note 1: This means that a voice-only UE will accept a video call request, but the call will automatically be transformed to a voice-only call. In CS telephony, the call is rejected when the terminating client cannot support all offered media (that is a voice-only terminal will reject a video call offer). Hence, this section describes a behaviour that is new to telephony.

UEs using the full set of media functions, have the option to try to update the session by sending SIP (re-)INVITE requests that include SDP offers containing multiple “m=” lines, to indicate the desire to expand the session into a more advanced multimedia session. The UE following this profile must accept such an offer and allow the use of whatever media streams the UE supports. The UE must, in the SDP answer, set the port number to 0 (zero) for the media streams it does not support.

Note 2: This means that a voice-only capable UE will accept a request to update the session to video using a SIP 200 (OK) response. But since the SDP answer will disable the video stream, the call will continue as a voice-only call.

2.5 SMS over IP

The UE must implement the roles of an SM-over-IP sender and an SM-over-IP receiver, according to sections 5.3.1 and 5.3.2 of 3GPP TS 24.341 [19].

The status report capabilities, delivery reports, and notification of having memory available, according to sections 5.3.1.3, 5.3.2.4 and 5.3.2.5 of 3GPP TS 24.341 [19] must be supported by the UE and the IMS core network.

The IMS core network must take the role of an IP-SM-GW and support the general procedures in section 5.3.3.1 of 3GPP TS 24.341 [19], and the following functions:

- answering of routing information query and obtaining the routing information according to the procedures in section 5.3.3.3 in 3GPP TS 24.341 [19]; and
- transport layer interworking according to sections 5.3.3.4 of 3GPP TS 24.341 [19].

3 IMS Media

3.1 General

This section endorses a set of media capabilities specified in 3GPP TS 26.114 [35]. The section describes the needed SDP support in UEs and in the IMS core network and it describes the necessary media capabilities both for UEs and for entities in the IMS core network that terminate the user plane. Examples of entities in the IMS core network that terminate the user plane are the Media Resource Function Processor (MRFP) and the Media Gateway (MGW).

3.2 Voice Media

3.2.1 Codecs

The UE must support the Adaptive Multi-Rate (AMR) speech codec, as described in 3GPP TS 26.071 [29], 3GPP TS 26.090 [31], 3GPP TS 26.073 [30], and 3GPP TS 26.104 [34], including all eight (8) modes and source rate controlled operations, as described in 3GPP TS

26.093 [32]. The UE must be capable of operating with any subset of these eight (8) codec modes. The UE must support handling of CMR within RTP payload as specified in clause 7.5.2.1.2.2 of 3GPP Release 13 TS 26.114 [35].

The UE must support the AMR wideband (AMR-WB) speech codec as described in 3GPP TS 26.114 [35], 3GPP TS 26.171 [38], 3GPP TS 26.190 [40], 3GPP TS 26.173 [39] and 3GPP TS 26.204 [42], including all nine (9) modes and source controlled rate operation 3GPP TS 26. 193[41]. The UEmust be capable of operating with any subset of these nine (9) codec modes. If the EVS codec is supported, then the EVS AMR-WB IO mode of operation may be used as an alternative implementation of AMR-WB as specified in clause 5.2.1.4 of 3GPP Release 12 TS 26.114 [35]. The UE must support handling of CMR within RTP payload as specified in clause 7.5.2.1.2.2 of 3GPP Release 13 TS 26.114 [35].

If super-wideband or fullband speech communication is offered, then the UE must support the EVS codec as described in 3GPP Release 12 TS 26.114 [35], 3GPP Release 12 TS 26.441 [76], 3GPP Release 12 TS 26.445 [79], 3GPP Release 12 TS 26.442 [77], 3GPP Release 12 TS 26.443 [78], 3GPP Release 12 TS 26.447 [81], 3GPP Release 12 TS 26.449 [82], 3GPP Release 12 TS 26.450 [83] and 3GPP Release 12 TS 26.451 [84].

When transmitting using the AMR codec, the AMR-WB codec or the EVS codec in the AMR-WB IO mode mode of operation, then the UE must be capable of aligning codec mode changes to every frame border, and must also be capable of restricting codec mode changes to be aligned to every other frame border e.g. as described for UMTS_AMR_2 in 3GPP TS 26.103 [33] based on the SDP offer-answer negotiation. The UE must also be capable of restricting codec mode changes to neighbouring codec modes within the negotiated codec mode set based on the SDP offer-answer negotiation.

When receiving using the AMR codec, the AMR-WB codec or the EVS codec in the AMR-WB IO mode mode of operation, then the UE and the entities in the IMS core network that terminate the user plane must allow codec mode changes at any frame border and to any codec mode within the negotiated codec mode set. As an exception, entities in the network that provide Circuit Switched (CS) interworking and apply Transcoder-Free Operation (TrFO) of Tandem-Free Operation (TFO) must accept codec mode changes in accordance with the capabilities at the CS network side.

Entities in the IMS core network that terminate the user plane supporting speech communication and supporting TFO and/or TrFO must support:

- AMR speech codec modes 12.2, 7.4, 5.9 and 4.75 as described in 3GPP TS 26.071 [29], 3GPP TS 26.090 [31], 3GPP TS 26.073 [30], and 3GPP TS 26.104 [34].

Entities in the IMS core network that terminate the user plane supporting wideband speech communication and supporting TFO and/or TrFO must support:

- AMR-WB speech codec modes 12.65, 8.85 and 6.60 as described in 3GPP TS 26.171 [38], 3GPP TS 26.190 [40], 3GPP TS 26.173 [39], and 3GPP TS 26.204 [42].

Entities in the IMS network that provide transcoding-free interworking to the CS network must be capable of requesting the UE to restrict codec mode changes to be aligned to every other frame border and also be capable of requesting the UE to restrict codec mode changes to neighbouring codec modes within the negotiated codec mode set.

Note: Restrictions in codec mode changes are required only for transcoder-free interworking with a CS GSM MS (Mobile Station).

3.2.2 RTP Profile and SDP Considerations

3.2.2.1 RTP Profile

The Real Time Protocol (RTP) profile, Audio Video Profile (AVP) IETF RFC 3551 [57] must be used by the UE and the IMS core network.

3.2.2.2 SDP Offer Considerations

The SDP Capability Negotiation framework described in IETF RFC 5939 [64] must not be used in the SDP offer by the UE and the IMS core network when the AVP profile is used.

3.2.2.3 SDP Answer Considerations

The UE and the IMS core network must be able to receive and answer to an SDP offer that uses SDPCapNeg. The answer must indicate the use of the RTP AVP profile.

Note: In section 6.2.1a of 3GPP TS 26.114 [35], it is recommended that that a UE or the IMS core network use the SDPCapNeg attributes 'tcap' and 'pcfg' to indicate the support of both the RTP profiles AVP and AVP Feedback Profile (AVPF). Hence, to be forward compatible with equipment using the full set of media functions, a minimum set UE and the IMS core network must be able to ignore the SDPCapNeg attributes and answer to the RTP AVP profile in the offer.

3.2.2.4 SDP Bandwidth Negotiation

The UE and network must use the b=AS parameter in SDP offers and answers for bandwidth negotiation as defined in section 6.2.5.2 of 3GPP Release 10 TS 26.114 [35] for UEs and networks that support AMR and AMR-WB, and 3GPP Release 12 TS 26.114 [35] for UEs and networks that support EVS.

3.2.3 Data Transport

The UE and the entities in the IMS core network that terminate the user plane must use RTP over UDP as described in IETF RFC 3550 [56] and IETF RFC 768 [53], respectively, to transport voice and use symmetric RTP as defined in IETF RFC 4961 [72].

3.2.4 RTCP Usage

The RTP implementation must include an RTP Control Protocol (RTCP) implementation according to IETF RFC 3550 [56].

The UE and the entities in the IMS core network that terminates the user plane must use symmetric RTCP as defined in IETF RFC 4961 [72].

The bandwidth for RTCP traffic must be described using the "RS" and "RR" SDP bandwidth modifiers at media level, as specified by IETF RFC 3556 [58]. Therefore, a UE must include the "b=RS:" and "b=RR:" fields in SDP, and a UE and the entities in the IMS core network that terminate the user plane must be able to interpret them. If the “b=RS:” field or “b=RR:” field or both these fields are not included in a received SDP (offer or answer), then the UE must use the recommended default value for the missing field(s) as defined in IETF RFC 3556 [58].

The UE and the entities in the IMS core network that terminate the user plane must send RTCP packets when media (including early media) is sent or received. Once an RTCP packet is sent according to received SDP of a SIP dialog, RTCP packets must be sent by UEs and entities in the IMS core network that terminate the user plane according to the received SDP of the SIP dialog for the remaining duration of the SIP dialog. For uni-directional media (e.g. early media or during call hold), RTCP packets must always be sent by both UEs and entities in the IMS core network that terminate the user plane. If multiple early dialogs are created due to forking (see section 2.2.5), the UE must send the RTCP packets according to received SDP answers of those early dialogs for which the IP address and port received in the SDP match the IP address and port of received media.

Note 1: The RTCP is based on the periodic transmission of control packets to all participants in the session, as described in IETF RFC 3550 [56]. In the context of this document, the primary uses of RTCP are voice quality monitoring, and to provide link aliveness information while the media are on hold. The latter implies that the RTCP transmission must continue when the media are on hold.

The UE and the entities in the IMS core network that terminates the user plane must set the sending frequency of control packets to a value calculated from the values of "RS" and "RR" SDP bandwidth modifiers according to rules and procedures in IETF RFC 3550 [56]. The UE must set the "RS" and "RR" SDP bandwidth modifiers such that RTCP packets are sent to the UE at least once every 5 seconds, in order to allow a sufficiently tight inactivity detection.

The UE and the entities in the IMS core network that terminate the user plane must support the transmission of RTCP packets formatted according to the rules in IETF RFC 3550 [56] and with the following clarifications below.

The UE and the entities in the IMS core network that terminate the user plane must use the RTCP compound packet format. When sent, the compound packet must include one report packet and one Source Description (SDES) packet. When no RTP packets have been sent in the last two reporting intervals, the UE and the entities in the IMS core network that terminate the user plane should send a Receiver Report (RR). Receiving of a Sender Report (SR) instead of an RR must be handled and accepted as valid by the UE and the entities in the IMS core network that terminate the user plane.

The SR, RR and SDES packets must be formatted as described below:

For SR and RR RTCP packets:

- Version 2 must be used; and
- Padding must not be used (and therefore padding bit must not be set).

For SDES RTCP packets:

- version and Padding as described for SR packet must be used;

- the SDES item CNAME must be included in one packet; and
- other SDES items should not be used.

Note 2: Because the randomly allocated SSRC identifier may change, the CNAME item must be included to provide the binding from the SSRC identifier to an identifier for the source that remains constant. Like the SSRC identifier, the CNAME identifier must be unique among all other participants within one RTP session.

To be forward compatible and interwork with legacy equipment, the UE and the entities in the IMS core network that terminate the user plane must be able to receive all types of RTCP packets, according to the rules specified in IETF RFC 3550 [56].

RTCP is controlled on a per session basis by the SDP offer/answer exchange as defined in 3GPP TS 26.114 [35] with the following clarifications:

- If the UE receives an SDP offer that contains “b=RS” attribute set to zero, then the UE must set the “b=RS” attribute to zero in an SDP answer to that SDP offer. If the UE receives an SDP offer that contains “b=RR” attribute set to zero, then the UE must set the “b=RR” attribute to zero in an SDP answer to that SDP offer. If the UE receives an SDP offer that contains both "b=RR" and "b=RS" attributes set to zero, then the UE must not send RTCP packets and must consider RTCP to be disabled for the session.
- If the UE received an SDP answer containing zero values in both of the “b=RS” and “b=RR” attributes, then (regardless of the values assigned to these attributes in the corresponding SDP offer) the UE must not send RTCP packets and must consider RTCP to be disabled for the session.
- The UE must accept receiving RTCP packets for a session that the UE considers RTCP to be disabled. The UE is not required to process these received RTCP packets.

3.2.5 Speech Payload Format Considerations

The Adaptive Multi Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) payload format(s) specified in IETF RFC 4867 [63] must be supported by the UE and the entities in the IMS core network that terminate the user plane. If the EVS codec is supported, then the EVS payload format specified in 3GPP Release 12 TS 26.445 [79] must be supported.

The UE and the entities in the IMS core network that terminates the user plane must support the bandwidth-efficient and the octet-aligned formats of the AMR and AMR-WB payload formats. The UE and the entities in the IMS core network that terminates the user plane must request the use of bandwidth-efficient format of the AMR and AMR-WB payload format when originating a session.

The UE and the entities in the IMS core network that terminates the user plane must send the number of speech frames, or fewer, encapsulated in each RTP packet, as requested by the other end using the ptime SDP attribute.

The UE and the entities in the IMS core network that terminates the user plane must request to receive one speech frame encapsulated in each RTP packet, but must accept any number of frames per RTP packet, up to the maximum limit of 12 speech frames per RTP packet.

Note 1: This means that the ptime attribute must be set to 20 and the maxptime attribute must be set to 240 in the SDP negotiation.

An IMS MGW not supporting redundancy may limit the maxptime attribute to 80 in the SDP negotiation.

The UE and the entities in the IMS core network that terminates the user plane must be able to sort out the received frames based on the RTP Timestamp and must remove duplicated frames, if present. If multiple versions of a frame are received, e.g. encoded with different bit rates, then the frame encoded with the highest bit rate should be used for decoding.

Note 2: UEs and the entities in the IMS core network that terminate the user plane, using the full set of media functions, have the option to send frames several times (for redundancy) to adapt for conditions with high packet-loss ratios. It is thus important that a UE and the entities in the IMS core network that terminate the user plane that use this profile are capable to detect and drop the duplicated frames.

RTCP-APP must not be used for Codec Mode Requests (CMR) by the UE and the entities in the IMS core network that terminate the user plane.

Note 3: As the speech media uses the RTP AVP profile as specified in section 3.2.2.1, the adaptation using RTCP may be too slow and therefore unsuitable.

3.2.6 Jitter Buffer Management Considerations

The minimum performance requirements for jitter buffer management of voice media, as described in 3GPP TS 26.114 [35] must be met. If the EVS codec is supported, then the jitter buffer management requirements in section 8.2 of 3GPP Release 12 TS 26.114 [35] must be met.

3.2.7 Front End Handling

UEs used for IMS voice services must conform to the minimum performance requirements on the acoustic characteristics of 3G terminals specified in 3GPP TS 26.131 [36]. The codec modes and source control rate operation (DTX) settings must be as specified in 3GPP TS 26.132 [37].

3.3 DTMF Events

The UE and the IMS core network must support DTMF events as defined in Annex G of 3GPP TS 26.114 [35].

4 Radio and Packet Core Feature Set

4.0 General

The LTE radio capabilities included in this specification are applicable to UEs and networks supporting FDD LTE only, TDD LTE only, or both FDD LTE and TDD LTE.

4.1 Robust Header Compression

The UE and the network must support Robust Header Compression (RoHC) as specified in 3GPP TS 36.323 [51], IETF RFC 3095 [54] and IETF RFC 4815 [62]. The UE and network must be able to apply the compression to packets that are carried over the radio bearer dedicated for the voice media. At minimum, the UE and network must support "RTP/UDP/IP" profile (0x0001) to compress RTP packets and "UDP/IP" profile (0x0002) to compress RTCP packets. The UE and network must support these profiles for both IPv4 and IPv6.

4.2 LTE Radio Capabilities

4.2.1 Radio Bearers

The UE must support the following combination of radio bearers (see Annex B in 3GPP TS 36.331 [52]):

- SRB1 + SRB2 + 4 x AM DRB + 1 x UM DRB

The network must support the following combination of radio bearers:

- SRB1 + SRB2 + 2 x AM DRB + 1 x UM DRB

One AM Data Radio Bearer (DRB) is utilized for an Evolved Packet System (EPS) bearer with Quality of Service Class Indicator (QCI) = 5 and another AM DRB for EPS bearer with QCI = 8/9. UM DRB is utilized for EPS bearer with QCI = 1. EPS bearer usage is described in section 4.3.

4.2.2 DRX Mode of Operation

In order to maximize lifetime of the UE battery, the UE and the network must support LTE Discontinuous Reception (DRX) method as specified in 3GPP TS 36.300 [49] and 3GPP TS 36.321 [50] must be deployed.

4.2.3 RLC configurations

Radio Link Control (RLC) entities must be configured to perform data transfer in the following modes as specified in 3GPP TS 36.322 [69]:

- Unacknowledged Mode (UM) for EPS bearers with QCI = 1
- Acknowledged Mode (AM) for EPS bearers with QCI = 5
- Acknowledged Mode (AM) for EPS bearers with QCI = 8/9

Voice service can tolerate error rates on the order of 1%, while benefiting from reduced delays, and is mapped to a radio bearer running the RLC protocol in unacknowledged mode (UM).

EPS bearer usage is described in section 4.3.

4.2.4 GBR and NGBR Services, GBR Monitoring Function

Voice is one of the LTE services that require a GBR bearer, although it is a very low data rate compared to LTE peak rates, as described in 3GPP TS 23.401 [10]. The GBR bearer for voice requests dedicated network resources related to the GBR for AMR codec values. The network resources associated with the EPS bearer supporting GBR must be permanently allocated by an admission control function in the eNodeB at bearer establishment. Reports from the UE, including buffer status and measurements of UE's radio environment, must be required to enable the scheduling of the GBR as described in 3GPP TS 36.300 [49]. In UL it is the UE's responsibility to comply with GBR requirements.

The non-GBR bearer (NGBR) does not support a guaranteed bit rate over the radio link and is thus not suitable for IMS-based voice services.

4.3 Bearer Management

4.3.1 EPS Bearer Considerations for SIP Signalling and XCAP

For SIP signalling, the IMS application in the UE must use the IMS well-known APN as defined in GSMA PRD IR.88 [67]; the UE must prevent non-IMS applications from using the IMS well-known APN.

Unless preconfigured by the home operator to provide the IMS well-known APN during initial attach, the UE must not provide the IMS well-known APN during the E-UTRAN initial attach procedure.

Note 1: The network has to be prepared to receive any APN, including the IMS well-known APN, during the E-UTRAN initial attach procedure, as per 3GPP TS 23.401 [10] and 3GPP TS 24.301 [17].

Note 2: When preconfiguring the UE to provide the IMS well-known APN during initial attach, the home operator needs to ensure that the IMS well-known APN is part of the subscription of the user of the UE in order to avoid attach failure. How the home operator preconfigures the UE is out of the scope of this PRD.

If procedures in section 2.2.1 require the UE to register with IMS, and the PDN connection to the IMS well-known APN does not exist yet (e.g. when the PDN connection established during the initial attach is to an APN other than the IMS well-known APN then the UE must establish a PDN connection to the IMS well-known APN.

Note 3: PDN Connection establishment can be caused by a SIP registration request. Sending a SIP registration request per the note in section 2.2.1 can cause PDN Connection establishment even if the IMS voice over PS Session indicator indicates that IMS voice over PS session is not supported.

Note 4: For all cases when the UE provides the IMS well-known APN, the APN Operator Identifier is not included by the UE.

A default bearer must be created when the UE creates the PDN connection to the IMS well-known APN, as defined in 3GPP specifications. A standardised QCI value of five (5) must be used for the default bearer. It is used for IMS SIP signalling.

The UE must and the network can support T3396 IE in PDN Connectivity Reject and the UE must start the ESM back-off timer according to the value indicated by the T3396 IE as specified in 3GPP Release 12 TS 24.301 [17]. If the T3396 IE is not included in a PDN Connectivity Reject, and the PDN Connectivity Reject is for a standalone PDN CONNECTIVITY REQUEST, then the UE must apply a default value of 12 minutes for the ESM back-off timer for the cause values #8 "operator determined barring", #27 "missing or unknown APN", #32 "service option not supported", and #33 "requested service option not subscribed" as described in section 6.5.1.4.3 of 3GPP Release 12 TS 24.301 [17].

For XCAP requests, the UE must be preconfigured or provisioned by the home operator with the Network Identifier part of the APN for Home Operator Services to be used for these requests (see GSMA PRD IR.88 [67] for more information).

Note 5: How the home operator preconfigures or provisions the UE with the Network Identifier part of the APN for Home Operator Services is out of the scope of this PRD.

4.3.2 EPS Bearer Considerations for Voice

For an IMS session request for a Conversational Voice call (originating and terminating), a dedicated bearer for IMS-based voice must be created utilising interaction with dynamic PCC. The network must initiate the creation of a dedicated bearer to transport the voice media. The dedicated bearer for Conversational Voice must utilise the standardised QCI value of one (1) and have the associated characteristics as specified in 3GPP TS 23.203 [4]. Since the minimum requirement for the UE is the support of one (1) UM bearer that is used for voice (see section 7.3.1 and Annex B of 3GPP TS 36.331 [52]), the network must not create more than one dedicated bearer for voice media. Therefore, the UE and network must be able to multiplex the media streams from multiple concurrent voice sessions.

Note 1: A single bearer is used to multiplex the media streams from multiple concurrent voice sessions; this is necessary in some supplementary services (e.g. CW, CONF).

Note 2: The sharing of a single GBR bearer for voice means that different QCI and/or ARP values are not possible for different voice streams.

When the UE has an ongoing conversational voice call, the UE must follow the procedures for access domain selection related to “Persistent EPS bearer context” as specified in sections 5.5.3.2.4 and 5.5.3.3.4.3 of 3GPP Release 10 TS 24.301 [17], sections 5.1.3.1 and L.2A.0 of 3GPP Release 10 TS 24.229 [15], and section 8.2 of 3GPP Release 10 TS 24.237 [16].

For IMS session termination of a Conversational Voice call, the dedicated bearer must be deleted utilising interaction with dynamic PCC. The network must initiate the deletion of the bearer.

4.4 P-CSCF Discovery

The UE and packet core must support the procedures for P-CSCF discovery via EPS. These are described in Annex L.2.2.1of 3GPP TS 24.229 [15], as option II for P-CSCF discovery.

The UE must indicate P-CSCF IPv6 Address Request and P-CSCF IPv4 Address Request when performing the following procedures (see also section 4.3.1):

- during the initial attach when establishing PDN connection to the default APN;
- during the initial attach when establishing PDN connection to the IMS well-known APN;
- during the establishment of the PDN connection to the IMS well-known APN when already attached;
- during the attach procedure for emergency bearer services; and
- during the establishment of the PDN connection for emergency bearer services when already attached.

The UE must use the P-CSCF addresses received during PDN connection establishment to the IMS well-known APN when accessing non-emergency services, and must use the P-CSCF addresses received during PDN connection establishment for emergency bearer services when accessing emergency services, as defined in section 5.1 of this document and 3GPP TS 24.229 [15].

If the UE receives a Modify EPS Bearer Context Request message containing a list of P-CSCF addresses that do not include the address of the currently used P-CSCF, the UE must acquire a P-CSCF different from the currently used P-CSCF and initiate a new initial registration as described in section L.2.2.1C 3GPP Release 12 TS 24.229.

Note: The above behavior can result in any ongoing calls being released.

5 Common Functionalities

5.1 IP Version

The UE and the network must support both IPv4 and IPv6 for all protocols that are used: SIP, SDP, RTP, RTCP and XCAP/HTTP. At initial attach and PDN connection establishment, the UE must request the PDN type IPv4v6, as specified in section 5.3.1.1 of 3GPP TS 23.401 [10] and section 6.2.2 of 3GPP TS 24.301 [17]. If both IPv4 and IPv6 addresses are assigned by the network to the UE, the UE must prefer the IPv6 address type when the UE discovers the P-CSCF. If only an IPv4 address or only IPv6 address is assigned by the network to the UE then the network must send ESM cause #50 "PDN type IPv4 only allowed" or #51 "PDN type IPv6 only allowed", respectively, to the UE and the UE must not request another PDN connection to the APN utilised in the initial attach or PDN connection establishment for the other IP version, as specified in section 6.2.2 of 3GPP Release 12 TS 24.301 [17].

After the UE has discovered the P-CSCF and registered to IMS with a particular IPv4 or IPv6 address, the UE must use this IP address for all SIP communication for as long as the IMS registration is valid. For all SDP and RTP/RTCP communication, the UE must use the IPv4 address used for SIP communication or an IPv6 address with the IPv6 prefix same as the IPv6 prefix of the IPv6 address used for SIP communication.

Note: There are certain situations where interworking between IP versions is required. These include, for instance, roaming and interconnect between networks using different IP versions. In those cases, the network needs to provide the interworking in a transparent manner to the UE.

5.2 Emergency Service

5.2.1 General

The UE and the network must support emergency services in the IMS domain.

The UE and the network must support the IMS emergency services as specified in 3GPP Release 9 TS 24.229 [15], chapter 6 and Annex H of 3GPP Release 9 TS 23.167 [3], and emergency procedures as specified in 3GPP Release 9 TS 24.301 [17].

When the UE has an ongoing emergency call, the UE must follow the procedures for access domain selection related to “Persistent EPS bearer context” as specified in sections 5.5.3.2.4 and 5.5.3.3.4.3 of 3GPP Release 10 TS 24.301 [17] and section 8.2 of 3GPP Release 10 TS 24.237 [16].

Recognizing that some network operators will continue a parallel CS network whilst their IMS network is deployed, and that support of emergency calls with CS support may be a local regulatory requirement, emergency calls in the CS domain are addressed in Annex A.

The UE and network must support the 3GPP IM CN subsystem XML body as defined in section 7.6 of 3GPP TS 24.229 [15].

The usage of the 3GPP IM CN subsystem XML body in the network is an operator option.

Note: This implies that the P-CSCF must support also the option that the XML body is not used.

The SUPL enabled UE sends the emergency SUPL messages related to the UE detectable emergency session within the PDN connection for emergency bearer services. The SUPL enabled UE sends the emergency SUPL messages related to the non UE detectable emergency session within the PDN connection to the IMS well known APN. The UE selects the bearer to be used based on the TFTs of the bearers of the PDN connection. QCI of the selected bearer is provided by the network.

If the UE:

- receives the Emergency Service Support indication during EPS attach or tracking area updating procedures;
- attempts an emergency registration with IMS;
- receives a SIP 3xx, 4xx (except 401), 5xx or 6xx response to the emergency REGISTER request; and
- is still in a tracking area that has received the Emergency Service Support indication;

then the UE must perform the procedures defined in subclause 5.1.6.8.2 of 3GPP Release 9 TS 24.229.

5.2.2 Interactions between supplementary services and PSAP callback

The network must not invoke the use of supplementary services on a call identified as a PSAP callback as specified in 3GPP Release 12 TS 24.604 [20], 3GPP Release 12 TS 24.605 [21], 3GPP Release 12 TS 24.610 [25] and 3GPP Release 12 TS 24.611 [26].

Note: UE procedures for PSAP callback are not specified by 3GPP.

5.3 Roaming Considerations

This profile has been designed to support IMS voice roaming. For more information on the IMS voice roaming models see GSMA PRD IR.65 [65] and GSMA PRD IR.88 [67].

5.4 Accesses in addition to E-UTRAN

UEs that support cellular (e.g. E-UTRAN) and non-cellular accesses that are not EPC-integrated (e.g. non-EPC integrated Wi-Fi access) must use:

- the cellular access as transport of the Gm reference point; and
- the cellular access as transport of the Ut reference point, unless preconfigured by the home operator.

5.5 Data Off and Services Availability

5.5.1 General

When Data Off is activated, the UE must not send via any PDN connection any uplink IP packet of any service other than Data Off Enabled Services.

Data Off Enabled Services defined in the present document are specified in the following sub-sections.

Note: The UE can disconnect PDN connections that are not required by Data Off Enabled Services.

5.5.2 Supplementary Service Settings Management

The UE must be able to perform supplementary service settings management as described in section 2.3 regardless of whether Data Off is activated.

Unless transport of the uplink IP packets via PDN connection for the APN for Home Operator Services in 3GPP access is already enabled, the UE must temporarily enable transport of the uplink IP packets via PDN connection for the APN for Home Operator Services (see section 4.3.1) for the period required to complete the supplementary service management procedure (e.g. for the period required to complete changing of a Communication Forwarding setting). During that period, the UE must not send via the PDN connection for the APN for Home Operator Services any uplink IP packet of any service other than:

- the supplementary service settings management as described in section 2.3; and
- any other Data Off Enabled Services.

5.5.3 Voice Calls and SMS over IMS

The UE must be able to initiate and receive Voice Calls and SMS over IMS as described in the main body of this document regardless whether Data Off is activated, i.e. the UE continues to use (does not disconnect) the PDN connection via the IMS well-known APN as described in section 4.3.

5.6 Voice Calls and Smart Congestion Mitigation

The UE must and the network can support Smart Congestion Mitigation as specified in section J.2.1.2 of 3GPP Release 12 TS 24.173 [14], section 5.6.1.6 and annex D of 3GPP Release 12 TS 24.301 [17], and sections 5.3.3.2, 5.3.3.10 and 5.3.3.11 of 3GPP Release 12 TS 36.331 [52].

Annex A Complementing IMS with CS

A.1 General

In order to offer its customers a seamless service, the operator may wish to complement the IMS VoIP and SMSoIP capable radio coverage by utilising the CS radio access for voice and/or the CS core network for SMS over SGs. The IMS VoIP and SMSoIP coverage may be less or more extensive than the concurrent Circuit Switched (CS) coverage. This Annex describes the additional features that need to be implemented for the UEs and networks that wish to support such a deployment scenario.

The voice related requirements in this annex are applicable if the UE has the setting of “IMS PS Voice preferred, CS Voice as secondary”.

A.2 Domain Selection

The network and the UE must support the IMS voice over PS session supported indication as specified in section 4.3.5.8 of 3GPP TS 23.401 [10]

The UE must perform voice domain selection for originating sessions with the setting of “IMS PS Voice preferred, CS Voice as secondary” as specified in section 7.2a of 3GPP TS 23.221 [6], and must perform the procedures defined in 3GPP TS 24.301 [10].

Note: The behaviour of UEs with the setting “IMS PS Voice preferred, CS Voice as secondary” is illustrated in Annex A.2 of 3GPP TS 23.221 [6].

The UE must reject an incoming request if the UE is unable to support speech media on current PS access as specified in 3GPP TS 23.237 [8] and 3GPP TS 24.237 [16].

The UE must support Idle Mode Signalling Reduction (ISR) as specified in 3GPP Release 9 TS 23.401 [10] and 3GPP Release 9 TS 24.301 [17].

A.3 SR-VCC

The network must support the Single Radio Voice Call Continuity (SR-VCC) procedures for handover from E-UTRAN as described in 3GPP TS 23.216 [5] and 3GPP TS 23.237 [8].

The UE must support the SR-VCC procedures for single active call only as described in 3GPP TS 23.216 [5], 3GPP TS 24.008 [11], 3GPP TS 24.237 [16], and 3GPP TS 24.301 [17].

Note 1: The mechanisms to perform transfer of additional session / held state / conference call state / alerting calls are out of scope of the present version of this document.

Note 2: UEs using IMS Centralized Services (ICS) capabilities are out of scope of the present version of this document.

A.4 IMS Voice service settings management when using CS access

The UE must use service setting management as defined in section 2.3.2 and section 5.5.1 using the current cellular access, over the APN defined in section 4.3.1. UEs that support non-cellular accesses that are not EPC-integrated (e.g. non-EPC integrated Wi-Fi access), must comply to the requirements of section 5.4.

Note 1: This applies also when the UE is using CS network for voice service.

If:

- the UE attempts to perform supplementary service settings management via XCAP;
- the UE receives an HTTP failure code as described in section 5.3.1.2.2 of 3GPP Release 12 TS 24.623 [28];
- the UE is not configured with network operator's preference for the selection of the domain used by the UE when performing supplementary services setting control for voice services; and
- until the UE performs a power-off/power-on or the UE detects a change of USIM/ISIM

then:

- the UE must not perform supplementary service settings management via XCAP; and
- the UE must instead attempt to perform supplementary service settings management in the CS domain.

Note 2: By default, the UE is not configured with the network operator's preference for the selection of the domain used by the UE when performing supplementary service settings control for voice services.

A.5 Emergency Service

This section modifies the requirements defined in section 5.2 in the following ways:

The UE must, and the network can, support the procedures and capabilities defined in section 5.2.

If the support of one or more of the following scenarios is required, then the network must support the procedures in section 5.2:

- Deployment scenarios where the IMS VoIP capable radio coverage is not complemented by CS radio coverage.
- Provide voice service on LTE to UE with incompatible CS domain.
- Provide voice service on LTE to UE supporting LTE only

When emergency service support via CS domain is required, the UE and the network must support the CS emergency service as used today.

The UE must be able to perform domain selection for emergency calls, and automatically be able to retry in the CS domain if a SIP INVITE for an IMS emergency session is rejected with a SIP 3xx, 4xx (except 407), 5xx or 6xx response, as defined in section 7.3 and Annex H of 3GPP Release 9 TS 23.167 [3] and 3GPP Release 9 TS 24.229 [15]. The UE must be able to detect if the network is not supporting IMS emergency sessions as defined in 3GPP TS 23.401 [10], then select the CS domain for UE detected emergency sessions. The UE must be able to perform domain selection for emergency calls, and also automatically be able to retry in the IMS if a UE detected CS emergency call attempt fails and the network supports

IMS emergency sessions, as defined in subclause 7.3 and Annex H of 3GPP Release 9 TS 23.167 [3], 3GPP Release 9 TS 23.401 [10] and 3GPP Release 9 TS 24.229 [15].

The network must be able to reject a SIP INVITE for an IMS emergency session such that the UE can retry in the CS domain, as defined in 3GPP TS 24.229 [15] and section 6.2.1 of 3GPP TS 23.167 [3].

When IMS emergency service is not possible (e.g. the network does not support IMS emergency), and when the UE supporting CS Fallback (CSFB), as described in 3GPP TS 23.272 [9], is IMSI attached, then the UE must use the CSFB procedures for CS emergency service. If the network or the UE do not support CSFB, the UE must autonomously select the RAT that supports CS.

The UE must support SR-VCC for IMS emergency sessions as specified in 3GPP Release 9 TS 23.216 [5] and 3GPP TS 23.237 [8]. The SR-VCC UE that supports IMS emergency service must support the SIP instance ID as defined in section 7.2 3GPP TS 24.237 [16].

The network must support SR-VCC for IMS emergency sessions as specified in 3GPP Release 9 TS 23.216 [5] and 3GPP TS 23.237 [8]. The network must support the SIP instance ID as defined in 3GPP TS 24.237 [16].

In limited service state, it is recommended that a UE that is CS voice capable should always camp on a RAT that is likely to support the CS domain, e.g. GERAN or UTRAN or CDMA2000, as described in 3GPP TS 23.221 [6].

A.6 Roaming Considerations

When voice over IMS, see section 5.3, is not possible then the UE must follow the procedures defined in Annex A.2 to use CS for voice service.

A.7 SMS Support

This section modifies the requirements defined in section 2.5 in the following ways:

The UE must, and the network can, support SMS-over-IP as described in section 2.5. In addition, when support of SMS over NAS signalling is required, the UE and the network must support the necessary procedures as specified in 3GPP TS 23.272 [9], 3GPP TS 23.221 [6] and 3GPP TS 24.301 [17]. If the UE supports both SMS-over-IP and SMS over NAS signalling methods, the UE must support the functionality as specified according to section 7.2c of 3GPP TS 23.221 [6]. The UE must:

a) be pre-configured by the operator to use either SMS over IP or SMS over NAS signalling; and
b) be capable of being configured according to the parameter “SMS_Over_IP_Networks_Indication” in the IMS Management Object defined in 3GPP TS 24.167 [68], in order to give operator control to configure the UE to use SMS over NAS signalling when required.

If SMS over NAS signalling is not supported then the network must support the procedures in section 2.5.

A.8 Call Waiting in the CS domain

When the UE on an ongoing call is presented with a new incoming call from the CS core network and:

- if the Communication Waiting supplementary service (see section 2.3.4) is deactivated in the UE, then the UE must reject the incoming call by sending a RELEASE COMPLETE message with cause #17 "user busy" as specified in 3GPP TS 24.008 [11]; and
- if the Communication Waiting supplementary service is activated in the UE, the UE must generate a call waiting indication to the user.

A.9 USSD

For a UE that supports CSFB:

- If the UE has no ongoing IMS session for conversational voice calls and is not in the process of establishing an IMS session for conversational voice call (originated or terminated) then the UE must use the CSFB procedures as described in 3GPP TS 23.272 [9] for originating and terminating USSD requests in the CS domain.
- If the UE has one or more ongoing IMS sessions for conversational voice call or is in the process of establishing an IMS session for conversational voice call (originated or terminated) then:
    - the UE must not attempt originating USSD requests in the CS domain; and
    - if paging procedure takes place, the UE must reject the request to perform CS fallback for terminating USSD requests as specified in 3GPP TS 24.301 [17].

Annex B Features needed in certain regions

B.1 General

This Annex describes features that operators need to support in certain regions due to local regulatory requirements.

B.2 Global Text Telephony

In some regions, there are regulatory requirements that deaf/hearing impaired people must be able to perform text based communication to other users and government offices (e.g. to provide equal access to emergency services to all users). In this document, this service is referred to as Global Text Telephony and the following requirements outline how the Global Text Telephony service should be implemented in regions where required.

The UE must include the text media feature tag, as defined in IETF RFC 3840 [86], in the Contact header field of the SIP REGISTER request, in the Contact header field of the SIP INVITE request, and in the Contact header field of the SIP response to the SIP INVITE request, using procedures defined in 3GPP TS 24.229 [15].

Global Text Telephony/teletypewriter messages must use ITU-T Recommendation T.140 [70] real-time text according to the rules and procedures specified in 3GPP TS 26.114 [35] with the following clarifications:

- The UE must offer AVP for all media streams containing real-time text.
- For real-time text, RTCP reporting must be turned off by setting the SDP bandwidth modifiers “RS” and “RR” to zero.
- The sampling time used must be 300 ms.
- Change of the sampling time (rate adaptation) is not required.

B.3 Service Specific Access Control

In some regions, e.g. Japan, there are regulatory requirements that require the need to release only voice calls while allowing high priority calls and access for other packet service (e.g. email, web, disaster message board), as under disaster or emergency events, the mass simultaneous voice call requests are usually the main cause for network congestion.

To fulfil the regulatory requirements, the UE for such regions must support Service Specific Access Control (SSAC) as specified in 3GPP Release 9 TS 22.011 [1], 3GPP Release 9 TS 36.331 [52], 3GPP Release 9 TS 24.173 [14] and 3GPP Release 9 TS 27.007 [43].

Annex C MNO provisioning

C.1 General

This annex describes the capabilities to support MNO provisioning for the UE (e.g. for open market devices). An open market device:

- Supports non-roaming and roaming cases;
- Has a default configuration suitable for many MNOs; and
- Can be configured to the MNO's needs.

C.2 Remote Client Configuration

The UE and the network must support one of the two configuration methods in order to support MNO provisioning for the parameters that are defined in 3GPP (see also Table C.3.1):

- OMA DM V1.2 with http binding as specified in OMA-ERELD-DM-V1_2 [X2]; or
- Service provider device configuration as specified in GSMA PRD RCC.14 [X3].

Note 1: The requirement on which configuration method to support may differ between regions.

Note 2: GSMA PRD RCC.14 v3.0 [X3] supports only the Managed Object (MO) specified in 3GPP TS 24.167 [68].

C.3 Configuration Parameters

The following configuration parameters with their default values must be supported by the UE and the network. The UE must use the default value for each parameter unless configured differently by any of the methods as described in section C.2.

Parameter	Default value	Defined in	See also clause
IMS	0-Enabled	3GPP TS 24.305 [18]	2.2.1
Voice and/or Video over LTE allowed	Voice only allowed	not defined in 3GPP	2.2.1
Voice and/or Video over LTE allowed while roaming	Voice and video Prohibited	not defined in 3GPP	2.2.1, 5.3
When to use SMSoIP	SMSoIP irrespective of IMS voice support	not defined in 3GPP	2.2.1
RegRetryBaseTime	30 Sec	3GPP TS 24.167 [68]	2.2.1
RegRetryMaxTime	300 sec	3GPP TS 24.167 [68]	2.2.1
Local number when roaming	Geo-local	not defined in 3GPP	2.2.3
Timer_T1	2 sec	3GPP TS 24.167 [68]	2.2.4
Timer_T2	16 sec	3GPP TS 24.167 [68]	2.2.4
Timer_T4	17 sec	3GPP TS 24.167 [68]	2.2.4
XCAP Root URI	No default	3GPP TS 24.623 [X]	2.3.2
Conference Factory URI	None	3GPP TS 24.166 [Z]	2.3.3

Parameter	Default value	Defined in	See also clause
Use contents of FROM header if P-A-I is not present for incoming voice and video calls	Prohibit	not defined in 3GPP	2.3.6
SIP Preconditions used	Yes	not defined in 3GPP	2.4.1
Media on default (QCI=5) bearer	Prohibit	not defined in 3GPP	2.4.3
Network Identifier part of the HOS APN	Internet APN	not defined in 3GPP	4.3.1
APN in initial attach	No APN	not defined in 3GPP	4.3.1
SS_domain_setting	No default	3GPP TS 24.167 [68]	A.4
SMS_Over_IP_Networks_Indication	1 – SMS service is preferred to be invoked over the IP networks	3GPP TS 24.167 [68]	A.7

Table C.3.1 Configuration parameters and their default values

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Document History

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