# GSMA PRD IR.92 – "IMS Profile for Voice and SMS"

## 1.0

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# 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 *minimum* mandatory set of features 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:

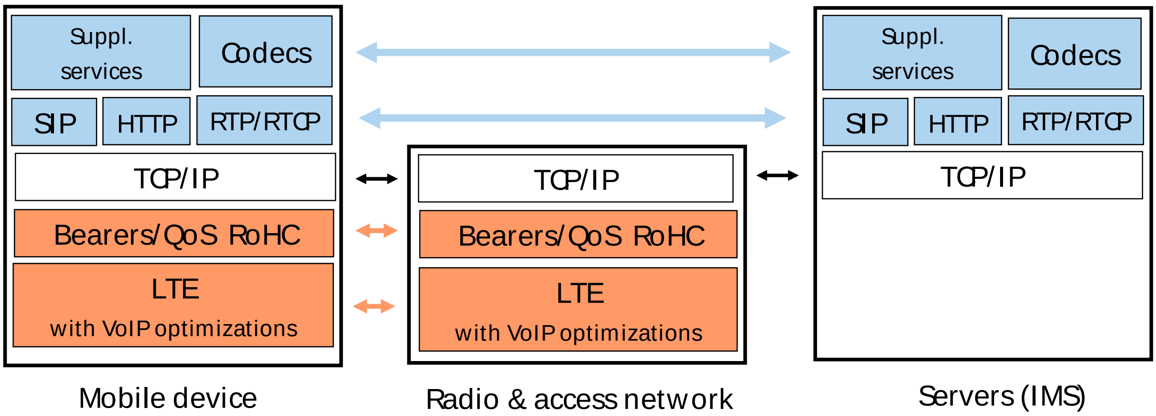


**Figure 1.1: Depiction of UE and Network Protocol Stacks in IMS Profile for Voice**

**Note**: TCP/IP includes User Datagram Protocol (UDP), and HTTP includes XML Configuration Access Protocol (XCAP) in the protocol suite.

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 requirements 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 differences 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. In general 3GPP Release 8, which is the first release supporting LTE, is taken as a basis. For some features however the functionality of 3GPP Release 9 is required, while for other features the functionality of 3GPP releases before Release 8 is considered as sufficient; the latter implies that some features, which are mandatory in 3GPP Release 8, are not required for compliance with the this profile. All such exceptions are explicitly mentioned in the following sections.

## 1.3 Scope

This document defines a voice over IMS profile by listing 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 |
| 3PCC | 3rd Party Call Control |
| AM | Acknowledged Mode |
| AMR | Adaptive Multi Rate |
| 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 |
| eNB | eNodeB |
| EPS | Evolved Packet System |
| E-UTRAN | Evolved Universal Terrestrial Radio Access Network |
| GBR | Guaranteed Bit Rate |
| GRUU | Globally Routable User agent URI |

| Term | Description |
|---|---|
| GSM | Global System for Mobile communications |
| 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 |
| 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 Initiated Protocol |
| SRB | Signalling Radio Bearer |
| SR-VCC | Single Radio Voice Call Continuity |
| T-ADS | Terminating Access Domain Selection |
| TAS | Telephony Application Server |
| 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 |
| XML | eXtensible Markup Language |

### 1.5 Document Cross-References

| Document | Name |
|---|---|
| 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 23.292 | IP Multimedia System (IMS) centralized services; 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 24.341 | 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 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; |

| Document | Name |
|---|---|
| | 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 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 |
| 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 3312 | Integration of resource management and Session Initiation Protocol (SIP) |
| 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 |
| RFC 3608 | Session Initiation Protocol (SIP) Extension Header Field for Service Route Discovery During Registration |
| 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 4032 | Update to the Session Initiation Protocol (SIP) Preconditions Framework |
| RFC 4575 | A Session Initiation Protocol (SIP) Event Package for Conference State |
| RFC 4815 | RObust Header Compression (ROHC): Corrections and Clarifications to RFC 3095 |

| Document | Name |
|---|---|
| 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 |
| OMA-ERELD-DM-V1_2-20070209-A | Enabler Release Definition for OMA Device Management, Version 1.2 |

# 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. This includes the support for service route discovery in IETF RFC 3608. The network must support the P-Visited-Network-ID header field. 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 as defined in section 5.1.1.2.1 of 3GPP TS 24.229.

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.

### 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 (IMPU)s.

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

### 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 (GRUU)s by UE or network is not required.

### 2.2.4 Call establishment and termination

UE and IMS core network must follow the SIP procedures defined in 3GPP TS 24.229 for establishment and termination of a call. In particular this includes usage of the Route header field. The UE must populate the P-Access-Network-Info header field according to 3GPP TS 24.229.

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.

### 2.2.6 Tracing of Signalling

The support of the debug event package as described in section 5.1.1.3A in 3GPP TS 24.229 is optional for the UE.

### 2.2.7 The use of Signalling Compression

The use of Signalling Compression (SigComp) must be an operator configurable option.

The support for Signalling Compression must be mandatory in the UE and the Proxy - Call Session Control Function (P-CSCF).

UE Support for Signalling Compression must follow the procedures described in Section 8.1 in 3GPP TS 24.229.

P-CSCF support for Signalling Compression must be according to the procedures of Section 8.2 in 3GPP TS 24.229.

## 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 Diversion Unconditional 3GPP TS 24.604 (Note 1) |
| Communication Diversion on not Logged in 3GPP TS 24.604 (Note 1) |
| Communication Diversion on Busy 3GPP TS 24.604 (Note 1) |
| Communication Diversion on not Reachable 3GPP TS 24.604 (Note 1) |
| Communication Diversion 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 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.

### 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 user 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 conference session between the UE and the IMS core network (conference server). The UE must add the Replaces header to the Refer-to header in REFER, 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 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 conditions and actions listed in Table 2.2. It is recommended that a UE should support the History-Info header for presentation of diverting parties.

**Note**: Support of other conditions and actions are out of scope 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 conditions listed in Table 2.3

**Note**: Support of other conditions is out of scope 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 IETF RFC 3312, and updated by IETF RFC 4032.

The UE must use the Supported header, and not the Require header, to indicate the support of precondition in accordance with Section 5.1.3.1 of 3GPP TS 24.229.

UE must always include the precondition-tag when originating an IMS session, as specified in Section 5.1.3.1 of 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 specification.

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 abort session request from Policy and Charging Rules Function (PCRF)).

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 Loss of SIP signalling bearer*

If the SIP signalling bearer is lost, the network must terminate all ongoing SIP sessions related to this UE, according to the procedures in section 5.2.8 in TS 24.229 (for example when the P-CSCF receives abort session request from PCRF).

If the SIP signalling bearer is lost, then the UE must re-establish the PDN connection (PDN connection request or PS attach, depending if the UE stays connected to a PDN or not). This will trigger the network to initiate a new SIP bearer in conjunction with the PDN connection establishment. After the SIP bearer is established, the UE must perform a new initial registration to the IMS core in case the IP address changed or the IMS registration expired during the absence of IP connectivity.

*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 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 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 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 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 and the entities in the IMS core network that terminate the user plane must support the 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 and the entities in the IMS core network that terminate the user plane must be capable of operating with any subset of these eight (8) codec modes.

When transmitting, the UE and the entities in the IMS core network that terminate the user plane 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. The UE and the entities in the IMS core network that terminates the user plane must also be capable of restricting codec mode changes to neighbouring codec modes within the negotiated codec mode set.

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.

### 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. The compound packet must include a sender report (SR) packet and a source description (SDES) packet, respectively. The SR and SDES packets must be formatted as described in detailed below:

- Sender report (SR) RTCP packet
- Version 2 must be used.
- Padding bit must not be set.
- Only one reception report must be included in one packet.
- 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).

Source description (SDES) RTCP packet

- Version and Padding as described for SR packet.
- Only one SSRC chunk must be included in one 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 comaptible 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) payload format 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 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.

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

#### *4.3.1.1 General*

**Note 1**: For IMS roaming to work, a "well-known" Access Point Name (APN) used for IMS telephony must be defined. The APN definition and method of APN name discovery is to be defined by GSMA. Once defined, the content of this section will be revised as needed to reflect the specific APN and the mechanism for discovery.

Two alternative ways can be used to configure the bearers for SIP signalling and XCAP. In the first alternative, the IMS application uses a specific APN; any other application must not use this APN. This is described in section 4.3.1.2. In the second alternative, a single APN is used for multiple applications, including IMS. This alternative is described in section 4.3.1.3. For purpose of this document, OMA device management is needed for APN configuration as defined in OMA-ERELD-DM.

The APN(s), which is (are) used for IMS signalling (SIP and XCAP) in home and visited PLMNs, must be configured in the UE.

Depending on operator policies, the UE can provide the APN in the Evolved Universal Terrestrial Radio Access Network (E-UTRAN) initial attach, or it can provide the APN when establishing another PDN connection. If UE sends an APN in the E-UTRAN initial attach, it must send the APN used for IMS telephony.

**Note 2**: Operator policies must decide if the initial attachment is used to establish connectivity to the default APN or used to establish connectivity to the telephony APN. Establishment of another PDN connection for IMS must be in conjunction with the initial PDN connection.

#### *4.3.1.2 IMS specific APN*

A default bearer must be created when the UE creates a connection to the PDN which is used for IMS telephony, as defined in 3GPP specifications. A standardised QCI value of five

(5) must be used for IMS SIP signalling, with default or dedicated bearer based on Policy and Charging Control (PCC) mechanisms.

**Note:** In the scenario where IMS is only used for telephony, it is highly recommended that the default bearer is used for SIP and XCAP. This reduces the number of EPS bearers used. However, in multimedia operations, another configuration can be used.

To enable the transport of XCAP, the PCRF must provide a PCC rule identifying the TAS's within the home network. This can be done from the home network over the S9 interface or through local configuration at the local PCRF.

#### *4.3.1.3 Multipurpose APN*

A default bearer must be created when UE creates a connection to the PDN which is used for IMS telephony, as defined in 3GPP specifications. The Combined Dedicated Bearer activation and Default Bearer activation procedure defined within 3GPP TS 23.401 must be used. The dedicated bearer for IMS SIP signalling must utilise the standardised QCI value of five (5) and have the associated characteristics as specified in 3GPP TS 23.203. The network must be provisioned with PCC rules which map the SIP signalling to an EPS bearer. These rules are unique for the APN and are used for IMS telephony and XCAP.

To enable the differentiated transport of XCAP, the PCRF must provide a PCC rule identifying the TAS's within the home network. This can be done from the home network over the S9 interface, or through local configuration at the local PCRF. This may result in XCAP being transported over the same bearer as the one created for IMS Signalling (QCI=5), or over the default bearer with QCI value eight/nine (8/9).

### 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 IMS emergency services as specified in 3GPP Release 9, TS 23.167, chapter 6.2 and annex H, and emergency procedures as specified in TS 23.401.

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.

## 5.3 Roaming considerations

**Note:** This section will be removed once GSMA has addressed Roaming for Voice over LTE in separate work.

This profile has been designed to support IMS roaming with both P-CSCF and PGW in the visited network. Other roaming models are out of the scope of the document.

The following considerations motivate this selection:

- session based charging via Rf interface is supported in home and visited network;
- signalling encryption is terminated in the P-CSCF in the visited network, which may be required to fulfil regulatory requirements;
- emergency services can be invoked most efficiently.

# ANNEX A: COMPLEMENTING IMS VOICE SERVICE WITH CIRCUIT SWITCHED VOICE ACCESS

## A.1 General

In many deployments the IMS/VoIP capable radio coverage may be initially less extensive than the concurrent Circuit Switched (CS) voice coverage. In order to offer its VoIP customers a seamless voice service already at that stage, the operator may wish to utilize the CS radio access as a complement to the IMS VoIP capable radio coverage. This Annex describes the features for the UEs and networks that wish to support such a deployment scenario need to implement, in addition to the IMS VoIP over LTE minimum feature set.

## 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 as specified in 3GPP Release 8, TS 23.221, Section 7.2a and Annexes A.1/A.2. The UE must follow the “UE behaviour when performing combined/ non-combined EPS/IMSI attach,” with the setting of: "prefer IMS PS Voice with CS Voice as secondary."

An UE must be able to assist the Service Centralization and Continuity Application Server (SCC AS) to execute terminating domain selection (UE T-ADS) as specified in 3GPP TS 23.237 and 3GPP TS 24.237.

## 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. 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 as described in TS 23.216. A SR-VCC capable terminal must indicate support for SR-VCC in the MS network capability parameter in the attach and tracking area/ routing area update messages (TS 24.301 section 9.9.3.20 and 24.008 10.5.5.12).

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

The UE must use service setting management as defined in section 2.2.2.

**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 and network can support the procedures in section 5.2.

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

If the UE supports both IMS emergency and CS emergency services, it 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. 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 UE or 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.

If the UE supports SR-VCC for IMS Telephony as described in Annex A, it must also 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 defined in 3GPP TS 24.237.

If the network supports the SR-VCC procedures for handover of an IMS telephony session from EUTRAN to CS described in Annex A, it must also support SR-VCC for IMS emergency sessions as specified in 3GPP Release 9 TS 23.216 and TS 23.237. In that case, 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

**Note:** This section will be removed once GSMA has addressed Roaming for Voice over LTE in separate work.

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 and network have the option to 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.

# 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.
Redundant transmission of real-time text characters must not be used.
The sampling time used must be 300 ms.
Change of the sampling time (rate adaptation) is not required.

# DOCUMENT MANAGEMENT

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