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One Voice;
Voice over IMS profile

The present document has been developed by AT&T, Orange, Telefonica, TeliaSonera, Verizon, Vodafone, Alcatel-Lucent, Ericsson, Nokia Siemens Networks, Nokia, Samsung and Sony Ericsson.

Keywords

LTE, VoIP, IMS, Voice

Contents

Foreword

3GPP specified IMS network is an access-independent and standard-based IP connectivity and service control architecture that enables various types of multimedia services to end-users using common Internet-based protocols. In fixed networks Voice over IP (VoIP) has been deployed or being deployed, whereas in mobile-side combinational applications using Circuit Switched voice, such as Rich Communication Suite, are driving deployments.

In the next coming years, more efficient radio solutions will be deployed at an increased rate to match growing demand in the packet switched domain. The migration to a packet-based access and core network creates the need for a voice evolution story from existing CS networks to Voice over IP using IMS.

3GPP has worked with the IMS almost ten years and there exist thousands of pages, in different specifications, that cover IMS related functionalities. In the meantime, a sophisticated architecture and feature set has been developed. Not all of this functionality is needed for an initial deployment of a cellular IMS based VoIP network. Moreover, 3GPP has specified multiple, different ways to complete single functions (e.g. authentication, session setup, supplementary service execution, bearer setup) which increases complexity of the IMS. To help industry secure a common standardized IMS voice solution, this specification defines a common recommended feature set and selects one recommended option when multiple options exist for single functionality.

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1 Scope

This document defines a voice over IMS profile by listing number of 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.

2 References

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

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document in 3GPP Release 8.

- 3GPP TR 21.801: "Specification drafting rules".
- 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; 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 36.322: "Evolved Universal Terrestrial Radio Access (E-UTRA); Radio Link Control (RLC) protocol specification".
- 3GPP TS 36.323: "Evolved Universal Terrestrial Radio Access (E-UTRA); Packet Data Convergence Protocol (PDCP) specification".
- RFC 768 (28 August 1980): "User Datagram Protocol".
- RFC 3095 (July 2001): "RObust Header Compression (ROHC): Framework and four profiles: RTP, UDP, ESP, and uncompressed".
- RFC 3261 (June 2002):"SIP: Session Initiation Protocol".
- RFC 3312 (October 2002): "Integration of resource management and Session Initiation Protocol (SIP)".
- RFC 3550 (July 2003): "RTP: A Transport Protocol for Real-Time Applications".
- RFC 3551 (July 2003): "RTP Profile for Audio and Video Conferences with Minimal Control".
- RFC 3556 (July 2003): "Session Description Protocol (SDP) Bandwidth Modifiers for RTP Control Protocol (RTCP) Bandwidth".
- RFC 3608 (October 2003): "Session Initiation Protocol (SIP) Extension Header Field for Service Route Discovery During Registration".
- RFC 3680 (March 2004): "A Session Initiation Protocol (SIP) Event Package for Registrations".
- RFC 3842 (August 2004) "A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)".
- RFC 4032 (March 2005): "Update to the Session Initiation Protocol (SIP) Preconditions Framework".
- RFC 4575 (August 2006): "A Session Initiation Protocol (SIP) Event Package for Conference State".
- RFC 4815 (February 2007): "RObust Header Compression (ROHC): Corrections and Clarifications to RFC 3095".
- RFC 4867 (April 2007): "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 (May 2009): "SDP Capability Negotiation".
- OMA-ERELD-DM-V1_2-20070209-A : "Enabler Release Definition for OMA Device Management, Version 1.2".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of this specification, the following terms and definitions, according to the 3GPP definitions in Annex E of TR 21.801, apply.

- The term “shall” or “must” means it is a requirement, or mandatory.
- The term “should” means it is recommended.
- The term “may” means it is permitted or allowed.

3.2 Abbreviations

For the purposes of this specification, the following abbreviations apply:

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 Fall-Back
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
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

4 General introduction

4.1 The One Voice Profile

One Voice profile defines a *minimum* mandatory set of features a wireless device (the UE) and network are required to implement in order to guarantee an interoperable, high quality IMS-based telephony service over LTE radio access. The scope includes the following aspects

- IMS basic capabilities and supplementary services for telephony [Chapter 5]
- Real-time media negotiation, transport, and codecs [Chapter 6]
- LTE radio and evolved packet core capabilities [Chapter 7]
- Functionality that is relevant across the protocol stack and subsystems [Chapter 8]

The UE and network protocol stacks forming the scope of the One Voice profile are depicted in the figure below:

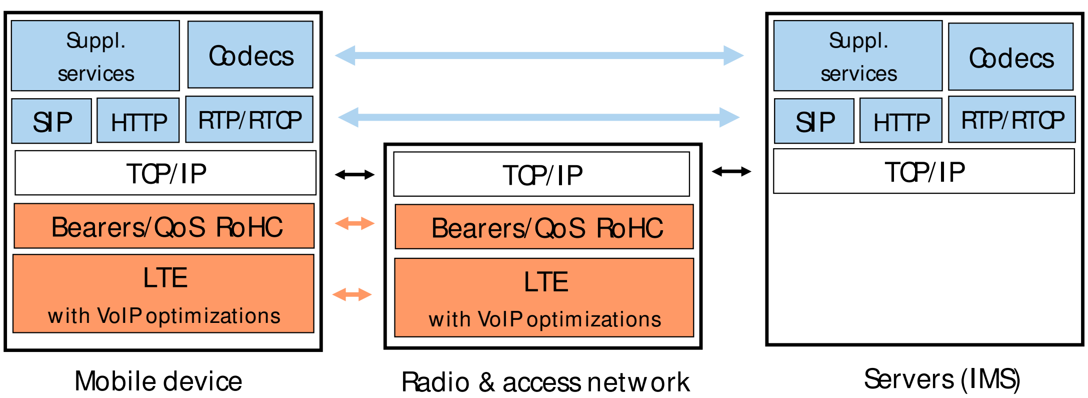

FIGURE 1: Depiction of UE and Network Protocol Stacks in One Voice

NOTE: TCP/IP includes UDP, and HTTP includes XCAP in the protocol suite

The main body of One Voice profile 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 *One Voice Main Profile,* 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 *3GPP One Voice CS-coexistence Profile*, 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 unaltered by [Annex A].

4.2 Relationship to the existing standards

4.2.1 3GPP Specifications

This profile is solely based on the open and published 3GPP specifications as listed in the chapter 2. 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 one voice profile. All such exceptions are explicitly mentioned in the following sections.

5 IMS feature set

5.1 General

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

5.2 Support of generic IMS functions

5.2.1 SIP Registration Procedures

UE and IMS core network shall follow the SIP registration procedures defined in 3GPP TS 24.229. This includes the support for service route discovery in IETF RFC 3608. The network shall support the P-Visited-Network-ID header field.

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

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

5.2.2 Authentication

UE and IMS core network shall follow the procedures defined in 3GPP TS 24.229 and 3GPP TS 33.203 for authentication with 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 shall support the procedures for ISIM based authentication. Support for ISIM based authentication in the UE is mandatory.

UE and IMS core network shall the support the procedures for USIM based authentication in case there is no ISIM present on the 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 URI header to handle barred IMPUs.

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

5.2.3 Addressing

UEs and IMS core network using this profile shall support SIP URIs (alphanumeric) and MSISDN based IMPU, which means a tel-URIs with an associated SIP-URI, e.g.

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

The UE and network shall 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 shall set the dialstring 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 shall 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 shall set the “phone-context” parameter to the home domain name, as it is used to address the SIP REGISTER request The UE and network may support geo-local numbers. If the UE supports geo-local numbers, it shall 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 shall support the P-Called-Party-ID header field; the network shall use this header field as defined in 3GPP TS 24.229.

The support of GRUUs by UE or network is not required.

5.2.4 Call establishment and termination

UE and IMS core network shall 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 shall populate the P-Access-Network-Info header field according to 3GPP TS 24.229.

UE and IMS core network shall support reliable provisional responses.

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

The usage of preconditions is discussed in Section 5.4.

5.2.5 Forking

Forking in the network is outside the scope of the present document. However for inter-operability and forward-compatibility reasons, the UE shall 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.

5.2.6 Tracing of Signalling

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

5.2.7 The use of Signalling Compression

The use of Signalling Compression (SigComp) shall be an operator configurable option. The support for Signalling Compression shall be mandatory in the UE and the P-CSCF.

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

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

5.3 Supplementary services

5.3.1 Supplementary services overview

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

UE and TAS shall support the supplementary services listed in Table 5.1. It is up to the operator to enable these services.

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

Table 5.1 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 5.3.3 3 – 5.3.9

NOTE 2: Barring of International Calls is a 3GPP Release 9 feature.

5.3.2 Supplementary Service Configuration

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

5.3.3 Ad-Hoc Multi Party Conference

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

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 shall 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 shall support the procedure described in Section 5.3.1.5.3 of 3GPP TS 24.147. The UE shall send the REFER method by using the existing dialog for conference session between the UE and the IMS core network (conference server). The UE shall 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 may and the IMS core network shall support the procedures in 3GPP TS 24.605 for subscription to conference state events. The IMS core network may support all, or a subset of the elements and attributes in IETF RFC 4575 As a minimum, the IMS core network shall 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 shall support audio media for the conference session.

NOTE: Support of other media types are 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.

5.3.4 Communication Waiting

UE and IMS core network shall support the terminal based service, as described in 3GPP TS 24.615. Network-based service is not required. 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.

5.3.5 Message Waiting Indication

UE and IMS core network shall support the MWI event package, as defined in 3GPP TS 24.606 and IETF RFC 3842.

5.3.6 Originating Identification Restriction

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

5.3.7 Terminating Identification Restriction

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

5.3.8 Communication Diversion

UE and IMS core network shall support the SIP procedures in 3GPP TS 24.604 for CDIV. The CDIV Notification (CDIVN) service is not required. For CDIV service activation, deactivation, and interrogation (XCAP operations), the UE and IMS core network shall support the conditions and actions listed in Table 5.2. 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 5.2 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 CFNRc implies diversion when the user is not registered in the CS core or cannot be reached. To mimic this behavior, it is recommended that an UE activates both the CFNRc (CDIV using condition not-reachable) and the CFNL (CDIV using condition not-registered) to the same target.

5.3.9 Communication Barring

UE and IMS core network shall support the SIP procedures in 3GPP TS 24.611. For service activation, deactivation, and interrogation (XCAP operations), the UE and IMS core network shall support the conditions listed in Table 5.3

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

Table 5.3 Supported conditions in CB

roaming
international
international-exHC

5.4 Call set-up considerations

5.4.1 SIP Precondition Considerations

The UE shall support the SIP preconditions framework, as specified in IETF RFC 3312, and updated by IETF RFC 4032.

The UE shall 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 shall always include the precondition-tag when originating an IMS session, as specified in Section 5.1.3.1 of 3GPP TS 24.229.

Operators may wish to disable the use of preconditions in the network; the means of which are outside the scope of this specification.

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

5.4.2 Integration of resource management and SIP

5.4.2.1 Loss of PDN connectivity

If the PDN connectivity between a UE and the network is lost, the network shall terminate all ongoing SIP sessions related to this UE, according to the procedures in Section 5.2.8 of 3GPP TS 24.229 (e.g. when the P-CSCF receives abort session request from PCRF).

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

5.4.2.2 Loss of SIP signaling bearer

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

If the SIP signaling bearer is lost, then the UE shall 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 shall 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.

5.4.2.3 Loss of media bearer and Radio Connection

If a GBR bearer used for voice fails to get established, or is lost mid-session, then network shall terminate the session associated to the voice stream according to the procedures in section 5.2.8 in TS 24.229 (P-CSCF shall 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, should either allow the session to continue as is, or terminate the SIP session that the GBR bearer is associated with (i.e. the network may 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 shall, based on its preferences, modify, reject or terminate the SIP session that the dedicated media bearer is associated with, according to Sub-section 6.1.1 in 3GPP TS 24.229. The UE may act differently per media type.

NOTE 3: In case the 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. But as a complement to this, the UE needs to have internal logic to react to the detection of loss of bearer/radio connection to handle its internal state. In case of multimedia communication, if the radio connection is not lost, but a bearer not used for voice is lost, then the UE is in charge of deciding if the session should be maintained as is, modified, or released,

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

5.4.3 Voice Media Considerations

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

5.4.4 Multimedia Considerations

UEs using the full set of media functions may 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 shall accept the offer and allow the use of whatever media streams it supports. The UE shall 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 (i.e. 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, may 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 shall accept such offer and allow the use of whatever media streams it supports. The UE shall, 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.

5.5 SMS over IP

The UE shall 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 shall be supported.

The IMS core network shall 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.

6 IMS media

6.1 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.

6.2 Voice Media

6.2.1 Codecs

The UE and the entities in the IMS core network that terminate the user plane shall 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 shall 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 shall be capable of aligning codec mode changes to every frame border, and shall also be capable of restricting codec mode changes to be aligned to every other frame border, e.g. like UMTS_AMR_2 as described in 3GPP TS 26.103. The UE and the entities in the IMS core network that terminates the user plane shall 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 shall allow codec mode changes at any frame border and to any codec mode within the negotiated codec mode set.

6.2.2 RTP Profile and SDP Considerations

6.2.2.1 RTP Profile

The RTP profile AVP IETF RFC 3551 shall be used.

6.2.2.2 SDP Offer Considerations

The SDPCapNeg framework [draft-ietf-mmusic-sdp-capability-negotiation-10 (May 2009): "SDP Capability Negotiation”] shall not be used in the SDP offer when the AVP profile is used.

6.2.2.3 SDP Answer Considerations

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

NOTE: In 3GPP TS 26.114 section 6.2.1a, it is specified that a UE or the IMS core network should use the SDPCapNeg attributes ‘tcap’ and ‘pcfg’ to indicate the support of both the RTP profiles AVP and 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.

6.2.3 Data Transport

The UE and the entities in the IMS core network that terminate the user plane shall use RTP over UDP as described in IETF RFC 3550 and IETF RFC 768, respectively, to transport voice.

The UE shall 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 shall accept RTP packets that are not received from the same remote port where RTP packets are sent.

6.2.4 RTCP Usage

The RTP implementation shall include an RTCP implementation according to IETF RFC 3550.

The UE and the entities in the IMS core network that terminates the user plane shall 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 shall accept RTCP packets that are not received from the same remote port where RTCP packets are sent.

The bandwidth for RTCP traffic shall 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 shall include the "b=RS:" and "b=RR:" fields in SDP, and shall be able to interpret them.

In active “speech-only sessions,” the RTCP transmission shall 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 shall 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 shall 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 shall support the transmission of RTCP packets formatted according to the rules in IETF RFC 3550 and with the clarifications below:

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

Sender report (SR) RTCP packet

- Version 2 shall be used.
- Padding bit shall not be set.
- Only one reception report shall be included in one packet.
- NTP timestamp should be set to zero (0) (usage not required).
- RTP timestamp should be the same as in the previously sent RTP packet (usage not required).
- Last SR timestamp (LSR) should 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 shall be included in one packet.
- The SDES item CNAME shall be included in one packet.
- Only one SDES item should be used (CNAME, usage of other SDES items not required).

NOTE 3: Because the randomly allocated SSRC identifier may change, the CNAME item shall 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 shall 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) shall 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.

6.2.5 AMR Payload Format Considerations

The AMR payload format IETF RFC 4867 shall be supported.

The UE and the entities in the IMS core network that terminates the user plane shall support the bandwidth-efficient and the octet-aligned format. The UE and the entities in the IMS core network that terminates the user plane shall 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 shall 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 shall request to receive one speech frame encapsulated in each RTP packet, but shall 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 shall be set to 20 and the maxptime attribute shall be set to 240 in the SDP negotiation.

The UE and the entities in the IMS core network that terminates the user plane shall be able to sort out the received frames based on the RTP Timestamp and shall remove duplicated frames, if present. If multiple versions of a frame are received, i.e. 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 terminates the user plane, using the full set of media functions, may send frames several times (i.e. 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.

6.2.6 Jitter Buffer Management Considerations

The minimum performance requirements for jitter buffer management of voice media, as described in 3GPP TS 26.114. shall be met. Voice packet flows can typically tolerate delays on the order of 100 ms and packet losses of up to 1 % (percent), since state-of-the-art AMR voice codecs perform well up to these error rates.

6.2.7 Front end handling

UEs used for IMS voice services shall 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 shall be as specified in 3GPP TS 26.132.

7 Radio and packet core feature set

7.1 General

7.2 Robust Header Compression

UE and network shall support Robust Header Compression (RoHC) as specified in 3GPP TS 36.323, IETF RFC 3095 and IETF RFC 4815. The UE and network shall 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 shall support "RTP/UDP/IP" profile (0x0001) to compress RTP packets and "UDP/IP" profile (0x0002) to compress RTCP packets. The UE and network shall support these profiles for both IPv4 and IPv6.

7.3 LTE radio capabilities

7.3.1 Radio Bearers

The UE shall 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 shall support the following combination of radio bearers:

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

One AM DRB is utilized for EPS bearer with QCI = 5 and another AM DRB for EPS bearer with QCI = 8/9. UM DRB is utilized for EPS bearer with QCI = 1.

7.3.2 DRX mode of operation

In order to maximize lifetime of the UE battery, LTE DRX method as specified in 3GPP TS 36.300 and TS 36.321 shall be deployed. Support of DRX is mandatory for both UE and network.

7.3.3 RLC configurations

RLC entity shall 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 (used for voice user plane traffic).
- Acknowledged Mode (AM) for EPS bearers with QCI = 5 (used for SIP signalling associated with voice service)
- Acknowledged Mode (AM) for EPS bearers with QCI = 8/9 (used for other IMS traffic)

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).

7.3.4 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 TS23.401. 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 shall 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, shall 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.

7.3.5 RF Performance

RF performance must meet the requirements for each frequency band and bandwidth configuration, according to 3GPP TS 36.104 (eNB) and TS 36.101 (UE).

7.4 Bearer management

7.4.1 EPS bearer considerations for SIP signalling and XCAP

7.4.1.1 General

Two alternative ways may 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 shall not use this APN. This is described in clause 7.4.1.2.

In the second alternative, a single APN is used for multiple applications, including IMS. This alternative is described in clause 7.4.1.3.

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, shall be configured in the UE

NOTE 1: For IMS roaming to work, the "well-known" APN used for IMS telephony must be defined. The APN definition and method of APN name discovery is out of scope for this document.

Depending on operator policies, the UE may provide the APN in the E-UTRAN initial attach, or it may provide the APN when establishing another PDN connection. If UE sends an APN in the E-UTRAN initial attach, it shall send the APN used for IMS telephony.

NOTE 2: Operator policies shall 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 shall be in conjunction with the initial PDN connection.

7.4.1.2 IMS specific APN

A default bearer shall 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 5 shall be used for IMS SIP signalling, with default or dedicated bearer based on 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 may be used.

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

7.4.1.3 Multipurpose APN

A default bearer shall 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 [d] shall be used. The dedicated bearer for IMS SIP signalling shall utilise the standardised QCI value of 5 and have the associated characteristics as specified in 3GPP TS 23.203. The network shall be provisioned with PCC rules which map the SIP signaling 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 shall provide a PCC rule identifying the TAS's within the home network. This may 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 8/9.

7.4.2 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 shall be created utilising interaction with dynamic PCC. The network shall initiate the creation of a dedicated bearer to transport the voice media. The dedicated bearer for Conversational Voice shall utilise the standardised QCI value of 1 and have the associated characteristics as specified in 3GPP TS 23.203. Since the minimum requirement for the UE is the support of 1 UM bearer which is used for voice (see Section 7.3.1 and Annex B in 3GPP TS 36.331), the network shall not create more than one dedicated bearer for voice media. Therefore, the UE and network shall be able to multiplex the media streams from multiple concurrent voice sessions.

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

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

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

7.5 P-CSCF discovery

The UE and packet core shall 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.

8 Common functionalities

8.1 IP version

The UE and the network shall 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 shall 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 shall 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 shall 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.

8.2 Emergency Service

8.2.1 General

One Voice UEs and network deployments shall support emergency services in the IMS domain,

The UE and the network shall 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 shall 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.

8.3 Roaming considerations

The One Voice main 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 shall 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 shall 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 shall 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 shall be able to assist the SCC AS to execute terminating domain selection (UE T-ADS) as specified in 3GPP TS 23.216 Section 5.3.4.2..

A.3 SR-VCC

The network shall support the SR-VCC procedures for handover from E-UTRAN as described in TS 23.216. The UE detects that 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 shall support the SR-VCC procedures as described in TS 23.216. A SR-VCC capable terminal shall 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 shall use service setting management as defined in section 5.3.2.

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 8.4 in the following ways:

The UE and network may support the procedures in section 8.4.

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

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

If the UE supports SR-VCC for IMS Telephony as described in Annex A, it shall 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 shall 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 shall 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 shall support the SIP instance ID as described in 3GPP TS 24.237.

A.7 Roaming Considerations

Clause 8.4 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 shall follow procedures defined in clause A.2 to use CS for voice service.

A.8 SMS Support

This section modifies the requirements defined in section 5.5 in the following ways:

The UE and network may support the SMS-over-IP as described in section 5.2.8. In addition, when support of SMS over SGs is required, the UE and network shall 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 shall 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 shall 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 shall offer AVP for all media streams containing real-time text
- For real-time text, RTCP reporting shall be turned of by setting the SDP bandwidth modifiers “RS” and “RR” to zero.
- Redundant transmission of real-time text characters shall not be used
- The sampling time used shall be 300 ms
- Change of the sampling time (rate adaptation) is not required.

Annex C: Change History

Date	Subject/Comment	Version
2009-11-3	One Voice Profile Version 1.0.0	1.0.0