IMS Profile for Voice and SMS

Version 6.0

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*This is a **Non Binding** permanent reference document of the GSM Association.*

Security Classification – NON CONFIDENTIAL GSMA Material

Table of Contents

1 General Introduction

1.1 The IMS Profile for Voice and SMS

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 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].

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

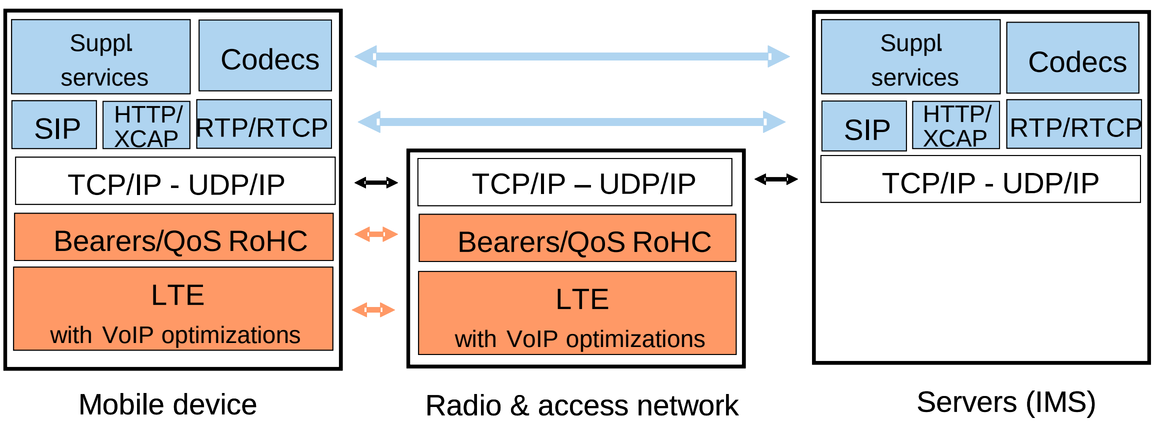

Figure 1.1: Depiction of UE and Network Protocol Stacks in IMS Profile for Voice

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

Note: Although, this version of the specification focuses on E-UTRAN, the defined IMS functionalities may be applied to other IP Connectivity Accesses.

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 Terms

Term	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
CB	Communication Barring
CDIV	Communication Diversion
CDIVN	CDIV Notification
CFNL	Communication Forwarding on Not Logged-in
CFNRc	Communication Forwarding on Not Reachable
CN	Core Network
CS	Circuit Switched
CSFB	CS Fallback
CW	Communication Waiting
DRB	Data Radio Bearer
DRX	Discontinuous Reception
DTX	Discontinuous Transmission

Term	Description
eNB	eNodeB
EPS	Evolved Packet System
E-UTRAN	Evolved Universal Terrestrial Radio Access Network
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
MMTel	Multimedia Telephony
MO	Managed Object
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
PS	Packet Switched
QCI	Quality of Service Class Indicator
RAT	Radio Access Technology
RLC	Radio Link Control
RoHC	Robust Header Compression
RTCP	RTP Control Protocol
RTP	Real Time Protocol
SCC AS	Service Centralization and Continuity Application Server
SDP	Session Description Protocol
SigComp	Signalling Compression
SIP	Session Initiation Protocol
SMSoIP	SMS over IP
SRB	Signalling Radio Bearer
SR-VCC	Single Radio Voice Call Continuity
TAS	Telephony Application Server
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
VoIP	Voice Over IP
XCAP	XML Configuration Access Protocol

Term	Description
XML	eXtensible Markup Language

1.5 Document Cross-References

Document	Name
3GPP TS 22.011	Service Accessibility
3GPP TS 23.003	Numbering, addressing and identification
3GPP TS 23.167	IP Multimedia Subsystem (IMS) emergency sessions
3GPP TS 23.203	Policy and charging control architecture
3GPP TS 23.216	Single Radio Voice Call Continuity (SRVCC); Stage 2
3GPP TS 23.221	Architectural requirements
3GPP TS 23.228	IP Multimedia Subsystem (IMS); Stage 2
3GPP TS 23.237	IP Multimedia Subsystem (IMS) Service Continuity; Stage 2
3GPP TS 23.272	Circuit Switched (CS) fallback in Evolved Packet System (EPS); Stage 2
3GPP TS 23.401	General Packet Radio Service (GPRS) enhancements for Evolved Universal Terrestrial Radio Access Network (E-UTRAN) access
3GPP TS 24.008	Mobile radio interface layer 3 specification; Core Network protocols; Stage 3
3GPP TS 24.147	Conferencing using the IP Multimedia (IM) Core Network (CN) subsystem; Stage 3
3GPP TS 24.173	IMS Multimedia telephony service and supplementary services; Stage 3
3GPP TS 24.229	IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3
3GPP TS 24.237	IP Multimedia Subsystem (IMS) Service Continuity; Stage 3
3GPP TS 24.301	Non-Access-Stratum (NAS) protocol for Evolved Packet System (EPS); Stage 3
3GPP TS 24.305	Selective Disabling of 3GPP User Equipment Capabilities (SDoUE) Management Object (MO);
3GPP TS 24.341	Support of SMS over IP networks; Stage 3
3GPP TS 24.604	Communication Diversion (CDIV) using IP Multimedia (IM)Core Network (CN) subsystem; Protocol specification
3GPP TS 24.605	Conference (CONF) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
3GPP TS 24.606	Message Waiting Indication (MWI)using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
3GPP TS 24.607	Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
3GPP TS 24.608	Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
3GPP TS 24.610	Communication HOLD (HOLD) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
3GPP TS 24.611	Anonymous Communication Rejection (ACR) and Communication Barring (CB) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
3GPP TS 24.615	Communication Waiting (CW) using IP Multimedia (IM) Core

Document	Name
	Network (CN) subsystem; Protocol Specification
3GPP TS 24.623	Extensible Markup Language (XML) Configuration Access Protocol (XCAP) over the Ut interface for Manipulating Simulation Services
3GPP TS 26.071	Mandatory speech CODEC speech processing functions; AMR speech Codec; General description
3GPP TS 26.073	ANSI C code for the Adaptive Multi Rate (AMR) speech codec
3GPP TS 26.090	Mandatory Speech Codec speech processing functions; Adaptive Multi-Rate (AMR) speech codec; Transcoding functions
3GPP TS 26.093	Mandatory speech codec speech processing functions Adaptive Multi-Rate (AMR) speech codec; Source controlled rate operation
3GPP TS 26.103	Speech codec list for GSM and UMTS
3GPP TS 26.104	ANSI-C code for the floating-point Adaptive Multi-Rate (AMR) speech codec
3GPP TS 26.114	IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction
3GPP TS 26.131	Terminal acoustic characteristics for telephony; Requirements
3GPP TS 26.132	Speech and video telephony terminal acoustic test specification
3GPP TS 26.171	Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; General description
3GPP TS 26.173	ANSI-C code for the Adaptive Multi-Rate - Wideband (AMR-WB) speech codec
3GPP TS 26.190	Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Transcoding functions
3GPP TS 26.193	Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Source controlled rate operation
3GPP TS 26.204	Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; ANSI-C code
3GPP TS 27.007	AT command set for User Equipment (UE)
3GPP TS 31.103	Characteristics of the IP Multimedia Services Identity Module (ISIM) application
3GPP TS 33.203	3G security; Access security for IP-based services
3GPP TS 36.101	Evolved Universal Terrestrial Radio Access (E-UTRA); User Equipment (UE) radio transmission and reception
3GPP TS 36.104	Evolved Universal Terrestrial Radio Access (E-UTRA); Base Station (BS) radio transmission and reception
3GPP TS 36.300	Evolved Universal Terrestrial Radio Access (E-UTRA) and Evolved Universal Terrestrial Radio Access Network (E-UTRAN); Overall description; Stage 2
3GPP TS 36.321	Evolved Universal Terrestrial Radio Access (E-UTRA); Medium Access Control (MAC) protocol specification
3GPP TS 36.323	Evolved Universal Terrestrial Radio Access (E-UTRA); Packet Data Convergence Protocol (PDCP) specification
3GPP TS 36.331	Evolved Universal Terrestrial Radio Access (E-UTRA);Radio Resource Control (RRC); Protocol specification
RFC 768	User Datagram Protocol
RFC 3095	RObust Header Compression (ROHC): Framework and four profiles: RTP, UDP, ESP, and uncompressed
RFC 3261	SIP: Session Initiation Protocol
RFC 3550	RTP: A Transport Protocol for Real-Time Applications
RFC 3551	RTP Profile for Audio and Video Conferences with Minimal Control
RFC 3556	Session Description Protocol (SDP) Bandwidth Modifiers for RTP Control Protocol (RTCP) Bandwidth

Document	Name
RFC 3680	A Session Initiation Protocol (SIP) Event Package for Registrations
RFC 3842	A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
RFC 4575	A Session Initiation Protocol (SIP) Event Package for Conference State
RFC 4815	RObust Header Compression (ROHC): Corrections and Clarifications to RFC 3095
RFC 4867	RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs
draft-ietf-mmusic-sdp-capability-negotiation-10	SDP Capability Negotiation
GSMA PRD IR.65	IMS Roaming and Interworking Guidelines
GSMA PRD IR.67	DNS/ENUM Guidelines for Service Providers and GRX/IPX Providers
GSMA PRD IR.88	LTE Roaming Guidelines

2 IMS Feature Set

2.1 General

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

2.2 Support of generic IMS functions

2.2.1 SIP Registration Procedures

UE and IMS core network must follow the Session Initiated Protocol (SIP) registration procedures defined in 3GPP TS 24.229. 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 evaluate to true, then the UE must register with the IMS. Selective Disabling of 3GPP User Equipment Capabilities as defined in TS 24.305 is not mandated in this profile, therefore in the case where TS 24.305 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, using procedures as defined in section 5.1.1.2.1 of 3GPP TS 24.229. If the UE supports SMS over IP (see section 2.5 and A.7), it must include feature tag used to indicate SMS over IP service, that being +g.3gpp.smsip as defined in section 5.3.2.2 of 3GPP TS 24.341.

UE and IMS core network must support network-initiated de-registration as defined in 3GPP TS 24.229.

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

The UE must include an IMEI URN (see 3GPP TS23.003 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. 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 section 4.2.4); or
b) The temporary public user identity derived from the IMSI (3GPP TS 24.229).

No other implicit registration set must be used for VoLTE.

Note 2: According to 3GPP TS 23.228, 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.

2.2.2 Authentication

UE and IMS core network must follow the procedures defined in 3GPP TS 24.229 and 3GPP TS 33.203 for authentication with IMS Authentication and Key Agreement (IMS-AKA), Sec-Agree and IPSec. Support of integrity protection is required for both UE and network. Confidentiality protection is optional, 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.

UE and IMS core network must the support the procedures for USIM based authentication in case there is no ISIM present on the Universal Integrated Circuit Card (UICC) as defined in 3GPP TS 23.228, Annex E.3.1 and 3GPP TS 24.229, Annex C.2. This includes support for the P-Associated Uniform Resource Identifier (URI) header to handle barred IP Multimedia Public Identities (IMPUs).

UE and IMS core network must support the procedures for authentication at the Ut reference point as specified in 3GPP TS 24.623.

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

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

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

2.2.3 Addressing

UEs and IMS core network using this profile must support SIP URIs (alphanumeric) and Mobile Subscriber ISDN Number (MSISDN) based IMPU, which means a tel-URIs with an associated SIP-URI, for example

- Alphanumeric SIP-URI
  - SIP: voicemail@example.com

- MSISDN based IMPU
  - tel: :+491721234512
  - SIP:+491721234512@example.com; user=phone

The UE and 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. 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. 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 have the option to support geo-local numbers. If the UE supports geo-local numbers, it must set the “phone-context” parameter as with home local numbers, but prefixed by the “geo-local.” string, according to the Alternative 8 in Section 7.2A.10.3 in 3GPP TS 24.229.

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.

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

2.2.4 Call Establishment and Termination

UE and IMS core network must follow 3GPP TS 24.229 for establishment and termination of a call.
UE and IMS core network must support reliable provisional responses.

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. 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.

The usage of preconditions is discussed in Section 2.4.

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, section 4.2.7.3 of 3GPP TS 23.228 and 3GPP TS 24.229. Furthermore, the UE should be able to maintain at least seven (7) parallel early dialogues until receiving the final response on one of them and the UE must support receiving media on one of these early dialogues.

2.2.6 Tracing of Signalling

The UE and the network are not required to support Tracing of Signalling, as described in 3GPP TS 24.229 section 4.8, and are not required to subscribe to the debug event package as described in sections 5.1.1.3A and 5.2.3A.

2.2.7 The use of Signalling Compression

The UE must not use SIGCOMP when the initial IMS registration is performed in E-UTRAN access.

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."

2.3 Supplementary Services

2.3.1 Supplementary Services Overview

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

UE and Telephony Application Server (TAS) must support the supplementary services listed in Table 2.1. It is up to the operator to enable these services.

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

Table 2.1 Supplementary services

Supplementary Service
Originating Identification Presentation 3GPP TS 24.607
Terminating Identification Presentation 3GPP TS 24.608
Originating Identification Restriction 3GPP TS 24.607 (Note 1)
Terminating Identification Restriction 3GPP TS 24.608 (Note 1)
Communication Forwarding Unconditional 3GPP TS 24.604 (Note 1)
Communication Forwarding on not Logged in 3GPP TS 24.604 (Note 1)
Communication Forwarding on Busy 3GPP TS 24.604 (Note 1)
Communication Forwarding on not Reachable 3GPP TS 24.604 (Note 1)
Communication Forwarding on No Reply 3GPP TS 24.604 (Note 1)
Barring of All Incoming Calls 3GPP TS 24.611 (Note 1)
Barring of All Outgoing Calls 3GPP TS 24.611 (Note 1)
Barring of Outgoing International Calls 3GPP TS 24.611 (Note 2)
Barring of Outgoing International Calls – ex Home Country 3GPP TS 24.611 (Note 2)
Barring of Incoming Calls - When Roaming 3GPP TS 24.611 (Note 1)
Communication Hold 3GPP TS 24.610
Message Waiting Indication 3GPP TS 24.606 (Note 1)
Communication Waiting 3GPP TS 24.615 (Note 1)
Ad-Hoc Multi Party Conference 3GPP TS 24.605 (Note 1)

Note 1: Recommended options are described in sections 2.3.3 – 2.3.9.

Note 2: 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.

The home operator can configure the UE with an XCAP Root URI as specified in 3GPP TS 24.623. If the UE has not been configured with an XCAP Root URI, then the UE must construct an XCAP root URI as defined in 3GPP TS 23.003.

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

Note: When not registered with IMS, a UE can use the default public user identity received during the last successful registration as in clause 2.2.1 in this document.

2.3.3 Ad-Hoc Multi Party Conference

The UE and IMS core network must support the procedures defined in 3GPP TS 24.605, with the clarifications defined in this sub section.

Note: As per Section 4.2 of 3GPP TS 24.605, the invocation and operation for conferencing is described in 3GPP TS 24.147.

For conference creation, the UE and IMS core network must support Three Way Session creation as described in Section 5.3.1.3.3 of 3GPP TS 24.147.

For inviting other users to the conference, the UE and IMS core network must support the procedure described in Section 5.3.1.5.3 of 3GPP TS 24.147. The UE must send the 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 REFER request, as described in Section 5.3.1.5.3 of 3GPP TS 24.147.

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

The UE can and the IMS core network must support the procedures in 3GPP TS 24.605 for subscription to conference state events. The IMS core network can support all, or a subset of the elements and attributes in IETF RFC 4575. 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 IMS core network must support audio media for the conference session.

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

Floor control for conferencing as described in section 8 in 3GPP TS 24.147 is not required. Consent procedures for list server distribution as described in 5.3.1.7 in 3GPP TS 24.147 are not required.

2.3.4 Communication Waiting

UE and IMS core network must support the terminal based service, as described in 3GPP TS 24.615. Network-based service is not required. Communication Waiting (CW) indication as defined in Section 4.4.1 of 3GPP TS 24.615 is not required. The UE is required to support Alert-Info, with values as specified in 3GPP TS 24.615. Service activation, deactivation, and interrogation are not required.

2.3.5 Message Waiting Indication

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

2.3.6 Originating Identification Restriction

UE and IMS core network must support the SIP procedures in 3GPP TS 24.607. Service configuration as described in Section 4.10 of 3GPP TS 24.607 is optional.

2.3.7 Terminating Identification Restriction

UE and IMS core network must support the SIP procedures in 3GPP TS 24.608. Service configuration, as described in section 4.9 of 3GPP TS 24.608, is optional.

2.3.8 Communication Diversion

UE and IMS core network must support the SIP procedures in 3GPP TS 24.604 for Communication Diversion (CDIV). The CDIV Notification (CDIVN) service is not required. For CDIV service activation, deactivation, and interrogation (XCAP operations), the UE and IMS core network must support the XML rules for Call Forwarding Unconditional and the conditions and actions listed in Table 2.2. UE and IMS core network shall support the XML rules as described in 3GPP TS 24.604 section 4.9.1. It is recommended that a UE should support the History-Info header for presentation of diverting parties.

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

Table 2.2 Supported conditions and actions in CDIV

Type	Parameter
Condition	busy
Condition	media (supported media types: audio, audio AND video)
Condition	no-answer
Condition	not-registered
Condition	not-reachable (Note)

Action	target
Action	NoReplyTimer

Note: The GSM 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.

2.3.9 Communication Barring

UE and IMS core network must support the SIP procedures in 3GPP TS 24.611. For service activation, deactivation, and interrogation (XCAP operations), the UE and 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. UE and IMS core network shall support the XML rules as described in 3GPP TS 24.611 section 4.9.1.3.

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

Table 2.3 Supported conditions in CB

roaming
international
international-exHC

2.4 Call Set-up Considerations

2.4.1 SIP Precondition Considerations

The UE must support the SIP preconditions framework, as specified in 3GPP TS 24.229. Operators can 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.

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 (for example, when the P-CSCF receives an abort session request from the Policy and Charging Rules Function (PCRF)).

If the PDN connectivity is lost, then the UE must re-establish the PDN connection. This will trigger the network to initiate a new SIP signalling bearer in conjunction with the PDN connection establishment.

When the UE regains PDN connectivity, the UE must perform a new initial registration to IMS, in case the IP address changed, or the IMS registration expired during the absence of IP connectivity.

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 in TS 24.229 (P-CSCF must be informed about loss of bearer by the PCRF).

Note 1: The loss of GBR bearer may be due to loss of radio connection indicated by a 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 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 to continue 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, 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 in 3GPP TS 24.229. The UE can act differently per media type.

Note 3: In the case where voice bearer is lost or fails to get established, the network will, in normal cases, release the session as described in the beginning of the 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. In the case of 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, or should be modified, or should be released.

If the UE, having lost radio connectivity, then regains radio connectivity, the UE must perform a new initial registration to IMS in case the IMS registration expired during the absence of radio connectivity.

2.4.3 Voice Media Considerations

The Session Description Protocol (SDP) offer/answer for voice media must be formatted as specified in Section 6.2.2 of 3GPP TS 26.114, with the restrictions included in the present document.

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 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 offer and allow the use of whatever media streams it supports. The UE must, in the SDP answer, set the port number to zero for the media streams it does not support.

Note 2: This means that a voice-only 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 the procedures in Sections 5.3.1 and 5.3.2 in 3GPP TS 24.341.

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 in 3GPP TS 24.341 must be supported.

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, and the functions (answering of routing information query, and transport layer interworking) according to the procedures in Sections 5.3.3.3 and 5.3.3.4 in 3GPP TS 24.341.

3 IMS Media

3.1 General

This section endorses a set of media capabilities specified in 3GPP TS 26.114. 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 MRFP and the 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, TS 26.090, TS 26.073, and TS 26.104, including all eight (8) modes and source rate controlled operations, as described in 3GPP TS 26.093. The UE must be capable of operating with any subset of these eight (8) codec modes.

If wideband speech communication is offered, the UE must support AMR wideband codec as described in 3GPP TS 26.114, 3GPP TS 26.171, 3GPP TS 26.190, 3GPP TS 26.173 and 3GPP TS 26.204, including all nine (9) modes and source controlled rate operation 3GPP TS 26.193. The UE shall be capable of operating with any subset of these nine (9) codec modes.

When transmitting, 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, for example as described for UMTS_AMR_2 in 3GPP TS 26.103 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, 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) shall 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 shall support:

- AMR speech codec modes 12.2, 7.4, 5.9 and 4.75 as described in 3GPP TS 26.071, TS 26.090, TS 26.073, and TS 26.104.

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

- AMR-WB speech codec modes 12.65, 8.85 and 6.60 as described in 3GPP TS 26.171, TS 26.190, TS 26.173, and TS 26.204.

Entities in the IMS network that provide transcoding-free interworking to the CS network shall 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 only required for transcoder-free inter-working 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 must be used.

3.2.2.2 SDP Offer Considerations

The SDPCapNeg framework [draft-ietf-mmusic-sdp-capability-negotiation-10 (May 2009): "SDP Capability Negotiation"] must not be used in the SDP offer 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 which uses SDPCapNeg. The answer must indicate the use of the RTP AVP profile.

Note: In 3GPP TS 26.114 section 6.2.1a, 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.

3.2.3 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 and IETF RFC 768, respectively, to transport voice.

The UE must use the same port number for sending and receiving RTP packets. This facilitates interworking with fixed/broadband access. However, the UE and the entities in the IMS core network that terminate the user plane must accept RTP packets that are not received from the same remote port where RTP packets are sent.

3.2.4 RTCP Usage

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

The UE and the entities in the IMS core network that terminates the user plane must use the same port number for sending and receiving RTCP packets. This facilitates interworking with fixed/broadband access. However, the UE and the entities in the IMS core network that terminates the user plane must accept RTCP packets that are not received from the same remote port where RTCP packets are sent.

The bandwidth for RTCP traffic must be described using the "RS" and "RR" SDP bandwidth modifiers at media level, as specified by RFC 3556. Therefore, a UE and the entities in the IMS core network that terminate the user plane must include the "b=RS:" and "b=RR:" fields in SDP, and must be able to interpret them.

In active "speech-only sessions," the RTCP transmission must be turned off by the UE and the entities in the IMS core network that terminates the user plane, by setting the "RS" and "RR" SDP bandwidth modifiers to zero. When media is put on hold, the transmission of

RTCP must be temporarily enabled by (re-)negotiating the RTCP bandwidth with "RS" and "RR" SDP bandwidth modifiers greater than zero.

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. In context of the Voice over IMS profile, the primary function of RTCP is to provide link aliveness information while the media are on hold.

Once RTCP is enabled, the UE and the entities in the IMS core network that terminates the user plane must set the sending frequency to a value equal or less than 5 seconds. The recommended value is 5 seconds.

Note 2: The minimum sending frequency is calculated from the values of "RS" and "RR" SDP bandwidth modifiers according to rules and procedures in IETF RFC 3550.

The UE and the entities in the IMS core network that terminates the user plane must support the transmission of RTCP packets formatted according to the rules in IETF RFC 3550 and with the clarifications below:

RTCP compound packet format must be used. 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, a Receiver Report (RR) should be sent. Some implementations may send a Sender Report (SR) instead of a Receiver Report (RR), and this must be handled and accepted as valid.

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

Sender report (SR) RTCP packet

- Version 2 must be used.
- Padding must not be used (and therefore padding bit must not be set).
- It is recommended that NTP timestamp be set to zero (0) (usage not required).
- It is recommended that RTP timestamp be the same as in the previously sent RTP packet (usage not required).
- It is recommended that Last SR timestamp (LSR) be set to zero (0) (usage not required).

Receiver report (RR) RTCP packet

- Version and padding as described for SR packet.
- It is recommended that Last SR timestamp (LSR) be set to zero (0) (usage not required).

Source description (SDES) RTCP packet

- Version and Padding as described for SR packet.
- The SDES item CNAME must be included in one packet.
- It is recommended that only one SDES item be used (CNAME, usage of other SDES items not required).

Note 3: 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 terminates the user plane must be able to receive all types of RTCP packets, according to the rules specified in IETF RFC 3550.

Note 4: For link aliveness monitoring, the compound RTCP packet (SR + SDES) must be used, as described above. For other RTCP packets, the UE and the entities in the IMS core network that terminates the user plane which use this Voice over IMS profile, it is not required to use the information in the received RTCP packets.

3.2.5 AMR Payload Format Considerations

The Adaptive Multi Rate (AMR) and when applicable Adaptive Multi-Rate Wideband (AMR-WB) payload format(s) specified in IETF RFC 4867 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 format. The UE and the entities in the IMS core network that terminates the user plane must request the use of bandwidth-efficient 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.

IMS media gateway 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, for example, 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 terminates the user plane which use this profile are capable to detect and drop the duplicated frames.

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 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. The codec modes and source control rate operation (DTX) settings must be as specified in 3GPP TS 26.132.

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.

4 Radio and Packet Core Feature Set

4.1 Robust Header Compression

UE and network must support Robust Header Compression (RoHC) as specified in 3GPP TS 36.323, IETF RFC 3095 and IETF RFC 4815. 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, 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 for Voice over IMS profile (see Annex B in 3GPP TS 36.331):

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 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, LTE Discontinuous Reception (DRX) method as specified in 3GPP TS 36.300 and TS 36.321 must be deployed. Support of DRX is mandatory for both UE and network.

4.2.3 RLC configurations

Radio Link Control (RLC) entity must be configured to perform data transfer in the following modes as specified in TS 36.322:

- 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 guaranteed bit rate (GBR) bearer, although it is a very low data rate compared to LTE peak rates, as described in 3GPP TS 23.401. The GBR bearer for voice requests dedicated network resources related to the Guaranteed Bit Rate (GBR) for AMR codec values. The network resources associated with the EPS bearer supporting GBR must be permanently allocated by admission control function in the eNodeB at bearer establishment. Reports from 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. 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 must use the IMS well known APN as defined in PRD IR.88; any other application must not use this APN.

The UE must not provide the IMS well known APN in the Evolved Universal Terrestrial Radio Access Network (E-UTRAN) initial attach.

If the PDN connection established during the initial attach is to an APN other than the IMS well known APN, then prior to registering with IMS the UE must establish another PDN connection to the IMS well known APN.

Note: 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.

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.

For XCAP requests, the UE must be preconfigured or provisioned by the home operator with the APN to be used for XCAP requests.

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. Since the minimum requirement for the UE is the support of one (1) UM bearer which is used for voice (see Section 7.3.1 and Annex B in 3GPP TS 36.331), 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 (for example 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.

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 3GPP TS 24.229, Annex L.2.2.1 as option II for P-CSCF discovery.

5 Common Functionalities

5.1 IP Version

The UE and the network must support both IPv4 and IPv6 for all protocols that are used for the VoIP application: SIP, SDP, RTP, RTCP and XCAP/HTTP. At PS attach, the UE must request the PDN type: IPv4v6, as specified in Section 5.3.1.1 in 3GPP TS 23.401. If both IPv4 and IPv6 addresses are assigned for the UE, the UE must prefer to IPv6 address type when the UE discovers the P-CSCF.

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

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

UEs and network deployments must support emergency services in the IMS domain.

The UE and the network must support the Release 9 IMS emergency services as specified in TS 24.229, TS 23.167, chapter 6 and Annex H, and Release 9 emergency procedures as specified in TS 24.301.

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. UEs and networks compliant with this profile must implement support for the 3GPP IM CN subsystem XML body as defined in section 7.6 of 3GPP TS 24.229.

Note 1: This body is used to re-direct emergency calls to the CS domain.

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

Note 2: 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 emergency PDN connection. 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.

5.3 Roaming Considerations

This profile has been designed to support IMS roaming with both P-CSCF and PGW in the visited network. For more information on this roaming model see GSMA PRD IR.65 and GSMA PRD IR.88. Other roaming models are out of the scope of this profile.

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 TS 23.401 (section 4.3.5.8) in 3GPP Release 8.

An UE must perform voice domain selection for originating sessions with the setting of “IMS PS Voice preferred, CS Voice as secondary” as specified in TS 23.221, Section 7.2a and must perform the procedures in TS 24.301.

Note 1: The behaviour of UEs with the setting “IMS PS Voice preferred, CS Voice as secondary” is illustrated in TS 23.221, Annex A.2.

An UE must reject the incoming request if the UE is unable to support speech media on current PS access as specified in 3GPP TS 23.237 and 3GPP TS 24.237.

An UE must support Idle Mode Signalling Reduction (ISR) as specified in Release 9 specifications TS 23.401 and TS 24.301.

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 TS 23.216 and TS 23.237. The UE detects that the network support SR-VCC from the reply from the MME on the Attach request message (TS 23.216 section 6.2.1).

The UE must support the SR-VCC procedures for single active call only as described in TS 23.216, TS 24.008, TS 24.237 section 9.2.1, and TS 24.301.

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 profile.

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

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.

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

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 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 other domain if an emergency session attempt fails, as defined in TS 23.167 chapter 7.3 and TS 24.229. The UE must be able to detect if the network is not supporting IMS emergency sessions as defined in TS 23.401, then select the CS domain for UE detected emergency sessions.

The network must be able to reject an IMS emergency session attempt such that the UE can retry in the CS domain, as defined in 3GPP TS 24.229 and 3GPP TS 23.167, chapter 6.2.1.

When IMS emergency service is not possible (for example, the network does not support IMS emergency), and when the UE supporting CS Fallback (CSFB), as described in 3GPP TS 23.272, is IMSI attached, and emergency services are supported in the CS domain, the UE must use the CSFB procedures for CS emergency service. If the network or the UE does not support CSFB, the UE must autonomously select the RAT which supports CS emergency service.

The UE must support SR-VCC for IMS emergency sessions as specified in 3GPP Release 9 TS 23.216 and TS 23.237. The SR-VCC UE which supports IMS emergency service must support SIP instance ID as per the procedures in 3GPP TS 24.237 section 7.2.

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

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

A.6 Roaming Considerations

Section 5.3 defines the preferred roaming model, but this model may not always be possible, due to the IMS roaming restrictions or lack of P-CSCF in the visited network. When voice over IMS is not possible the UE must follow procedures defined in Annex section 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 network can, support the SMS-over-IP as described in section 2.5. In addition, when support of SMS over SGs is required, the UE and network must support the necessary procedures as specified in 3GPP TS 23.272, TS 23.221 and TS 24.301. If the UE supports both SMS-over-IP and SMS over SGs methods, the UE must support functionality as specified according to section 7.2c in 3GPP TS 23.221. The UE must:

a) be pre-configured by the operator to use either SMS over IP or SMS over SGs;

b) be capable of being configured according to IMS Management Object defined in 3GPP TS 24.167 parameter "SMS_Over_IP_Networks_Indication", in order to give operator control to configure UE to use SMS over SGs when required.

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

Annex B: Features Needed In Certain Markets

B.1 General

This annex describes features that only a subset of the IMS telephony operators need to support in certain markets. These features typically are operationally required due to national regulatory requirements.

B.2 Global Text Telephony

In some markets, there are regulatory requirements that deaf/hearing impaired people must be able to perform text based communication to other users and government offices. In this document, this service is referred to as Global Text Telephony and the following requirements outlines how the Global Text Telephony service should be implemented in markets where required.

Global Text Telephony/teletypewriter messages must use ITU-T Recommendation T.140 real-time text according to the rules and procedures specified in 3GPP TS 26.114 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 markets, for example Japan, there are regulatory requirements that require the need to stop only voice calls while allowing high priority calls and access for other packet (for example email, web, disaster message board) services, 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 markets must support Service Specific Access Control (SSAC) as specified in Release 9 version of 3GPP TS 22.011, TS 36.331, 3GPP TS 24.173 and 3GPP TS 27.007.

Document Management

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