IMS Profile for Voice, Video and Messaging over 5GS

Version 1.0

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1 Introduction

Overview

The IP Multimedia Subsystem (IMS) Profile for voice, video and messaging, documented in this Permanent Reference Document (PRD), defines a profile that identifies a minimum mandatory set of features which are defined in 3GPP and GSMA specifications that a wireless device (the User Equipment (UE)) and network are required to implement in order to guarantee interoperable, high quality IMS-based communication services for voice, video and messaging over NG (Next Generation) radio access connected to 5GC. The content includes the following aspects:

1. Basic capabilities and supplementary features for IMS-based communication services for voice, video and messaging [Section 2].
2. IMS media negotiation, transport, and codecs [Section 3].
3. Radio and packet core feature set [Section 4].
4. Common functionalities that are relevant across the protocol stack and subsystems [Section 5].
5. Additional features that only a subset of the IMS telephony operators needs to support in certain regions [Annex B].
6. UE configuration to provide all necessary information to connect to, and to receive service from, a specific IMS telephony and messaging operator [Annex C].
7. Support for Unstructured Supplementary Service Data (USSD) Simulation Service in IMS (USSI) as optional feature [Annex A].

The main body of this PRD is applicable for a scenario where IMS-based voice, video and messaging services are deployed in the 5G System (NG-RAN,5GC, and UE. In order to be fully compliant with this IMS Profile for voice, video and messaging, the UEs and networks must be compliant with all normative statements in the main body. Messaging related statements in this document apply to only those 5G UEs that support messaging. Messaging covered in this document includes both RCS (Rich Communication Suite) Messaging services and SMS (Short Message Service).

The present version of this PRD is restricted to profiling related to NG-RAN option SA NR per the definition in 3GPP TS 23.501 [4].

Relationship to existing standards

3GPP and GSMA specifications

This profile is based solely on the 3GPP and GSMA specifications as listed in Section 1.5. 3GPP Release 15 is taken as a basis unless otherwise stated. When GSMA documents are referenced, the 3GPP release reference is as specified in those GSMA documents.

IMS features are based on 3GPP Release 15 unless otherwise stated, including those needed to support interworking with EPS (e.g. EPS Fallback (Evolved Packet System Fallback)).

Note: The RCS Universal Profile version to be taken as the basis for the RCS Messaging and MSRP (Message Session Relay Protocol)-based Enriched Calling services referred in section 2.5 and 2.6 respectively, is for further study.

Scope

This document defines a profile for voice, video, RCS Messaging and MSRP based Enriched Calling services over IMS, as well as SMS by listing a number of NG-RAN, 5GC, IMS core and UE features and procedures that are considered essential to launch interoperable services. The defined profile is compliant with and based on:

1. 3GPP specifications related to 5GS, voice and video services over IMS and SMS, and
2. GSMA specifications related to RCS Messaging and MSRP based Enriched Calling.

The scope of this profile is the interface between the UE (User Equipment) and the network.

The profile does not limit, by any means, deploying other standardized features or optional features, in addition to those defined in this profile.

Definition of Acronyms and Terms

Acronyms

Acronym	Description
3GPP	3rd Generation Partnership Project
3PCC	3rd Party Call Control
5GC	5G Core Network
AM	Acknowledged Mode
AN	Access Node
AMR	Adaptive Multi-Rate
AMR-WB	Adaptive Multi-Rate Wideband
APN	Access Point Name
APP	Application
AVP	Audio Video Profile
AVPF	AVP Feedback Profile
B2BUA	Back to Back User Agent
BSF	Bootstrapping Server Function
CB	Communication Barring
CDIV	Communication Diversion
CFNL	Communication Forwarding on Not Logged-in
CFNRc	Communication Forwarding on Not Reachable
CMR	Codec Mode Request

Acronym	Description
CS	Circuit Switched
CVO	Coordination of Video Orientation
CW	Communication Waiting
DNN	Data Network Name
DRB	Data Radio Bearer
DRX	Discontinuous Reception
DTX	Discontinuous Transmission
ECT	Explicit Communication Transfer
EPC	Evolved Packet Core
EPS	Evolved Packet System
E-UTRAN	Evolved Universal Terrestrial Radio Access Network
EVS	Enhanced Voice Services
FDD	Frequency-Division Duplexing
GBR	Guaranteed Bit Rate
GRUU	Globally Routable User agent URI
GSM	Global System for Mobile communications
GTT-IP	Global Text Telephony over IP
HOS	Home Operator Services
HPMN	Home Public Mobile Network
ICSI	IMS Communication Service Identifier
IM	IP Multimedia
IMEI	International Mobile Equipment Identity
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
IRAT	Inter-Radio Access Technology
ISIM	IM Services Identity Module
LTE	Long Term Evolution
MGW	Media Gateway
MMTel	Multimedia Telephony
MO	Management Object
MRFP	Media Resource Function Processor
MRF	Media Resource Function
MSISDN	Mobile Subscriber ISDN Number

Acronym	Description
MSRP	Message Session Relay Protocol
MTSI	Multimedia Telephony Services for IMS
MWI	Message Waiting Indication
NAS	Non-Access-Stratum
NGBR	Non-Guaranteed Bit Rate
NG	Next Generation
PCC	Policy and Charging Control
P-CSCF	Proxy - Call Session Control Function
PCF	Policy Control Function
PCO	Protocol Configuration Options
PDN	Packet Data Network
PDU	Protocol Data Unit
PPS	Picture Parameter Sets
PS	Packet Switched
QCI	Quality of Service Class Indicator
RAN	Radio Access Network
RAT	Radio Access Technology
RCS	Rich Communication Suite
RLC	Radio Link Control
RoHC	Robust Header Compression
RR	Receiver Report
RTCP	RTP Control Protocol
RTP	Real Time Protocol
SDES	Source Description
SDP	Session Description Protocol
SIGCOMP	Signalling Compression
SMS	Short Message Service
simservs	MMTel supplementary services XML document
SIP	Session Initiation Protocol
SMSoIP	SMS over IP
SPS	Sequence Parameter Sets
SR	Sender Report
SUPL	Secure User Plane Location
TAS	Telephony Application Server
TDD	Time-Division Duplexing
TFO	Tandem-Free Operation
TrFO	Transcoder-Free Operation
TTY	Teletype Writer

Acronym	Description
UDP	User Datagram Protocol
UDUB	User Determined User Busy
UE	User Equipment
UICC	Universal Integrated Circuit Card
UM	Unacknowledged Mode
URI	Uniform Resource Identifier
UP	Universal Profile
USSD	Unstructured Supplementary Service Data
USSI	Unstructured Supplementary Service Data (USSD) using IP Multimedia (IM) Core Network (CN) subsystem (IMS)
SR-VCC	Single Radio Voice Call Continuity
XCAP	XML Configuration Access Protocol
XML	eXtensible Markup Language

Terms

Term	Description
3GPP PS Data Off	A feature which when configured by the HPMN (Home Public Mobile Network) and activated by the user prevents transport via PDN (Packet Data Network) connections in 3GPP access of all IP (Internet Protocol) packets except IP packets required by 3GPP PS (Packet Switch) Data Off Exempt Services, as defined in 3GPP TS 22.011 [92]. Data Off can be activated only when the UE roams or regardless whether the UE roams or not, depending on UE implementation.
3GPP PS Data Off Exempt Services	A set of operator services that are allowed even if the 3GPP PS Data Off feature has been activated in the UE by the user, as defined in 3GPP TS 22.011 [92].
3GPP PS Data Off status	Indicates state of usage of the 3GPP PS data off. 3GPP PS data off status at the UE can be either "active" or "inactive", as defined in 3GPP Release 14 TS 24.229 [8].
Call Composer Element	An information element set by the originating user for an outgoing call invitation and used by the terminating UE when notifying the user of an incoming call invitation. The following Call Composer Elements are defined: importance subject picture location The mapping of Call Composer Elements to SIP headers is as defined in section 2.4.4.2 of GSMA PRD RCC.20 [72].
Downloadable IMS Client	An IMS client that has been downloaded or pre-installed onto a device carrying a UICC but is unable to access the credentials on that UICC.

Term	Description
MMTEL Call Composer	The "Call Composer service using the Multimedia Telephony session" as defined in section 2.4.4 of GSMA PRD RCC.20 [72] This service, when provided by the HPMN (Home Public Mobile Network), allows: the originating UE to send one or more Call Composer Elements with a call invitation. the terminating UE to accept Call Composer Elements received in a call invitation.
MSRP Call Composer	The "Call Composer service using an Enriched Calling session " as defined in section 2.4.3 of GSMA PRD RCC.20 [72].
MSRP based Enriched Calling services	The set of services as defined in section 3.3 of GSMA PRD RCC.07 [73] and GSMA PRD RCC.20 [72]. These services comprise: Shared Map, Shared Sketch, MSRP Call Composer & Post-call Service.
Native IMS Client	An IMS client that is provided as part of the handset implementation of a device carrying a UICC and is able to access the credentials on that UICC.
RCS Messaging services	The set of services as defined in sections 3.2 and 3.6 of GSMA RCC.07 [73]. These services comprise: 1-to-1 Chat, Standalone Messaging (Pager Mode & Large Message Mode), Group Chat, File Transfer (FT), Geolocation Push, Audio Messaging & Chatbots.
Region	A part of a country, a country or a set of countries.

References

Ref	Doc Number	Title
[1]	IETF RFC 2119	Key words for use in RFCs to Indicate Requirement Levels.
[2]	GSMA PRD IR.92	IMS Profile for Voice and SMS.
[3]	GSMA PRD IR.65	IMS Roaming and Interworking Guidelines
[4]	3GPP TS 23.501	System Architecture for the 5G System
[5]	3GPP TS 23.502	Procedures for the 5G System
[6]	3GPP TS 23.503	Policy and Charging Control Framework for the 5G System
[7]	3GPP TS 24.501	Non-Access-Stratum (NAS) Protocol for the 5G System (5GS) stage 3
[8]	3GPP TS 24.229	IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3
[9]	3GPP TS 24.305	Selective Disabling of 3GPP User Equipment Capabilities (SDoUE) Management Object (MO)
[10]	3GPP TS 24.173	IMS Multimedia telephony service and supplementary services; Stage 3
[11]	3GPP TS 24.341	Support of SMS over IP networks; Stage 3
[12]	3GPP TS 24.237	IP Multimedia Subsystem (IMS) Service Continuity; Stage 3
[13]	3GPP TS 23.003	Numbering, addressing and identification

Ref	Doc Number	Title
[14]	3GPP TS 31.103	Characteristics of the IP Multimedia Services Identity Module (ISIM) application
[15]	3GPP TS 23.228	IP Multimedia Subsystem (IMS); Stage 2
[16]	3GPP TS 26.114	IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction
[17]	IETF RFC 3261	SIP: Session Initiation Protocol
[18]	IETF RFC 3840	Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)
[19]	IETF RFC 3841	Caller Preferences for the Session Initiation Protocol (SIP)
[20]	IETF RFC 4122	Universally Unique IDentifier (UUID) URN Namespace
[21]	IETF RFC 3262	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
[22]	3GPP TS 22.030	Man-Machine Interface (MMI) of the User Equipment (UE)
[23]	IETF RFC 5009	Private Header (P-Header) Extension to the Session Initiation Protocol
[24]	3GPP TS 24.628	Common Basic Communication procedures using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[25]	3GPP TS 33.203	3G security; Access security for IP-based services
[26]	3GPP TS 24.623	Extensible Markup Language (XML) Configuration Access Protocol (XCAP) over the Ut interface for Manipulating Supplementary Services
[27]	3GPP TS 33.222	Generic Authentication Architecture (GAA); Access to network application functions using Hypertext Transfer Protocol over Transport Layer Security (HTTPS)
[28]	3GPP TS 24.109	Bootstrapping interface (Ub) and network application function interface (Ua); Protocol details
[29]	3GPP TS 24.301	Non-Access-Stratum (NAS) protocol for Evolved Packet System (EPS)
[30]	3GPP TS 38.413	NG-RAN; NG Application Protocol (NGAP)
[31]	3GPP TS 38.331	NR; Radio Resource Control (RRC) protocol specification
[32]	3GPP TS 24.606	Message Waiting Indication (MWI)using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[33]	3GPP TS 24.607	Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[34]	3GPP TS 24.608	Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[35]	3GPP TS 24.610	Communication HOLD (HOLD) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification

Ref	Doc Number	Title
[36]	3GPP TS 24.611	Anonymous Communication Rejection (ACR) and Communication Barring (CB) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[37]	3GPP TS 24.615	Communication Waiting (CW) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol Specification
[38]	3GPP TS 24.623	Extensible Markup Language (XML) Configuration Access Protocol (XCAP) over the Ut interface for Manipulating Supplementary Services
[39]	3GPP TS 24.604	Communication Diversion (CDIV) using IP Multimedia (IM)Core Network (CN) subsystem; Protocol specification
[40]	3GPP TS 24.605	Conference (CONF) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[41]	3GPP TS 24.629	Explicit Communication Transfer (ECT) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[42]	3GPP TS 24.147	Conferencing using the IP Multimedia (IM) Core Network (CN) subsystem; Stage 3
[43]	IETF RFC 4745	Common Policy: A Document Format for Expressing Privacy Preferences
[44]	IETF RFC 4825	The Extensible Markup Language (XML) Configuration Access Protocol (XCAP)
[45]	IETF RFC 4575	A Session Initiation Protocol (SIP) Event Package for Conference State
[46]	IETF RFC 3842	A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
[47]	IETF RFC 3095	RObust Header Compression (ROHC): Framework and four profiles: RTP, UDP, ESP, and uncompressed
[48]	IETF RFC 4815	RObust Header Compression (ROHC): Corrections and Clarifications to RFC 3095
[49]	3GPP TS 38.300	NR; NR and NG-RAN Overall Description
[50]	3GPP TS 38.321	NR; Medium Access Control (MAC) Protocol Specification
[51]	3GPP TS 38.322	NR: Radio Link Control (RLC) Protocol Specification
[52]	ITU-T Recommendation T.140	Protocol for multimedia application text conversation
[53]	IETF RFC 4028	Session Timers in the Session Initialization Protocol (SIP)
[54]	3GPP TS 38.323	NR: Packet Data Convergence Protocol (PDCP) Specification
[55]	GSMA NG.113	5G Roaming Guidelines
[56]	3GPP TS 26.441	Codec for Enhanced Voice Services (EVS); General overview
[57]	3GPP TS 26.442	Codec for Enhanced Voice Services (EVS); ANSI C code (fixed-point)
[58]	3GPP TS 26.445	Codec for Enhanced Voice Services (EVS); Detailed Algorithmic Description

Ref	Doc Number	Title
[59]	3GPP TS 26.446	Codec for Enhanced Voice Services (EVS); AMR-WB Backward Compatible Functions
[60]	3GPP TS 26.447	Codec for Enhanced Voice Services (EVS); Error Concealment of Lost Packets
[61]	3GPP TS 26.449	Codec for Enhanced Voice Services (EVS); Comfort Noise Generation (CNG) Aspects
[62]	3GPP TS 26.450	Codec for Enhanced Voice Services (EVS); Discontinuous Transmission (DTX)
[63]	3GPP TS 26.451	Codec for Enhanced Voice Services (EVS); Voice Activity Detection (VAD)
[64]	3GPP TS 26.443	Codec for Enhanced Voice Services (EVS); ANSI C code (floating-point)
[65]	3GPP TS 23.221	Architectural Requirements
[66]	IETF RFC 4796	The Session Description Protocol (SDP) Content Attribute
[67]	3GPP TS 24.167	3GPP IMS Management Object (MO)
[68]	3GPP TS 24.275	Management Object (MO) for Basic Communication Part (BCP) of IMS Multimedia Telephony (MMTEL) communication service
[69]	3GPP TS 24.368	Non-Access Stratum (NAS) configuration Management Object (MO).
[70]	3GPP TS 24.424	Management Object (MO) for Extensible Markup Language (XML) Configuration Access Protocol (XCAP) over the Ut interface for Manipulating Supplementary Services (SS)
[71]	3GPP TS 24.008	Mobile Radio Interface Layer 3; Core Network protocols;
[72]	GSMA PRD RCC.20	Enriched Calling Technical Specification
[73]	GSMA PRD RCC.07	Rich Communication Suite 9.0 Advanced Communications Services and Client Specification
[74]	GSMA PRD RCC.14	Service Provider Device Configuration
[75]	GSMA PRD TS.32	Technical Adaptation of Devices through Late Customisation
[76]	IETF RFC 3550	RTP: A Transport Protocol for Real-Time Applications
[77]	IETF RFC 4961	Symmetric RTP / RTP Control Protocol (RTCP)
[78]	IETF RFC 3556	Session Description Protocol (SDP) Bandwidth Modifiers for RTP Control Protocol (RTCP) Bandwidth
[79]	IETF RFC 5939	Session Description Protocol (SDP) Capability Negotiation
[80]	3GPP TS 23.167	IP Multimedia Subsystem (IMS) emergency sessions
[81]	3GPP TS 24.166	3GPP IP Multimedia Subsystem (IMS) conferencing Management Object (MO)
[82]	3GPP TS 24.391	Unstructured Supplementary Service Data (USSD) using IP Multimedia (IM) Core Network (CN) subsystem (IMS) Management Object (MO)

Ref	Doc Number	Title
[83]	ITU-T Recommendation H.264 (04/2013): :	"ITU-T RecommendaAdvanced video coding for generic audiovisual services"
[84]	ITU-T Recommendation H.265 (04/2013):	High efficiency video coding
[85]	IETF RFC 6228	Session Initiation Protocol (SIP) Response Code for Indication of Terminated Dialog
[86]	IETF RFC 7798	RTP Payload Format for High Efficiency Video Coding (HEVC)
[87]	3GPP TS 24.390	Unstructured Supplementary Service Data (USSD) using IP Multimedia (IM) Core Network (CN) subsystem IMS; Stage 3
[88]	GSMA PRD RCC.15	IMS Device Configuration and Supporting Services
[89]	3GPP TS 26.131	Terminal acoustic characteristics for telephony; Requirements
[90]	3GPP TS 26.132	Speech and video telephony terminal acoustic test specification
[91]	3GPP TS 24.417	Management Object (MO) for Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR) using IP Multimedia (IM) Core Network (CN) subsystem
[92]	3GPP TS 22.011	Service Accessibility

Conventions

The key words "must", "must not", "required", "shall", "shall not", "should", "should not", "recommended", "may", and "optional" in this document are to be interpreted as described in RFC 2119 [1].

2 Feature Set

General

The IMS profile part lists the mandatory and optional capabilities, that are required over the Gm and Ut reference points in this profile. In addition, the mandatory feature SMS over NAS is covered.

Support of general IMS functions

This profile is applicable to a Native IMS Client. Downloadable IMS Clients are out of scope for this profile.

SIP Registration

Registration Options and APN/DNN Usage

The UE must support the following registration options:

1. A single IMS registration for all IMS services,

2. Two separate IMS registrations using different APNs/DNNs (Data Network Name) each supporting a subset of the IMS services.

Note 1: These separate registrations can be to a single converged IMS core network or to two separate IMS core networks.

The number of IMS registrations to be used for the IMS services depends on the RCS VOLTE SINGLE REGISTRATION parameter as defined in Annex A.1.6.2 of GSMA PRD RCC.07 [73]. This parameter is only applicable if at least one of the RCS Messaging services or MSRP based Enriched Calling services is enabled; if none of the RCS Messaging services or MSRP based Enriched Calling services is enabled, the UE uses only the IMS registration via the PDU (Protocol Data Unit) session to the IMS well-known APN/DNN. Table 2.2.1.1-1 illustrates the usage of the parameter for a UE.

RCS VOLTE SINGLE REGISTRATION (see GSMA PRD RCC.07 [73])	UE roaming outside of HPMN? (Note 2)	UE behavior
0	Not relevant	Two IMS registrations
1	Not relevant	Single IMS registration
2	No	Single IMS registration
	Yes	Two IMS registrations

Table 2.2.1.1-1 Summary of parameters controlling IMS Registrations

Note 2: Roaming is taken into consideration due to the fact that MSRP may not be supported in the VPMN (Visited PMN).

The APN/DNNs used for IMS registration are shown in Table 2.2.1.1-2.

REGISTRATION OPTION	APN/DNN to be used
Single IMS Registration	IMS well-known APN/DNN
Two IMS Registrations	IMS well-known APN/DNN
	HOS APN/DNN

Table 2.2.1.1-2 APN/DNN Usage for IMS Registration

P-CSCF Discovery Mechanism

A UE must be able to discover the P-CSCF for the IMS APN/DNN and for the HOS (Home Operator Services) APN/DNN. The P-CSCF discovery mechanism for a registration will depend on the APN/DNN used for that registration as defined in Table 2.2.1.1-2. The UE must discover the P-CSCF(s) (Proxy Call Session Control Function) prior to initial registration as shown in Table 2.2.1.2-1.

APN/DNN Used for Registration	P-CSCF Discovery Mechanism
IMS APN/DNN	PCO (Protocol Configuration Options) returned from 5GC as described in section 4.7.

HOS APN/DNN	RCS Client Configuration (IMS MO – see section 2.4.5 of GSMA PRD RCC.07 [73]).

Table 2.2.1.2-1 P-CSCF Discovery Mechanism

Authentication for SIP Registration

The authentication mechanism used for IMS registration depends on the related APN/DNN (see Table 2.2.1.1-2) and on the "IMS Mode Authentication Type" configuration parameter as defined in Table 2 of GSMA PRD RCC.15 [88] and shown in Table 2.2.1.3-1.

APN/DNN Used for Registration	IMS Mode Authentication Type	Authentication Method Used
IMS APN/DNN	N/A	IMS-AKA – see section 2.2.2.1.
HOS APN/DNN	SIP Digest	SIP Digest – see section 2.2.2.2.
	IMS-AKA (see Note)	IMS-AKA – see section 2.2.2.1.

Table 2.2.1.3-1 Authentication for SIP Registration

Note: Since only a single security association can be set up to a converged IMS core using the procedures of IMS-AKA, the case of "IMS Authentication and Key Agreement (AKA)" over the HOS APN/DNN is only applicable if two separate IMS core networks are used for the two registration case.

IMS user and device identifiers in SIP Registration

The derivation of the IMPU (IMS Public Identity), IMPI (IMS Private User Identity) and +sip.instance identifiers in the SIP REGISTER request, is dependent on the APN/DNN for the registration (see Table 2.2.1.1-2) and authentication mechanism as shown in Table 2.2.1.4-1.

User and device Identifiers	Authentication Mechanism used	IMS APN/DNN	HOS APN/DNN
"+sip.instance" header field parameter	N/A	an IMEI (International Mobile Equipment Identity) URN (see 3GPP TS 23.003 [13] section 13.8) matching the IMEI of the UE	A UUID (Universal Unique Identifier) based on the rule defined in section 2.4.2 of GSMA PRD RCC.07 [73]
IMS Public User Identity (IMPU)	IMS AKA	If an ISIM (IM Services Identity Module) application is present on the UICC, the public user identity in the first (or only) record in the Elementary File in the ISIM (see 3GPP TS 31.103 [14] section 4.2.4). Otherwise, the temporary public user identity derived from the IMSI (3GPP TS 23.003 [13]).	
	SIP Digest	N/A	The IMPU obtained from the configuration parameter

			Public_user_identity_List of the IMS Management Object as defined in section 2.2 of GSMA PRD RCC.15 [88] (see Note).
IMS Private User Identity (IMPI)	IMS AKA	If an ISIM application is present on the UICC, the IMPI in the EFIMPI Elementary File in the ISIM (see section 4.2.2 of 3GPP TS 31.103 [15) must be used. Otherwise, the IMPI derived from the USIM's IMSI as per section 13.3 of 3GPP TS 23.003 [13].	
	SIP Digest	N/A	The IMPI obtained from the IMS Management Object as defined in section 2.2 of GSMA PRD RCC.15 [88]

Table 2.2.1.4-1 Identifiers used in the SIP REGISTER request

Note: The HPMN should ensure that the same IMS Public User Identities are registered via the IMS APN/DNN and HOS APN/DNN for user addressing consistency reasons.

Registration of IMS Services

Table 2.2.1.5-1 describes the APN/DNN to use, for specific IMS Services and for both single registration and two registrations scenarios (see Table 2.2.1.1-2).

The UE registers a service only if the service is activated in the UE by configuration, as described in sections 2.2.1.6, 2.2.1.7 and Annex C.3.

The UE must register MMTel Voice, SMSoIP, MMTel Conversational Video, and MMTEL Call Composer, always via the IMS well-known APN. RCS Messaging services and MSRP based Enriched Calling services, if enabled, are registered as described in Table 2.2.1.5-1.

IMS services supported by UE	APN/DNN to be used for registration for IMS Services	
	Single Registration used for IMS Services	Two Registrations used for IMS Services
MMTEL Voice	IMS APN/DNN	IMS APN/DNN
SMSoIP	IMS APN/DNN	IMS APN/DNN
MMTEL Conversational Video	IMS APN/DNN	IMS APN/DNN
MMTEL Call Composer	IMS APN/DNN	IMS APN/DNN
RCS Messaging services (if enabled)	IMS APN/DNN	HOS APN/DNN
MSRP based Enriched Calling services (if enabled)	IMS APN/DNN	HOS APN/DNN

Table 2.2.1.5-1: Service Registrations on IMS APN/DNN and HOS APN/DNN

SIP Registration Procedures

The UE and the IMS core network must follow the Session Initiated Protocol (SIP) registration procedures defined in 3GPP TS 24.229 [8]. In addition, when the conditions for performing the IMS registration in bullets 2, 3, 4, 5, 6 and 7 in section U.3.1.2 of 3GPP TS 24.229 [8] are fulfilled, then the UE must register with the IMS. Selective Disabling of 3GPP User Equipment Capabilities as defined in 3GPP TS 24.305 [9] is not mandated in this profile, therefore in the case where 3GPP TS 24.305 [9] Managed Object (MO) is not deployed, it is assumed that the IMS is enabled in the terminal and the IMS parameter is set to the value "enabled" as specified in Annex C.3.

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

The home operator can configure the UE with RegRetryBaseTime and RegRetryMaxTime parameters as specified in Annex C.3.

The UE must register IMS services as described in Table 2.2.1.5-1 and this is further clarified in section 2.2.1.7.

If the UE is a Session Continuity UE (SC-UE) (e.g. due to support of SR-VCC (Single Radio Voice Call Continuity) for handover from E-UTRAN connected to EPC as described in Annex A.3 of GSMA PRD IR.92 [2]), then the UE must include the g.3gpp.accesstype media feature tag as specified in section 6.2.2 of 3GPP TS 24.237 [12].

The UE and the IMS core network must support network-initiated de-registration as defined in section 5.4.1.5 of .3GPP TS 24.229 [8].

The UE must subscribe to the registration event package as defined in section 5.1.1.3 of 3GPP TS 24.229 [8].

The UE must include the "+sip.instance" header field parameter (Instance ID) of the contact address as described in Table 2.2.1.4-1.

As stated in 3GPP TS 24.229 [8], the UE must include a user part in the URI of the contact address such that the user part is globally unique and does not reveal any private information.

Note 2: To generate this user part, the UE can use a time-based UUID (Universal Unique Identifier) generated as defined in section 4.2 of IETF RFC 4122 [20].

All IMS public user identities provided in the implicit registration set used for the enabled IMS services by the IMS core network must be alias user identities and must include a tel URI (Uniform Resource Identifier). The public user identity that is assigned to the implicit registration set used for the enabled IMS services, must be used by the UE when registering for the enabled IMS services and is derived as in Table 2.2.1.4-1.

Note 3: According to 3GPP TS 23.228 [15], a public user identity is an alias of another public user identity, if both identities belong to the same implicit registration set, are linked to the same service profile and have the same service data configured for each and every service.

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

Note 4: The "tag" header field parameter can differ in the From header field and in the To header field for the different REGISTER requests

For backwards compatibility the network must support all formats of URIs compliant with 3GPP TS 24.229 [8].

The UE must perform a re-registration prior to the expiry time of the existing registration as described in section 5.1.1.4.1 of 3GPP TS 24.229 [8].

If the UE receives a SIP 305 (Use Proxy) response to a re-registration, then the UE must acquire a P-CSCF different from the currently used P-CSCF and initiate a new initial registration as described in section 5.1.1.4.1 of 3GPP TS 24.229 [8]. If the UE receives a SIP 503 (Service Unavailable) response without a Retry-After header field, the SIP 503 (Service Unavailable) response must be treated as a SIP 500 (Server Internal Error) response (as stated in IETF RFC 3261 [17]) and the UE must initiate a new initial registration as described in section 5.1.1.4.1 of 3GPP TS 24.229 [8]. For the new initial registration, the UE must: -

1. for the IMS APN/DNN, select a different P-CSCF from the P-CSCF list received from the last PCO (if not all of them have been attempted), otherwise the UE must re-establish a new PDU session to the IMS well-known APN/DNN and get a new list of P-CSCFs (as stated in section 4.7) and choose from one of these P-CSCFs, as specified in section 5.1.1.4.1 of 3GPP TS 24.229 [8].
2. for the HOS APN/DNN, select a different P-CSCF from the P-CSCF list received via configuration (if not all of them have been attempted); otherwise the UE must re-attempt an initial registration via the same P-CSCF after the amount of time indicated in the Retry-After header field, has expired.

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

Note 5: The above condition assumes that the UE has IP connectivity when the UE re-attempts an initial registration.

Service Specific Aspects in Registration

For MMTEL Voice/Conversational Video,

- the UE must include the IMS Communication Service Identifier (ICSI) value used to indicate the IMS Multimedia Telephony service, that being urn: urn-7: 3gppservice.ims.icsi.mmtel per 3GPP TS 24.173 [10], using the procedures defined in section 5.1.1.2.1 of 3GPP TS 24.229 [8].

- The UE must also include the “audio” media feature tag, as defined in IETF RFC 3840 [18], in the Contact header field of the SIP REGISTER request, using the procedures defined in 3GPP TS 24.229 [8].
- Unless prohibited by Media_type_restriction_policy, the UE must additionally indicate the capability to handle MMTEL conversational video calls by adding a “video” media feature tag, as defined in IETF RFC 3840 [18], in the Contact header field of the SIP REGISTER request using the procedures of 3GPP TS 24.229 [8].

For SMSoIP:

- the UE must include the feature tag used to indicate SMS over IP service (that being +g.3gpp.smsip as defined in section 5.3.2.2 of 3GPP TS 24.341 [11]) if configured to do so, as described in section 2.4 and Annex C.3.

For MMTEL Call Composer service:

- the UE must include the +g.gsma.callcomposer media feature tag as defined in GSMA PRD RCC.20 [72], if the service is enabled as described in section 2.3.13.

For RCS Messaging services / MSRP based Enriched Calling services:

- the UE must include the feature tags of all enabled services as described in section 2.4.4 of GSMA PRD RCC.07 [73]. The RCS Messaging services are enabled as described in section 2.5 and the MSRP based Enriched Calling services are enabled as described in section 2.6.

Authentication

SIP Authentication via IMS-AKA

The UE and the IMS core network must follow the procedures defined in 3GPP TS 24.229 [8] and 3GPP TS 33.203 [25] for authentication with IMS Authentication and Key Agreement (IMS-AKA), Sec-Agree and IPsec. Support of integrity protection is mandatory for both UE and network. Support of confidentiality protection is optional in the network, considering that lower layer security is available.

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

SIP Authentication via Digest

As described in section 2.2.1.3, the UE must and the IMS core network can, follow the procedures for SIP Digest Authentication for registration via the HOS APN/DNN as specified in section 2.12.1 of GSMA PRD RCC.07 [73]. The digest credentials are retrieved by the UE via remote client configuration as described in section 2.12.1 of GSMA PRD RCC.07 [73].

HTTP Authentication

The UE and IMS core network must support the procedures for USIM based authentication if there is no ISIM present on the Universal Integrated Circuit Card (UICC) defined in Annex E.3.1 of 3GPP TS 23.228 [15] and Annex C.2 of 3GPP TS 24.229 [8]. This includes the support for the P-Associated-URI header to handle barred IMS Public User Identities (IMPUs).

The UE and the IMS core network must support the procedures for authentication at the Ut reference point as specified in 3GPP TS 24.623 [26], and 3GPP TS 33.222 [27].

The UE must and the IMS core network can, support the procedures for authentication at the HTTP Content Server as defined in section 3.2.5.3 of GSMA PRD RCC.07 [73]. If the UE supports the OpenID Connect procedures for the HTTP Content Server as defined in GSMA PRD RCC.07 [73], then the UE must support the user authentication via HTTP embedded EAP-AKA for the authentication with the OpenID Connect authorization endpoint as defined in GSMA PRD RCC.14 [74].

If the UE supports the Generic Authentication Architecture procedures specified in 3GPP TS 24.623 [26], 3GPP TS 33.222 [27] and 3GPP TS 24.109 [28], then the UE must construct the Bootstrapping Server Function (BSF) address as defined in section 16.2 of 3GPP TS 23.003 [13] in all supported modes by both the UE and the network.

Note 1: It is recommended that the UE supports the Generic Authentication Architecture procedures specified in 3GPP TS 24.623 [26], 3GPP TS 33.222 [27] and 3GPP TS 24.109 [28].

The UE must support receiving a HTTP 2xx response to HTTP requests for XCAP and the HTTP Content Server, without being challenged by an HTTP 401 (Unauthorized) response.

Note 2: The above authentication scenario is possible only if the APN/DNN used for HTTP for XCAP and HTTP Content Server traffic (see section 4.5.1) is routed to the home network. The home network is able to authenticate the UE without challenging the UE for Ut authentication.

Note 3: The above authentication scenario is applicable for the HTTP Server if the MMTEL Call Composer service is enabled, and no values are provided for the configuration parameters FT HTTP CS USER and FT HTTP CS PWD (see Annex C.3), and authentication via HTTP embedded AKA or the Generic Authentication Architecture is not available.

Addressing

Public User Identities

The UE and IMS core network must support Public User Identities as defined in section 13.4 of 3GPP TS 23.003 [13], which includes all of the following types of addresses:

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

Note: Further requirements to support Public User Identities in the network are specified in GSMA PRD IR.65 [3].

Local numbers

The UE and IMS core network must support local numbers as defined in Alternative 2 in sections 5.1.2A.1.3 and 5.1.2A.1.5 of 3GPP TS 24.229 [8]. 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 and network must support home-local numbers and geo-local numbers. The UE must set the “phone-context” parameter as defined in sections 5.1.2A.1.5 and 7.2A.10 of 3GPP TS 24.229 [8]:

1. 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.
   Example of phone-context for home-local number: if the home network domain used in SIP REGISTER R-URI is “ims.mnc026.mcc567.3gppnetwork.org” then the “phone context” parameter is set to the same string.
2. for geo-local numbers the UE must set the “phone-context” parameter with additional visited network information when available.
   Example of phone-context for geo-local number: if the visited network has MCC = 234, MNC = 15, and the home network has MCC = 567, MNC = 26, the “phone context” parameter is set to the string “234.15.5gs.ims.mnc026.mcc567.3gppnetwork.org”

Note 1: The UE on E-UTRAN knows the access information and hence the “phone-context” can be set accordingly.

If the local number type is not explicitly indicated by the user in the outgoing voice calls to either geo-local numbers or home-local numbers, then the UE must use the Policy_on_local_numbers parameter as specified in Annex C.3 to determine the local number type. For outgoing SMS messages to local numbers, the UE must use home-local numbers.

Note 2: SMS does not currently specify how to route messages to a VPMN for onward routing to a destination, therefore, outgoing SMS messages cannot be destined to geo-local numbers.

Other addressing related SIP header fields

The UE and IMS core network must support the P-Called-Party-ID header field; the network must use this header field as defined in 3GPP TS 24.229 [8].

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

Call Establishment and Termination

The UE and the IMS core network must follow 3GPP TS 24.229 [8] for establishment and termination of a call.

The UE and the IMS core network must support reliable provisional responses as defined in IETF RFC 3262 [21]. The UE must support reliable SIP 18x policy and procedures as specified in section 5.1.4.2 of 3GPP TS 24.229 [8].

The home operator can configure the UE with Timer_T1, Timer_T2 and Timer_T4 parameter as specified in Annex C.3.

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 [8]. See sections 2.2.4.1 and 2.2.4.2.

If the UE receives an incoming SIP request for a service that is not supported over the used IMS registration, the UE must reject that request with a 488 “Not Acceptable Here” error response.

The usage of preconditions is discussed in section 2.2.5.

If the user rejects an incoming call by invoking User Determined User Busy (UDUB) as described in 3GPP TS 22.030 [22], then the UE must send a SIP 486 (Busy here) response to the network.

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

When the UE sends a CANCEL or a BYE request, the UE must include a Reason header field with a protocol value set to "RELEASE_CAUSE" and follow the procedures specified in sections 5.1.3.1 and 5.1.5 of 3GPP TS 24.229 [8].

If information is available, the UE must insert the P-Access-Network-Info header field into the ACK request acknowledging a 2XX response to the INVITE request, as specified in section 5.1.2A.1.1 of 3GPP TS 24.229 [8].

Considerations for Voice/Video Sessions

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 [10].

Note 1: The means to differentiate between an NG.114 video call and another video service in MMTel is FFS.

The UE and the network must be able to establish a video call directly during session establishment and by adding video to a voice session by sending Session Initiation Protocol (SIP) (re-)INVITE request with a Session Description Protocol (SDP) offer that contains both voice and video media descriptors.

The UE must include the audio and video media feature tags, as defined in IETF RFC 3840 [18], in the Contact header field of the SIP INVITE request, and in the Contact header field of the SIP response to the SIP INVITE request, as specified in 3GPP TS 24.229 [8].

The UE shall indicate the capability to handle video by including a “video” media feature tag in the Contact header of an INVITE request independent of video media being part of the SDP offer or not.

The UE shall indicate the capability to handle video by including a “video” media feature tag in the Contact header of any 18x or 200 responses to an INVITE request independent of video media being part of the SDP answer or not.

If the MMTEL Call Composer service is enabled, then the UE must include the media feature tag for MMTEL Call Composer in the Contact header field of the SIP INVITE request, and in the Contact header field of the SIP response to the SIP INVITE request, as specified in section 2.4.4 of GSMA PRD RCC.20 [72].

If the MMTEL Call Composer service is enabled, then the UE must include in the SIP INVITE request, the header fields for the Call Composer elements as defined in section 2.4.4.2 of GSMA PRD RCC.20 [72].

If the MMTEL Call Composer service is enabled and the UE receives in an incoming SIP INVITE request header fields for Call Composer Elements as defined in section 2.4.4.2 of GSMA PRD RCC.20 [72], then the UE must process the header fields as defined in sections 2.4.4.3 and 2.4.4.5 of GSMA PRD RCC.20 [72].

If the MMTEL Call Composer service is not enabled, then the UE must ignore any Call Composer Elements in the incoming INVITE request.

To guide forking at the remote side, the UE must include a “video” media feature tag in the Accept-Contact header of an INVITE request for a video call, as described in IETF RFC 3840 [18] and RFC 3841 [19].

Note 2: Together with the ICSI value in a media feature tag as specified in section 2.2.4 of GSMA PRD IR.92 [2], will give preference for a conversational video and voice capable device in the forking algorithm.

SDP: negotiating voice/video media streams

The UE must be able to accept a SIP INVITE request without a Session Description Protocol (SDP) offer, and the UE must then include an SDP offer in the first non-failure reliable response to a SIP INVITE request without SDP offer. The SDP offer must contain all codecs (for audio only or for both audio and video) that the UE is currently able and willing to use.

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

Note 2: How the UE determines to be able and willing to use audio only or both audio and video is out of scope of this document.

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

The UE must be able to send a SDP offer and answer with full duplex video media.

An SDP answer may decline the video media by setting the port number of the video media descriptor to zero, accept the video media in full duplex mode by omitting SDP direction attribute or using the “a=sendrecv” SDP attribute, or accept the video media in simplex mode by using the “a=sendonly” or “a=recvonly” SDP attribute.

The video stream in a video call may be changed between simplex or duplex mode, or be made inactive, by sending a re-INVITE request with an SDP offer using the appropriate attribute in the video media descriptor (sendrecv, sendonly, recvonly or inactive).

A video stream in a video call can be removed by sending a SIP re-INVITE request with an SDP offer where the port number of the video media descriptor is set to zero.

If the UE receives an SDP offer with multiple video streams, and the UE can only handle a single video stream then:

- The user may be given the possibility to choose which video stream to accept.
- If the user cannot be given the possibility to choose which video stream to accept, then the UE must accept or decline the main video stream and decline all other video streams. If the UE cannot determine which video stream in the SDP offer is the main video stream, then it must consider the first video stream in the SDP offer that is supported by the UE as the main video stream.

Note 3: A UE can provide information to the user on which video stream to accept by using information provided in an "i=" SDP line, if available, or by other means, and can determine the main video stream from an “a=content:main” SDP attribute or by other means. The main video stream is the video stream taken from the main video source as described in IETF RFC 4796 [66].

Considerations for RCS Messaging Services / MSRP based Enriched Calling Services

The UE must include the relevant ICSI / IARI / Feature tags as described in section 2.6.1 of GSMA PRD RCC.07 [73].

The UE must and the network can, support the negotiation of a MSRP session as described in GSMA PRD RCC.07 [73].

SIP Precondition Considerations

For any RCS services using MSRP, the UE must not use SIP Preconditions.

For MMTEL Voice/Conversational Video sessions, the UE must support the preconditions mechanism as specified in sections 5.1.3.1 and 5.1.4.1 of 3GPP TS 24.229 [8]. If the precondition mechanism is enabled by the Precondition_disabling_policy node in Annex C.3, the UE must use the precondition mechanism. If preconditions are used, and the originating UE receives the selected codec in the SDP of a SIP 18x response, then the UE must include only the same codec with its selected configuration parameters in the SDP of the SIP UPDATE request, used for precondition status update.

The network may disable the use of preconditions in the network as specified in GSMA PRD IR.65 [3].

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

Upon receiving an INVITE request, if the use of preconditions is disabled by the home operator using the Precondition_disabling_policy parameter as specified in Annex C.3 or if preconditions are not supported in the received INVITE request, and the local resources required at the terminating UE are not available, then the terminating UE according to [15], must act as specified in 3GPP TS 24.229 [8] section U.3.1.4.

Early media and announcements

The UE must behave as specified in section 4.7.2.1 of 3GPP TS 24.628 [24]. The UE must support reception of voice and video media associated with one (1) early dialogue.

In addition, the UE must support the P-Early-Media header field as defined in IETF RFC 5009 [23], and must include a P-Early-Media header field with the “supported” parameter to initial INVITE requests it originates as specified in section 5.1.3.1 of 3GPP TS 24.229 [8].

The UE must also maintain an early media authorization state per dialog as described in IETF RFC 5009 [23].

As stated in 3GPP TS 24.628 [24], the UE must render locally generated communication progress information, if:

1. an early dialog exists where a SIP 180 response to the SIP INVITE request was received;
2. no early dialog exists where the last received P-Early-Media header field as described in IETF RFC 5009 [23] contained "sendrecv" or "sendonly"; and
3. in-band information is not received from the network.

For SIP response 181 and 182 to the SIP INVITE, the UE must not locally render tones to indicate diversion or queueing of calls.

The UE must evaluate the above rules again after each subsequent request or response received from the remote party, and when in-band information starts, and when the UE determines the in-band media to have stopped.

Note 1: A SIP request or response received without a P-Early-Media header does not change the early media authorization state for the early dialog in which it was received.

Note 2: In-band information arriving at the UE, will always override locally generated communication progress information as defined in section 4.7.2.1 of 3GPP TS 24.628 [24].

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 [17], section 4.2.7.3 of 3GPP TS 23.228 [15], 3GPP TS 24.229 [8] and section 4.7.2.1 of 3GPP TS 24.628 [24]. Furthermore, the UE should be able to maintain at least forty (40) parallel early dialogs until receiving the final response on one of them and the UE must support receiving media on one of these early dialogs.

If the originating UE needs to release an early dialog, the UE must send a BYE request within the early dialog to be released, in accordance with section 15 of IETF RFC 3261 [17], e.g. when the UE receives the first response that would create an early dialog it cannot maintain, the UE sends a BYE request on that early dialog without saving dialog data.

It is also possible that the network or the terminating UE will need to release an early dialog using the 199 (Early Dialog Terminated) response defined in IETF RFC 6228 [85]. To support this, the originating UE must include the "199" option tag in the Supported header field in the initial INVITE request and must understand a 199 (Early Dialog Terminated) response code and act as specified in section 5.1.3.1 of 3GPP TS 24.229 [8].

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

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

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

1. SIP; cause=200; text="Call completed elsewhere"
2. SIP; cause=603; text="Declined"
3. SIP; cause=600; text=” Busy Everywhere”

for forked calls as defined in 3GPP TS 24.229 [8].

The use of Signalling Compression

The UE must not use SIGCOMP when the initial IMS registration is performed in NG-RAN access as specified in 3GPP TS 24.229 [8].

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

SIP Session Timer

The UE must support and use IETF RFC 4028 [53] as follows:

1. for an initial SIP INVITE request, the UE must include a Supported header with the option tag “timer” and must either insert Session-Expires header field with the delta-seconds portion set to 1800, or must not include the Session-Expires header field in the initial SIP INVITE request;
2. if the UE receives a SIP 422 response to an INVITE request, the UE must follow the procedures of section 7.4 of IETF RFC 4028 [53];
3. it is recommended that the UE does not include the "refresher" parameter in the Session-Expires header field of the SIP INVITE request. If the UE includes the "refresher" parameter in the Session-Expires header field of the SIP INVITE request, the UE must set the "refresher" parameter to “uac”;
4. if a received SIP INVITE request indicates support of the "timer" option tag, and does not contain the Session-Expires header field, the UE must include a Session-Expires header field with the delta-seconds portion set to the greater of 1800 or the value contained in the Min-SE header (if present in the received INVITE) and the

"refresher" parameter with the value "uac" in SIP 2xx response to the SIP INVITE request; and
5. if a received SIP INVITE request indicates support of the "timer" option tag, and contains the Session-Expires header field without "refresher" parameter, the UE must include the "refresher" parameter with the value "uac" in the Session-Expires header field of the SIP 2xx response to the SIP INVITE request, and must set the delta-seconds portion of the Session-Expires header field of the SIP 2xx response to the SIP INVITE request to the value indicated in the delta-seconds portion of the Session-Expires header field of the SIP INVITE request.

Note: The network can choose to influence the session timer negotiation by modifying any of the related header fields or header field parameters within the constraints of IETF RFC 4028 [53].

Capability Discovery

The UE must and the network can support Capability Discovery as defined in section 2.6 of GSMA PRD RCC.07 [73]. If Capability Discovery is supported by the UE and the network, then Capability Discovery is configured via the configuration parameter CAPABILITY DISCOVERY MECHANISM (see Annex C.3). Capability Discovery is enabled if the value of the configuration parameter CAPABILITY DISCOVERY MECHANISM is set to "0" or "1".

In the two registrations case (see Table 2.2.1.1-1), the capability exchange can take place over either registration, and a given capability exchange must advertise only the services applicable to the related registration (see Table 2.2.1.2-1).

Note: The interworking at the NNI between IMS Networks that have deployed different registration options (i.e. single registration versus two registrations) as described in section 2.2.1.1 is out of scope for this document.

Capability discovery is applicable for the following services:

- MMTEL Conversational video,
- MMTEL Call Composer,
- RCS Messaging services,
- MSRP based Enriched Calling services.

The feature tags for OPTIONS based and service tuples for Presence based capability discovery are as described in section 2.6.1 of GSMA PRD RCC.07 [73].

Regardless of whether capability exchange has been enabled, if MMTEL Voice is registered, a Contact header field in a SIP OPTIONS request and in the 200 OK response to a SIP OPTIONS request must include the IMS Communication Service Identifier (ICSI) value of “urn:urn-7:3gpp-service.ims.icsi.mmtel”, as defined in 3GPP TS 24.173 [10].

Unless prohibited by Media_type_restriction_policy, the UE must additionally include the “video” media feature tag, as defined in IETF RFC 3840 [18], in the Contact header field of the SIP OPTIONS request and in the 200 OK response to a SIP OPTIONS request.

If the MMTEL Call Composer service is enabled and registered, then a Contact header field in SIP OPTIONS request and in the 200 OK response to a SIP OPTIONS request must include the media feature tag for MMTEL Call Composer as defined in GSMA PRD RCC.20 [72].

If any RCS Messaging services or MSRP based Enriched Calling services are enabled and registered, then a Contact header in a SIP OPTIONS request and in the 200 OK response to a SIP OPTIONS request must include the related ICSIs / IARIs / media feature tags for those services as described in section 2.6.1.3 of GSMA PRD RCC.07 [73].

Note: In the two registrations case, SIP OPTIONS can be sent/received on either registration, and a given OPTIONS/200 OK response advertises only the services applicable to the related registration (see Table 2.2.1.2-1).

User Agent and Server Headers

The UE must include the User-Agent header in all SIP requests and the Server header in all SIP responses. The UE must include the User-Agent header in HTTP requests for XCAP. The headers must be compiled as defined in section C.4.1 of GSMA PRD RCC.07 [73] including the following amendment:

```
product-list =/ enabler *(LWS enabler)
                     [LWS terminal]
                     [LWS device-type]
                     [LWS mno-customisation]
                     *(LWS list-extension)
```

The rule "enabler" is defined in section C.4.1 of GSMA PRD RCC.07 [73] and is extended as follows:

```
enabler =/ GSMA-PRD
GSMA-PRD = "PRD-" PRD-code SLASH major-version-number
PRD-code = "NG114" / token
major-version-number = 1*DIGIT ; the major version number of the GSMA PRD
; document version
```

The rule "terminal" is defined in section C.4.1 of GSMA PRD RCC.07 [73].

The rules "mno-customization" and "device-type" are defined as an extension to section C.4.1 of GSMA PRD RCC.07 [73] as follows:

```
list-extension =/ device-type / mno-customisation
device-type = "device-type" SLASH device-classification
device-classification = token ; token taken from the
            ; device classification registry defined in IETF RFC 7852 [104]
mno-customisation = "mno-custom" SLASH customisation
customisation = open-market / customised
open-market = "none"
customised = MCC-MNC / free-text
MCC-MNC = MCC MNC
MCC = 3DIGIT ; the E.212 Mobile Country code assigned to the operator
; country
MNC = 3DIGIT ; the E.212 Mobile Network Code assigned to the
; operator, a two digit MNC is padded out to 3 digits
; by inserting a 0 at the beginning
```



```
free-text = token ; a human readable string to identify the
; customisation
```

Examples of User-Agent header constructed according to the rules above:

```
User-Agent: PRD-NG114/1 term-Vendor1/Model1-XXXX device-type/feature-phone
mno-custom/none
User-Agent: PRD-NG114/2 term-Vendor2/Model2-YYYY device-type/smart-phone
mno-custom/235015
User-Agent: PRD-NG114/3 term-Vendor3/Model3-zzzz device-type/smart-phone
mno-custom/ex.telekom
```

Note 1: The User-Agent and Server headers are meant to assist persons in analyzing the network behavior. It is not intended that their presence, content or syntax, influence the network behavior.

Note 2: Within a trust domain, network(s) are expected not to add, remove or modify User-Agent and Server headers. This applies whether a given network element functions as a SIP proxy or Back to Back User Agent (B2BUA).

Supplementary Services for Multimedia Telephony Calls

The UE and the network must support the MMTEL supplementary services defined in section 2.3.1 that are managed as defined in section 2.3.2 and further detailed in their respective subsections of this section. The UE must and the network can support also the MMTEL Call Composer service described in section 2.3.13.

MMTEL Supplementary Services Overview

Supplementary services must be supported as defined in 3GPP TS 24.173 [10], with the constraints described in this section.

The UE and the Telephony Application Server (TAS) must support the supplementary services listed in Table 2.3.1-1. The provisioning of these supplementary services for a subscriber is optional and is an operator decision.

Supplementary Service
Originating Identification Presentation 3GPP TS 24.607 [33]
Terminating Identification Presentation 3GPP TS 24.608 [34]
Originating Identification Restriction 3GPP TS 24.607 [33] (Note 1)
Terminating Identification Restriction 3GPP TS 24.608 [34] (Note 1)
Communication Forwarding Unconditional 3GPP TS 24.604 [39] (Note 1)
Communication Forwarding on not Logged in 3GPP TS 24.604 [39] (Note 1)
Communication Forwarding on Busy 3GPP TS 24.604 [39] (Note 1)
Communication Forwarding on not Reachable 3GPP TS 24.604 [39] (Note 1)
Communication Forwarding on No Reply 3GPP TS 24.604 [39] (Note 1)
Barring of All Incoming Calls 3GPP TS 24.611 [36] (Note 1)

Supplementary Service
Barring of All Outgoing Calls 3GPP TS 24.611 [36] (Note 1)
Barring of Outgoing International Calls 3GPP TS 24.611 [36]
Barring of Outgoing International Calls – ex Home Country 3GPP TS 24.611 [36]
Barring of Outgoing International Calls - When Roaming 3GPP TS 24.611 [36]
Barring of Incoming Calls - When Roaming 3GPP TS 24.611 [36] (Note 1)
Communication Hold 3GPP TS 24.610 [35]
Message Waiting Indication 3GPP TS 24.606 [32] (Note 1)
Communication Waiting 3GPP TS 24.615 [37] (Note 1)
Ad-Hoc Multi Party Conference 3GPP TS 24.605 [40] (Note 1)
Explicit Communication Transfer - Consultative 3GPP TS 24.629 [41] (Note 1)

Table 2.3.1-1 Supplementary services

Note 1: Recommended options are described in sections 2.3.3 – 2.3.12.

The UE must and the network can support the MMTEL Call Composer service. The provisioning of the MMTEL Call Composer service for a subscriber is optional and is an operator decision.

Note 2: Support of other supplementary services is out of scope of this document.

MMTEL Supplementary Service Configuration

For MMTEL supplementary service configuration, the UE and IMS core network must support XCAP at the Ut reference point as defined in 3GPP TS 24.623 [38].

The home operator can configure the UE with the “XCAP Root URI” parameter as specified in Annex C.3 with an XCAP root URI as specified in 3GPP TS 24.623 [38]. If the UE has not been configured with an XCAP root URI, then the UE must construct an XCAP root URI as defined in section 13.9 of 3GPP TS 23.003 [13].

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

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

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

1. if the UE has an ISIM, then the UE must use the public user identity in the first (or only) record in the EFIMPU Elementary File in the ISIM (see section 4.2.4 of 3GPP TS 31.103 [15]) as XUI in further XCAP requests sent until the next successful IMS registration.
2. if the UE has a USIM but not an ISIM, then the UE must use the temporary public user identity derived from the IMSI (see section 13.4B of 3GPP TS 23.003 [13]) as XUI in further XCAP requests sent until the next successful IMS registration.

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

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

The UE must configure settings of one supplementary service only per XCAP request. If the supplementary service to be configured contains a <ruleset> element with multiple <rule> elements as defined in IETF RFC 4745 [43] (e.g. as for Communication Diversion (CDIV), Communication Barring (CB)), then the UE must modify at most one <rule> element of the supplementary service per XCAP request.

The UE must perform HTTP PUT and HTTP DELETE as conditional operations using the If-Match header field as defined in section 7.11 of IETF RFC 4825 [44].

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

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

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

A <rule> element matches a supplementary service if:

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

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

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

Ad-Hoc Multi Party Conference

The UE and the IMS core network must support the procedures defined in 3GPP TS 24.605 [40] and section 5.3.1.3.2 of 3GPP TS 24.147 [42], with the clarifications defined in this sub section.

Note 1: As per section 4.2 of 3GPP TS 24.605 [40], the invocation and operation for conferencing is described in 3GPP TS 24.147 [42].

For conference creation, the UE and the IMS core network must support Three Way Session creation as described in section 5.3.1.3.3 of 3GPP TS 24.147 [42]. The UE must apply option 2b) when inviting the remote user to the conference. If the UE has not been configured with Conf_Factory_URI parameter as specified in Annex C.3, then the UE must construct the “Default Conference Factory URI for MMTel” as specified in section 13.10 of 3GPP TS 23.003 [13].

The UE can and the IMS core network must support the procedures in 3GPP TS 24.605 [40] for subscription to conference state events. The SIP SUBSCRIBE to conference state events must be sent outside the SIP INVITE dialog between the UE and the conference server. If the SUBSCRIBE request outside the existing INVITE dialog is rejected by a SIP 403 (Forbidden) response, the UE must send a SUBSCRIBE request in the existing INVITE dialog, as specified in section 5.3.1.2 of 3GPP Release 12 TS 24.147 [42]. The UE is recommended to send the SUBSCRIBE request in the INVITE dialog until the next initial IMS registration.

To ensure compatibility with UEs compliant with older versions of this specification the IMS core network must support SUBSCRIBE requests received within an INVITE dialog. The IMS core network can support all, or a subset of the elements and attributes specified in IETF RFC 4575 [45]. As a minimum, the IMS core network must support the following elements and attributes:

1. Conference-info: entity
2. Maximum-user-count
3. Users
4. User: entity
5. Display-text
6. Endpoint: entity
7. Status (supported values: connected, disconnected, on-hold)

If the Display-text is not available for a conference participant, the IMS core network should provide the same value for the <User: Entity> and <Display-Text> fields.

Note 2: This behaviour will enable all participants of the conference to be uniquely distinguished from each other.

When inviting other users to a conference, the UE and the IMS core network must support the procedure described in section 5.3.1.5.3 of 3GPP TS 24.147 [42]. The UE must send the SIP REFER method by using the existing dialog for a conference session between the UE and the IMS core network (conference server).

The UE must add the Replaces header to the Refer-to header field in the SIP REFER request, as described in section 5.3.1.5.3 of 3GPP TS 24.147 [42].

Note 3: In Three-Way session creation procedures, the UE has an existing session with the SIP REFER target.

If the UE has subscribed to the conference event package, the UE must wait for the related SIP NOTIFY confirming the last invited participant and its identity before inviting a new participant.

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

For video calls, the UE and the IMS network must support one audio and one video media stream for the conference session.

A video conference is established by the conference creator UE with an SDP offer that contains both voice and video media descriptors in the INVITE to the conference focus, as described in section 5.3.1.3.3 of 3GPP TS 24.147 [42].

A UE that is invited to the conference as described in section 5.3.1.5.3 of 3GPP TS 24.147 [42] may participate in the conference as a voice only participant by using a SDP answer where the port number of the video descriptor is set to zero.

A conference participant in an established video conference may request that its video connection to the conference focus is removed by sending a SIP re-INVITE request with an SDP offer where the port number of the video descriptor is set to zero. The video stream can be re-established by sending a new SIP re-INVITE request with an SDP offer where the port number of the video descriptor is set to a non-zero value.

If the conference participant that established the video conference sends a SIP re-INVITE where the video is removed it is up to network policies if the conference shall be degraded to a voice only conference or if the video conference can continue with the conference originator participating only using voice.

Floor control for conferencing as described in section 8 of 3GPP TS 24.147 [42] is not required.

Consent procedures for list server distribution as described in section 5.3.1.7 of 3GPP TS 24.147 [42] are not required.

The conference server should send notifications for Conference Event Package immediately and, in any case, within 5 seconds.

Communication Waiting

The UE and the IMS core network must support the terminal-based service, as described in 3GPP TS 24.615 [37]. The network-based service is not required. The Communication Waiting (CW) indication as defined in section 4.4.1 of 3GPP TS 24.615 [37] is not required. The UE is required to support Alert-Info, with values as specified in 3GPP TS 24.615 [37]. The UE must provide the ability for the user to activate, deactivate and interrogate the terminal-based service without using UE-to-network signalling (e.g. XCAP/Ut).

Message Waiting Indication

The UE must and the IMS core network can support the Message Waiting Indication (MWI) event package, as defined in 3GPP TS 24.606 [32] and IETF RFC 3842 [46].

Originating Identification Restriction

The UE and the IMS core network must support the SIP procedures in 3GPP TS 24.607 [33]. Service configuration as described in section 4.10 of 3GPP TS 24.607 [33], is not required.

Terminating Identification Restriction

The UE and the IMS core network must support the SIP procedures in 3GPP TS 24.608 [34]. Service configuration, as described in section 4.9 of 3GPP TS 24.608 [34], is not required.

Communication Diversion

The UE and the IMS core network must support the SIP procedures described in 3GPP TS 24.604 [39] for Communication Diversion (CDIV). For CDIV service activation, deactivation, and interrogation (XCAP operations), the UE and the IMS core network must support the XML rules for Call Forwarding Unconditional and the conditions, actions and elements listed in Table 2.3.8-1. However, the operator decides which rules are included in the XML document, e.g. depending on the subscription. The UE must handle a missing rule as defined in section 2.3.2.

The UE and the IMS core network must support the XML rules as described in section 4.9.1 of 3GPP TS 24.604 [39]. The UE must support the History-Info header for identification of diverting parties at the terminating side and for identification of diverted-to parties at the originating side. At the terminating side, a History-Info entry must be used for the identification of the diverting party and that the call has been diverted only if another History-Info entry exists that has assigned the next index in sequence and includes a cause value in a cause-param SIP URI parameter as described in section 4.5.2.6.2 of 3GPP TS 24.604 [39]. At the originating side only History-Info entries including a cause value must be used for presentation of the diverted-to party.

Note 1: The UE can deduce that the received call is a diverted call based on the cause-param values.

Note 2: Support of subscription options and other conditions and actions are out of scope of the document.

Type	Parameter
Rule containing condition	Busy
Rule containing condition	media (supported media types: audio, audio AND video)
Rule containing condition	no-answer
Rule containing	not-registered

Type	Parameter
condition	
Rule containing condition	not-reachable (Note 3)
Rule containing condition	rule-deactivated
Action	Target
Element	NoReplyTimer

Table 2.3.8-1 Supported conditions, actions and elements in CDIV

Note 3: In contrast to the CS domain, IMS networks distinguish between Communication Forwarding on Not Logged-in (CFNL) (CDIV using a rule with the condition not-registered) and Forwarding on Not Reachable (CFNRc) (CDIV using a rule with the condition not-reachable). An operator may choose to not apply CFNL and would therefore not include the rule containing the condition not-registered in the XML document.

If both CFNL and CFNRc are included in the XML document by the operator, then the UE must activate both CFNRc and CFNL to the same target, in order to create compatible user experience with the CS domain.

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

1. Communication diversion supplementary service activation, no-reply-timer or both.
2. If the <cp:ruleset> element is present, and a <NoReplyTimer> element is to be created, the UE must include the <cp:ruleset> element in the HTTP PUT request.
3. For the communication diversion services supported in this PRD, elements of one <rule> element for communication diversion supplementary service only.

Note 4: It is not possible to create a no-reply timer without including the rule set in a document where a rule set already exists since this would create an invalid XML document according to IETF RFC 4825 [44].

If:

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

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

Communication Barring

The UE must support the procedures in 3GPP TS 24.611 [36] as clarified in this section. The IMS core network must support the SIP procedures in 3GPP TS 24.611 [36] and can support the procedures in section 5.3.2.5 of 3GPP TS 24.623 [38]. For service activation, deactivation, and interrogation (XCAP operations), the UE and the IMS core network must support the XML rules for Barring of All Incoming Calls, Barring of All Outgoing Calls and the conditions listed in Table 2.3.9-1. The UE and the IMS core network must support the XML rules as described in section 4.9.1.3 of 3GPP TS 24.611 [36].

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

Condition
Roaming
International
International-exHC
rule-deactivated

Table 2.3.9-1 Supported conditions in CB

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

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

Communication Hold

The UE invoking the HOLD service must not send any media to the other party. When the UE invokes HOLD for a video call the UE shall request both the voice and the video media streams to be held or resumed using the procedures described in section 4.5.2.1 of 3GPP TS 24.610 [35]. The network shall only initiate the procedures for the provisioning of announcement to the held user or use the network option to reduce bandwidth described in section 4.5.2.4 of 3GPP TS 24.610 [35], if both voice and video media streams are held.

When restoring a video media flow after a hold procedure, section 14.3 of 3GPP TS 26.114 [16] must be followed.

Explicit Communication Transfer - Consultative

The UE must and IMS core network can support the procedures for the consultative transfer defined in 3GPP TS 24.629 [41], with the clarifications defined in this sub section.

The UE as a Transferee must support the procedures with 3rd Party Call Control (3PCC) as defined in 3GPP TS 24.628 [24]. The UE procedure without 3PCC is not required.

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

Originating Identification Presentation

The UE and IMS core network must support the SIP procedures in 3GPP TS 24.607 [33].

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

The UE must support the operator's originating party identity determination policy as defined in section 4.5.2.12 of 3GPP TS 24.607 [33] also the UE must support being configured according to the "FromPreferred" parameter as specified in Annex C.3.

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

MMTEL Call Composer Service

The UE must and the network can support the MMTEL Call Composer service.

The MMTEL Call Composer service is enabled if the value of the configuration parameter COMPOSER AUTH (see Annex C.3) is set to "2" or "3".

The UE must allow the user to set the MMTEL Call Composer elements in a SIP INVITE request if the peer UE also supports the MMTEL Call Composer Service. The support of a Call Composer at the peer UE can be determined via:

10. The use of Capability Discovery, if supported and enabled as described in section 2.2.10, otherwise
11. The inclusion of the MMTEL Call Composer media feature tag in the Contact header field in the last INVITE request or response or SIP OPTIONS request or response received from the peer UE.

Note: An application server can choose to remove the Call Composer elements as specified in section 2.4.4.2 of GSMA PRD RCC.20 [72].

SMS

The UE must and the network can support both SMS over IP and SMS over NAS signalling methods. However, the network must support at least one of the above methods, i.e. either SMS over IP or SMS over NAS signalling method.

The UE must support the functionality as specified according to section 7.2c of 3GPP TS 23.221 [65].

The home operator must pre-configure and can re-configure the UE with the “SMS_Over_IP_Networks_Indication” parameter as specified in Annex C.3 that determines to use either SMS over IP or SMS over NAS signalling.

The UE must only include the +g.3gpp.smsip feature tag as defined in section 5.3.2.2 of 3GPP TS 24.341 [11] in a SIP REGISTER request when “SMS_Over_IP_Networks_Indication” is set to the value of 1.

SMS over IP

The UE must implement the roles of an SM-over-IP sender and an SM-over-IP receiver, according to sections 5.3.1 and 5.3.2 of 3GPP TS 24.341 [11].

The status report capabilities, delivery reports, and notification of having memory available, according to sections 5.3.1.3, 5.3.2.4 and 5.3.2.5 of 3GPP TS 24.341 [11] must be supported by the UE and the IMS core network.

If the network supports SMS over IP, 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 [11], and the following functions:

1. answering of routing information query and obtaining the routing information according to the procedures in section 5.3.3.3 of 3GPP TS 24.341 [11]; and
2. transport layer interworking according to section 5.3.3.4 of 3GPP TS 24.341 [11].

2.1.1 SMS over NAS

The UE must and, if the network supports SMS over NAS signalling, the network must support the necessary procedures as specified in 3GPP TS 23.221 [65] and 3GPP TS 24.501 [7].

RCS Messaging Services

The UE must support RCS Messaging services.

The network can support RCS Messaging services.

If supported and enabled, the services must be registered as defined in section 2.2.1.5.

The RCS Messaging services are enabled as defined in section 2.3 of GSMA PRD RCC.07 [73], and via the configuration parameters defined in Annex A.1.3 of GSMA PRD RCC.07 [73].

MSRP based Enriched Calling Services

The UE may and the network can support MSRP based Enriched Calling services. If supported and enabled, the services must be registered as defined in section 2.2.1.5.

The MSRP based Enriched Calling services are enabled as defined in section 2.3 of GSMA PRD RCC.07 [73], and via the configuration parameters defined in section 2.1.2 of GSMA PRD RCC.20 [72].

3 Media

General

This section endorses a set of media capabilities specified in 3GPP TS 26.114 [16]. The section describes the needed SDP support in UEs and in the IMS core network and it describes the necessary media capabilities both for UEs and for entities in the IMS core network that terminate the user plane. Examples of entities in the IMS core network that terminate the user plane are the Media Resource Function Processor (MRFP) and the Media Gateway (MGW).

Multimedia: SDP Considerations

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

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

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

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

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

Audio

Codecs

The UE must support and offer all speech codecs mandated for MTSI clients in terminal, including mandatory parts of the detailed per-codec requirements, as described in sections 5.2.1 and 7.5.2.1 of 3GPP TS 26.114 [16].

In particular, the UE must support the EVS codec as described in 3GPP TS 26.114 [16], 3GPP TS 26.441 [56], 3GPP TS 26.445 [58], 3GPP TS 26.446 [59], 3GPP TS 26.447 [60], 3GPP TS 26.449 [61], 3GPP TS 26.450 [62], 3GPP TS 26.451 [63] and either 3GPP TS 26.442 [57] or 3GPP TS 26.443 [64] or 3GPP Release 16 TS 26.452.

The UE must support the handling of CMR within RTP payload as specified in section 7.5.2.1.2.2 of 3GPP TS 26.114 [16].

Entities in the IMS core network that terminate the user plane supporting speech communication and supporting TFO and/or TrFO must support all speech codecs mandated for MTSI media gateways, including mandatory parts of the detailed per-codec requirements as described in clause 12.3.1 of 3GPP TS 26.114 [16].

Entities in the IMS core network that terminate the user plane supporting super-wideband speech communication must support:

- EVS speech codec as described in 3GPP TS 26.441 [56], 3GPP TS 26.445 [58], 3GPP TS 26.447 [60], 3GPP TS 26.449 [61], 3GPP TS 26.450 [62] and 3GPP TS 26.451 [63] and either 3GPP TS 26.442 [57], 3GPP TS 26.443 [64], or 3GPP Release 16 TS 26.452.

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

Note: Restrictions in codec mode changes are required only for transcoder-free interworking with a legacy device.

Speech Media: SDP Considerations

General

The SDP offer/answer for voice media must be formatted as specified in section 6.2.2 of 3GPP TS 26.114 [16], with the restrictions included in the present document.

The UE must include in an initial SDP offer at least:

1. one EVS payload type with one of the configurations supporting super-wideband speech as defined in section 3.2.2.3 of this document
2. one AMR-WB payload type with no mode-set specified as defined in table 6.1 of 3GPP TS 26.114 [16].
3. one AMR payload type with no mode-set specified as defined in table 6.1 of 3GPP TS 26.114 [16].

The codec preference order must be as specified in sections 5.2.1.5 and 5.2.1.6 of 3GPP TS 26.114 [16].

Note: The originating and terminating networks can modify the SDP offer for voice media.

AMR and AMR-WB

The UE must set the b=AS to match the highest codec mode for the offer (maximum codec bit rate if no mode set is included).

The UE, upon receiving an initial SDP offer containing a payload description for AMR with no mode-set included, and accepting the payload description with no mode-set, must include into the SDP answer the value assigned to the RateSet parameter for AMR as specified in Annex C.3. It is recommended to set the RateSet parameter for AMR to 0,2,4,7 (i.e. mode set=0,2,4,7 included in the SDP answer).

The UE, upon receiving an initial SDP offer containing a payload description for AMR-WB with no mode-set included, and accepting the payload description with no mode-set, must include into the SDP answer the value assigned to the RateSet parameter for AMR-WB as specified in Annex C.3. It is recommended to set the RateSet parameter for AMR-WB to "undefined" (i.e. no mode-set included). A UE that intends to use AMR-WB 12.65 as highest mode must have the RateSet parameter set to "0,1,2" and include mode-set=0,1,2 in the SDP answer.

The SDP answer for AMR with no mode-set included, must be interpreted by the UE as all eight AMR modes can be used.

The SDP answer for AMR-WB with no mode-set included must be interpreted by the UE as all nine AMR-WB modes can be used.

The UE must set the b=AS to match the highest codec mode for the answer (maximum codec bit rate if no mode set is included).

EVS

The UE that sends the SDP offer for voice media must include in this SDP offer at least one EVS payload type with one of the following EVS configurations:

1. EVS Configuration A1: br=5.9-13.2; bw=nb-swb.
2. EVS Configuration A2: br=5.9-24.4; bw=nb-swb.
3. EVS Configuration B0: br=13.2; bw=swb.
4. EVS Configuration B1: br=9.6-13.2; bw=swb.
5. EVS Configuration B2: br=9.6-24.4; bw=swb.

The UE may also include in this SDP offer, ch-aw-recv=*x* with *x* set to a value out of the set {-1,0,2,3,5,7}. The UE must support being configured according to the "ICM/INIT_PARTIAL_REDUNDANCY_OFFSET_RECV" parameter as specified in Annex C.3. If "ICM/INIT_PARTIAL_REDUNDANCY_OFFSET_RECV" parameter is undefined, then the UE must not include ch-aw-recv into the SDP offer. SDP parameters other than br, bw, max-red and ch-aw-recv must not be included in a media format description associated with the EVS codec within the initial SDP offer (for a list of SDP parameters see table 6.2a in 3GPP TS 26.114 [16]).

Note 1: If ch-aw-recv is not included in the SDP, this is identical to include ch-aw-recv=0, as specified in 3GPP TS 26.445 [58].

The configuration of the EVS payload type to be included first in the initial SDP offer for EVS is defined by the EVS/Br and EVS/Bw parameters as specified in Annex C.3, which must be configured to one of the five above EVS Configurations.

The UE that sends the initial SDP offer should also include in this initial SDP offer one EVS payload type with audio bandwidth range up to super-wideband and with no restrictions on bitrate range and no restriction on mode-set. If this EVS payload type is not included, then:

1. an initial SDP offer with EVS configuration B0 or B1 listed first must also include a second payload type with EVS configuration A1; and
2. an initial SDP offer with EVS configuration B2 listed first must also include a second payload type with EVS configuration A2.

The UE must support all SDP parameters applicable to EVS that can be received in a SDP offer as specified in 3GPP TS 26.114 [16] and 3GPP TS 26.445 [58].

A payload type in the received SDP offer is considered to match the EVS configuration ‘X’ if the values that the payload type indicates for the ‘br’ and ‘bw’ parameters are exactly as specified above for configuration ‘X’. The inclusion of additional parameters and the values to which those parameters are set, has no bearing on whether this inclusion matches the standard configuration ‘X’.

A UE must answer according to both the received SDP Offer and the UE's EVS configuration (i.e. EVS/Br and EVS/Bw parameters as specified in see Annex C.3) as described in Table 3.2.2.3-1 below:

	EVS configuration of the UE that received the SDP Offer				
Received SDP Offer for EVS (preferred payload type listed first)	**A1**	**A2**	**B0**	**B1**	**B2**
A1	A1	A1	A1	A1	A1
A2	A1	A2	A1	A1	A2
B0	B0	B0	B0	B0	B0
B1	A1	A1	B1	B1	B1
B2	A1	A2	B1	B1	B2

Table 3.2.2.3-1 SDP content to be included in an SDP answer for a super-wideband call

Note 2: This table applies only to received SDP Offers compliant with this PRD, i.e. including A1, if B0 or B1 are listed first and including A2, if B2 is listed first.

If the selected EVS configuration is A1, B0 or B1 then "mode set = 0,1,2" must be included in the SDP answer.

The UE must add only SDP parameters that are applicable to EVS in the SDP answer that are already present in the corresponding received and accepted SDP Offer. If SDP parameters applicable to EVS, are included in the accepted SDP offer, then the UE must handle these parameters as specified in 3GPP TS 26.445 [58].

If the SDP parameter ch-aw-recv is present in the corresponding received and accepted SDP offer, then the SDP parameter ch-aw-recv must be included in the SDP answer with the same value as received.

Default values as specified in 3GPP TS 26.445 [58] apply for all other SDP parameters applicable to EVS that are not included in the SDP Answer

Speech Payload Format Considerations

The UE and the entities in the IMS core network that terminate the user plane, must support and use the speech payload formats in the way described in section 7.4.2 of 3GPP TS 26.114 [16], for the corresponding codecs described in section 3.2.1, with the following exceptions and extensions:

- The UE and the entities in the IMS core network that terminate the user plane must support the bandwidth-efficient and the octet-aligned formats of the AMR and AMR-WB payload formats.
- The UE and the entities in the IMS core network that terminate the user plane must send the number of speech frames, or fewer, encapsulated in each RTP packet, as requested by the other end using the ptime SDP attribute.
- The UE and the entities in the IMS core network that terminate the user plane must set the ptime attribute value to receive one speech frame encapsulated in each RTP packet, but must accept any number of frames per RTP packet, up to the maximum limit of 12 speech frames per RTP packet.

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

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

The UE and the entities in the IMS core network that terminate the user plane must be able to receive and handle redundant speech frames in RTP packets, as described in section 9.2.3 of 3GPP TS 26.114 [16].

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

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

Note 3: As the speech media uses the RTP AVP profile as specified in section 3.5.1 and the referenced section 7.2 of 3GPP TS 26.114 [16], the adaptation using RTCP may be too slow and therefore unsuitable.

Jitter Buffer Management Considerations

The minimum performance requirements for jitter buffer management of voice media, as described in section 8.2 of 3GPP TS 26.114 [16], must be met.

Front End Handling

UEs used for IMS voice services must conform to the minimum performance requirements on the acoustic characteristics specified in 3GPP TS 26.131 [89]. The codec modes and source control rate operation (DTX) settings must be as specified in 3GPP TS 26.132 [90].

Video

Codecs

The entities in the IMS core network that terminate the user plane must support ITU-T Recommendation H.264 [83] Constrained Baseline Profile (CBP) Level 1.2 implemented as specified in section 5.2.2 of 3GPP TS 26.114 [16].

The UE must support ITU-T Recommendation H.264 [83] Constrained High Profile (CHP) Level 3.1 as specified in section 5.2.2 of 3GPP TS 26.114 [16].

The UE must support ITU-T Recommendation H.265 [84] Main Profile, Main Tier Level 3.1 as specified in section 5.2.2 of 3GPP TS 26.114 [16].

For backward compatibility, the UE must also support ITU-T Recommendation H.264 [83] Constrained Baseline Profile (CBP) Level 3.1 as specified in section 5.2.2 of 3GPP TS 26.114 [16], and when H.264 [83] (Advanced Video Coding (AVC)) CHP Level 3.1 is offered, then H.264 [83] CBP Level 3.1 must also be offered.

Note 1: For codec Levels allowing a very high maximum bit rate (like Level 3.1 for H.264 [83] or H.265 [84]) it is expected that the codec is used with a maximum bit rate much lower than the maximum bit rate allowed by the Level and operated with rate adaptation to adjust the bitrate to network transmission capabilities (rate adaptation mechanisms are described in section 3.5.3).

When sending the currently active H.264 [83] or H.265 [84] parameter set (Sequence Parameter Sets (SPS), Picture Parameter Sets (PPS) and Video Parameter Set (VPS)) in the RTP media stream, the UE and the entities in the network terminating the media plane must repeat the parameter set at multiple occasions with appropriate spacing with regards to the channel loss characteristics.

The VPS must be included in the compliant H.265 (HEVC) bitstream as specified in section 5.2.2 of 3GPP TS 26.114 [16] and section 1.1.2 of IETF RFC 7798 [86].

Note 2: The UE may implement different algorithms on how to repeat the parameter set in the RTP media. Since packet loss is expected to be higher at the start of transmission, it is recommended that the parameters are repeated in a progressively slowing rate.

Note 3: SPS, PPS and VPS are three types of parameter set structures (syntax structures). SPS and PPS can be used in the H.264 coded video, as described in sections 3, 7.3, and 7.4 of ITU-T Recommendation H.264 [83], while SPS, PPS and VPS can be used in the H.265 [84] coded video, as described in the sections 3, 7.3, and 7.4 of ITU-T Recommendation H.265 [84].

Change of video resolution mid-stream by transmission of new parameter sets in the RTP media stream must be supported, as long as those parameter sets conform to the media stream configuration negotiated in SDP.

Note 4: Change of video resolution can occur if a MRFP switches in for example a H.264 [83] CBP Level 3.1 source to a UE supporting H.264 [83] CBP Level 1.2. The MRFP may need to trigger the sending of the parameter set by sending a RTCP Full Intra Request (FIR) message to the new source.

When receiving a Full Intra Request (FIR) for a video media stream, the UE and the entities terminating the media plane in the network must send the currently active parameter sets followed by a decoder refresh point.

Note 5: A decoder refresh point can be sent using an Instantaneous Decoding Refresh (IDR) picture in H.264 [83] or an Intra Random Access Point (IRAP) picture having nal_unit_type equal to IDR_N_LP in H.265/HEVC [84].

Video Media: SDP Considerations

General

The Session Description Protocol (SDP) offer/answer for video media must be formatted as specified in section 6.2.3 of 3GPP TS 26.114 [16], along with the restrictions included in the present document.

Unless preconfigured otherwise by the home operator with the Media_type_restriction_policy parameter as specified in Annex C.3 and when offering video media that is not already part of the session, regardless if it is at the start of the session or at some later point in time, the UE must include in the SDP offer at least:

1. One H.265 (HEVC) Main Profile, Main Tier, Level 3.1 payload type as defined in sections 5.2.2 and 7.4.3 of 3GPP TS 26.114 [16].
2. One H.264 (AVC) Constrained High Profile Level 3.1 payload type as defined in sections 5.2.2 and 7.4.3 of 3GPP TS 26.114 [16].
3. One H.264 (AVC) Constrained Baseline Profile Level 3.1 payload type as defined in sections 5.2.2 and 7.4.3 of 3GPP TS 26.114 [16].

The payload type preference order on the SDP m= line must be as specified by the numbered list above.

Coordination of Video Orientation (CVO) as specified in 3GPP TS 26.114 [16] shall be supported with two (2) bits granularity by the UE and the entities in the IMS core network which terminate the user plane. The support for CVO shall be included in SDP offer and SDP answer as specified in section 6.2.3 of 3GPP TS 26.114 [16].

H.265

If an asymmetric video stream for H.265 (HEVC) is supported, the parameter ‘max-recv-level-id’ should be included in the SDP offer and SDP answer, and the level offered with it must be higher than the default level offered with the ‘level-id’ parameter in the SDP offer/answer respectively, as specified in section 7.1 of IETF RFC 7798 [86] and section 6.2.3 of 3GPP TS 26.114 [16].

Video Payload Format Considerations

Section 7.4.3 of 3GPP TS 26.114 [16] applies both for the UE and the entities in the IMS core network that terminate the user plane.

If the use of CVO has been accepted in the SDP negotiation as specified in section 3.3.2.1 then the CVO with the negotiated granularity shall be used by the UE and the entities in the IMS core network that terminate the user plane as specified in section 7.4.5 of in 3GPP TS 26.114 [16].

DTMF events

The UE and the IMS core network must support DTMF events as defined in Annex G of 3GPP TS 26.114 [16].

If the UE receives an SDP offer with no telephone-event codec included, then the UE must not reject the SDP offer for this reason and the UE must not send DTMF events using the telephone-event codec for the negotiated session.

Note: Transport of DTMF events from the UE using the telephone-event codec during a session is impossible unless the telephone-event payload type has been negotiated.

RTP: Profile and SDP Considerations

RTP Profile

Section 7.2 of 3GPP TS 26.114 [16] applies to both the UE and entities in the IMS core network that terminate the user plane.

RTP Data Transport

The UE and the entities in the IMS core network that terminate the user plane must use RTP over UDP as specified in section 7.2 of 3GPP TS 26.114 [16], and use symmetric RTP as defined in IETF RFC 4961 [77] and section 7.2 of 3GPP TS 26.114 [16].

The UE and the entities terminating the media plane in the network supporting IMS conversational video services are recommended to support RTP retransmission for video media, as specified in sections 6.2.3.5, 7.4.6, and 9.3 of 3GPP TS 26.114 [16]. If RTP retransmission is supported, the UE and the entities terminating the media plane in the network:

1. Must keep also the RTP retransmission payload type in SDP answer, if it was included in the corresponding SDP offer for the accepted video media.
2. Must set the RTP retransmission “rtx-time” SDP parameter to 800 in generated SDP offers and answers if RTP retransmission payload type is included in the SDP but must accept the reception of other values and be able to handle them.

RTCP Usage

The RTP implementation must include an RTP Control Protocol (RTCP) implementation according to section 7.3.1 of 3GPP TS 26.114 [16].

The UE and the entities in the IMS core network that terminate the user plane must use symmetric RTCP as defined in IETF RFC 4961 [77], and section 7.3.1 of 3GPP TS 26.114 [16].

The bandwidth for RTCP traffic must be described using the "RS" and "RR" SDP bandwidth modifiers at media level, as specified by IETF RFC 3556 [78], and section 7.3.1 of 3GPP TS 26.114 [16]. Therefore, a UE must include the "b=RS:" and "b=RR:" fields in SDP, and a UE and the entities in the IMS core network that terminate the user plane must be able to interpret them. If the “b=RS:” field or “b=RR:” field or both these fields are not included in a received SDP (offer or answer), then the UE must use the recommended default value for the missing field(s) as defined in IETF RFC 3556 [78].

RTCP is controlled on a per session basis by the SDP offer/answer exchange as defined in section 7.3 of 3GPP TS 26.114 [16] with the following clarifications:

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

The UE and the entities in the IMS core network that terminate the user plane must send RTCP packets when media (including early media) is sent or received. Once an RTCP packet is sent according to received SDP of a SIP dialog, RTCP packets must be sent by UEs and entities in the IMS core network that terminate the user plane according to the received SDP of the SIP dialog for the remaining duration of the SIP dialog. For uni-directional media (e.g. early media or during call hold), RTCP packets must always be sent by both UEs and entities in the IMS core network that terminate the user plane. If multiple early dialogs are created due to forking (see section 2.2.7), the UE must send the RTCP packets according to received SDP answers of those early dialogs for which the IP address and port received in the SDP match the IP address and port of received media.

Note 1: The RTCP is based on the periodic transmission of control packets to all participants in the session, as described in IETF RFC 3550 [76]. In the context of this document, the primary uses of RTCP are voice and video quality monitoring, and to provide link aliveness information while the media are on hold. The latter implies that the RTCP transmission must continue when the media are on hold.

The UE and the entities in the IMS core network that terminate the user plane must set the sending frequency of control packets to a value calculated from the values of "RS" and "RR"

SDP bandwidth modifiers according to rules and procedures in IETF RFC 3550 [76]. The UE must set the "RS" and "RR" SDP bandwidth modifiers such that RTCP packets are sent to the UE at least once every 5 seconds, in order to allow a sufficiently tight inactivity detection.

The UE and the entities in the IMS core network that terminate the user plane must support the transmission of RTCP packets formatted according to the rules in IETF RFC 3550 [76] and with the following clarifications below.

The UE and the entities in the IMS core network that terminate the user plane may use reduced size RTCP packets if agreed in SDP offer/answer as specified in section 7.3.6 of 3GPP TS 26.114 [16]. Otherwise, the RTCP compound packet format must be used. When sent, the compound packet must include one report packet and one Source Description (SDES) packet. When no RTP packets have been sent in the last two reporting intervals, the UE and the entities in the IMS core network that terminate the user plane should send a Receiver Report (RR). Receiving of a Sender Report (SR) instead of an RR must be handled and accepted as valid by the UE and the entities in the IMS core network that terminate the user plane.

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

1. For SR and RR RTCP packets:
2. Version 2 must be used; and
3. Padding must not be used (and therefore padding bit must not be set).
4. For SDES RTCP packets:
5. version and Padding as described for SR packet must be used;
6. the SDES item CNAME must be included in one packet; and
7. other SDES items should not be used.

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

To be forward compatible and interwork with legacy equipment, the UE and the entities in the IMS core network that terminate the user plane must be able to receive all types of RTCP packets, according to the rules specified in IETF RFC 3550 [76].

The UE and the entities in the IMS core network that terminate the user plane must support the use of the Full Intra Request (FIR) of Codec-Control Messages (CCM) as specified in section 7.3.3 of 3GPP TS 26.114 [16].

Note 3: The FIR message is used to request the video codec to send a decoder refresh point as soon as possible. This is needed for example when a video MRF switches the video signal to be sent to conference participants.

For interoperability and forward-compatibility reasons, the UE must support the reception of the Temporary Maximum Media Bit-rate Request (TMMBR) from the remote end and respond with the Temporary Maximum Media Bit-rate Notification (TMMBN) as described in section 10.3 of 3GPP TS 26.114 [16].

The UE and the entities in the IMS core network that terminate the user plane must support at least one adaptation trigger as specified in section 10.3.3 of 3GPP TS 26.114 [16]. The UE and the entities in the IMS core network that terminate the user plane must detect whether the bitrate needs to be reduced or can be increased and it must be able to send a corresponding TMMBR message as specified in sections 10.3.6 and 10.3.7 of 3GPP TS 26.114 [16].

The UE and the entities in the IMS core network that terminate the user plane must react to a received TMMBR message by responding with a TMMBN message and either reducing the bitrate as described in 10.3.4 of 3GPP TS 26.114 [16] or increasing the bitrate as described in 10.3.5 of 3GPP TS 26.114 [16].

The UE must support and use the sending and reception of the AVPF messages NACK and Picture Loss Indication (PLI) as described in section 9.3 of 3GPP TS 26.114 [16]. The entities in the IMS core network that terminate the user plane must support reception of the AVPF messages NACK and Picture Loss Indication (PLI) but are not required to take any action. For the case where no action is taken, forwarding of the messages must be supported.

Note 4: If core network nodes terminate AVPF messages without action rather than forward the messages, it prevents other entities (including the remote UE) to be aware of the messages and thus take any necessary corrective action.

The UE and the entities in the IMS core network that terminate the user plane and that support RTP retransmission must also support the use of NACK as input to RTP retransmission, as described in section 9.3 of 3GPP TS 26.114 [16].

SDP Offer Considerations

The SDP Capability Negotiation framework described in IETF RFC 5939 [82] must not be used in the SDP offer by the UE and the IMS core network when the AVP profile is used.

SDP Answer Considerations

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

Note 1: In section 6.2.1a of 3GPP TS 26.114 [16], 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.

If the initial SDP offer does not use the SDP Capability Negotiation 82] and if the UE receives either:

1. A response with an SDP answer where the video media component has been rejected and the Contact header field in the response does not contain a g.3gpp.icsi-ref feature tag indicating IMS Multimedia Telephony Service; or

2. A SIP 488 or 606 failure response with an SDP body indicating that only AVP is supported for video media.

Then the UE shall send a new SDP offer with AVP as transport for video as described in section 6.2.1a.2 of 3GPP TS 26.114 [16].

Note 2: If an SDP answer, indicating that only the voice media was accepted is received in a reliable SIP 183 session progress response, the new SDP offer shall be sent in the associated PRACK request, or in a subsequent UPDATE request. If an SDP answer, indicating that only the voice media was accepted is received in a 200 OK response, the new SDP offer shall be sent in a subsequent re-INVITE request.

Note 3: A UE supporting IMS Multimedia Telephony Service supports AVPF for video as described in section 6.2.3 of 3GPP TS 26.114 [16]. The rejection of the video media from such a device is due to the device not supporting video, the remote user rejecting the video media or network policy prohibits video.

Note 4: The use of SDP Capability Negotiation to offer the AVPF profile for video is optional in 3GPP. When a NG.114 device not supporting SDP CapNeg receives an offer using SDP CapNeg offering AVPF for video then the AVP offered in the media line will be used for the video media. This may cause a bad service experience for example when participating in a video conference or when an Inter-RAT handover occurs. The UE can upgrade the video session to use AVPF and rtcp-fb messages to ensure a good user experience.

SDP Bandwidth Negotiation

The UE and network must use the b=AS parameter in SDP offers and answers for bandwidth negotiation as defined in section 6.2.5 of 3GPP TS 26.114 [16].

IMS MSRP

For RCS Messaging services and MSRP based Enriched Calling services, the UE and the network must support MSRP as profiled in section 2.7 of GSMA PRD RCC.07 [73].

4 Radio and Packet Core Feature Set

General

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

Robust Header Compression

The UE and the network must support Robust Header Compression (RoHC) as specified in 3GPP TS 38.323 [54], IETF RFC 3095 [47] and IETF RFC 4815 [48]. The UE and network must be able to apply the compression to packets that are carried over the radio bearer dedicated for the voice media. At minimum, the UE and network must support "RTP/UDP/IP" profile (0x0001) to compress RTP packets and "UDP/IP" profile (0x0002) to compress RTCP packets. The UE and network must support these profiles for both IPv4 and IPv6.

NR Radio Capabilities

Radio Bearers

The UE must support 3GPP TS 38.331 [31].

One AM Data Radio Bearer (DRB) is utilized for an 5G System (5GS) QoS flow with 5G Quality of Service Flow Identifier (5QI) = 5 and another AM DRB for QoS flow with 5QI = 6/7/8/9. UM DRB is utilized for QoS flow with 5QI = 1 and for 5QI = 2. QoS flow usage is described in section 4.5.

A QoS Flow with a 5QI-value of 1 or 2 is not allowed to be mapped on the same DRB as any other QoS Flow.

The NG-RAN must set the discardTimer for the DRB with 5QI=5 to the value infinity.

DRX Mode of Operation

In order to maximize lifetime of the UE battery, the UE and the network must support NR Discontinuous Reception (DRX) method as specified in 3GPP TS 38.300 [49] and 3GPP TS 38.321 [50].

RLC configurations

Radio Link Control (RLC) entities must be configured to perform data transfer in the following modes as specified in 3GPP TS 38.322 [51]:

1. Unacknowledged Mode (UM) for QoS flows with 5QI = 1 and 5QI =2
2. Acknowledged Mode (AM) for QoS flow with 5QI = 5
3. Acknowledged Mode (AM) for QoS flow with 5QI = 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).

QoS flow usage is described in section 4.5.

GBR and NGBR Services, GBR Monitoring Function

Voice is one of the NR services that require a GBR radio bearer as described in 3GPP TS 23.501 [4]. The GBR radio bearer for voice is realized via dedicated network resources that are allocated by an admission control function in the gNodeB at bearer establishment. Reports from the UE, including buffer status and measurements of UE’s radio environment, are required to enable the scheduling of the GBR bearer as described in 3GPP TS 38.300 [49]. In UL it is the UE’s responsibility to comply with GBR requirements.

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

Unified Access Control

The UE must support Unified Access Control as specified in 3GPP TS 38.331 [31], 3GPP TS 24.501 [7], and 3GPP TS 24.173 [10].

Registration procedure in 5GC

The UE and the network must support the registration procedure as specified in section 5.5 of 3GPP TS 24.501 [7] and section 5.15.5.2.1 of 3GPP TS 23.501 [4].

QoS Flow Management

QoS Flow Considerations for Signalling: SIP and HTTP

SIP Signalling via IMS APN/DNN

The IMS application in the UE must use the IMS well-known APN/DNN as defined in GSMA PRD NG.113 [55]. The UE must prevent non-IMS applications from using the IMS well-known APN/DNN.

If procedures in section 2.2.1 require the UE to register with IMS, then the UE must establish a PDU session to the IMS well-known APN/DNN using the S-NSSAI as determined in section 4.9; the APN/DNN Operator Identifier must not be included by the UE.

Note 1: PDU session establishment can be caused by a SIP registration request. Sending a SIP registration request per Note1 in section 2.2.1.6 can cause PDU session establishment even if the IMS voice over PS Session indicator indicates that IMS voice over PS session is not supported.

A QoS flow associated with the default QoS rule must be created by the network when the UE creates the PDU session to the IMS well-known APN/DNN, as defined in 3GPP specifications. A standardised 5QI value of five (5) must be used for the QoS flow associated with the default QoS rule. This QoS flow is used for IMS SIP signalling.

A QoS flow associated with the default QoS rule must be created by the network when the UE creates the PDU session for emergency services, as defined in 3GPP specifications. A standardised 5QI value of five (5) and an ARP value that is reserved for emergency services must be used for the QoS flow associated with the default QoS rule. The default QoS flow is used for IMS SIP signalling.

The UE must and the network can support Back-off timer value IE in PDU SESSION ESTABLISHMENT REJECT and the UE must start the back-off timer according to the value indicated by the Back-off timer value IE as specified in 3GPP TS 24.501 [7].

Note 2: In 3GPP Rel-15 neither timer-based back-off mechanism nor automatically preventing a UE to send another PDU SESSION ESTABLISHMENT REQUEST message is available. Hence it is recommended that the network includes the Back-off timer value IE in PDU SESSION ESTABLISHMENT REJECT to prevent the UE sending the PDU SESSION ESTABLISHMENT REQUEST immediately.

The UE shall support the Default_QoS_Flow_usage_restriction_policy parameter as specified in Annex C.3. If the Default_QoS_Flow_usage_restriction_policy is set to disallow media as specified in Annex C.3, then if a QoS flow with GBR or with non-GBR for the media does not exist, the UE must consider itself not having local resources. If the UE has no local resources, the UE must not send media. See also section U.2.2.5.1 of 3GPP TS 24.229 [8].

Note 3: The existence of a QoS flow for the media does not by itself grant the UE authority to send media.

SIP Signalling via HOS APN/DNN

As described in section 2.2.1.1, the UE determines that another IMS registration is required, which must use the HOS APN/DNN.

On receipt of a PDU SESSION ESTABLISHMENT REJECT, the UE must start the back-off timer and re-attempt as described in section 4.5.1.1. If the UE is still unable to establish a PDU Session to the HOS APN/DNN and the HOS APN/DNN is not the same as the Internet APN/DNN, then the UE must establish a PDU Session to the Internet APN/DNN. In this case, if the UE successfully establishes a PDU Session connection to the Internet APN/DNN, the UE must proceed with the IMS registration procedure as described in section 2.2.1.6.

4.1.1.1 HTTP Signalling via HOS APN/DNN

For XCAP and HTTP Content Server requests, the UE must be preconfigured or provisioned by the home operator with the ToConRef parameter as specified in Annex C.3 with the Network Identifier part of the APN for Home Operator Services to be used for these requests (see GSMA PRD NG.113 [55] for more information).

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

For XCAP and HTTP Content Server requests, the UE must establish the PDN connection to the APN for HOS only when required (unless already established), e.g. when the user modifies a supplementary service's setting. In case the UE did not re-use an established PDN connection (e.g. in case the HOS APN is the internet APN), the UE must close the connection at most 120s after the last transaction has been completed.

QoS Flow Considerations for Voice via IMS APN/DNN

For an IMS session request for a Conversational Voice call (originating and terminating), a GBR QoS flow for IMS-based voice must be created by the network utilising interaction with dynamic PCC. The network must initiate the creation of a GBR QoS flow to transport the voice media. The GBR QoS flow for Conversational Voice must utilise the standardised 5QI value of one (1) and have the associated characteristics as specified in 3GPP TS 23.501 [4]. The network must not create more than one QoS flow with same ARP for voice media for an IMS session. Therefore, the UE and network must be able to multiplex the media streams from multiple concurrent voice sessions.

Note 1: A single QoS flow 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 QoS flow for all voice media streams means that the same 5QI and/or ARP value are used for all voice media streams of an IMS session.

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

QoS Flow Considerations for Video via IMS APN/DNN

For an IMS session request for a video call (originating or terminating), a GBR QoS flow for IMS-based voice as described in section 4.5.2 and another QoS flow for video must be created utilising interaction with dynamic PCC. The network must initiate the creation of QoS flows to transport the video media.

The QoS flow for Conversational Video stream may be a GBR or a non-GBR QoS flow. If a GBR QoS flow is used it must utilize the standardized 5QI value of two (2) and have the associated characteristics as specified in 3GPP TS 23.501 [4].

For IMS session termination of a session using conversational media, the QoS flows for voice and for Conversational Video stream must be deleted utilising interaction with dynamic PCC. The network must initiate the deletion of the QoS flows.

QoS Flow Considerations for voice media on emergency PDU Session

For an IMS session request for an emergency call on the PDU session for emergency services, and if as a result of SDP offer/answer a voice media is negotiated, then a GBR QoS flow for voice media must be created by the network utilising interaction with dynamic PCC as specified in 3GPP TS 23.503 [6]. The network must initiate the creation of a GBR QoS flow to transport the voice media of an emergency call. The GBR QoS flow for voice media of an emergency call:

1. must utilise the standardised 5QI value of one (1);
2. must have the associated characteristics as specified in 3GPP TS 23.501 [4]; and
3. must have an ARP value that is reserved for emergency services.

For IMS session termination of an emergency call, the GBR QoS flow must be deleted utilising the interaction with dynamic PCC. The network must initiate the deletion of the QoS flows.

QoS Flow Considerations for MSRP via IMS APN/DNN

For an IMS session request for MSRP on the PDU session to the IMS well-known APN/DNN, a non-GBR QoS flow must be created utilising interaction with dynamic PCC. The network must initiate the creation of QoS flows to transport MSRP.

For IMS session termination of a session using MSRP, the QoS flow for MSRP must be deleted utilising interaction with dynamic PCC. The network must initiate the deletion of the QoS flow.

Exception handling

Loss of PDU session

If the PDU session 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 [8] (e.g. when the P-CSCF receives an abort session request from the Policy Control Function (PCF)).

If the UE discovers (for example during a TAU procedure) that the PDU session has been lost, then the UE must attempt to re-establish the PDU session. This will trigger the network to initiate a new SIP signalling QoS flow in conjunction with the PDU session establishment.

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

Loss of media QoS flow and Radio Connection

If a Guaranteed Bit Rate (GBR) QoS flow used for voice, fails to get established, or it is lost in mid-session, then the network must terminate the session associated to the voice stream according to the procedures defined in section 5.2.8 of 3GPP TS 24.229 [8] (P-CSCF needs to be informed about loss of QoS flow by the PCF).

If the Guaranteed Bit Rate (GBR) QoS flow used for video fails to get established, or is lost in mid-session, then the network, based on its policies, has the option to either allow the session to continue as a voice only call, or terminate the SIP session that the QoS flow is associated with, according to the procedures in section 5.2.8 of 3GPP TS 24.229 [8].

Note 1: Termination of the SIP session due to loss of the voice GBR QoS flow 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 QoS flow used for another media type fails to get established, or is lost in mid-session, then the network, based on its policies, has the option to either allow the SIP session to continue as is, or terminate the SIP session that the GBR QoS flow is associated with (the network can handle loss of video in a video call in such a way that the session continues as voice only).

If a SIP session includes media streams, e.g. for video, and if a QoS flow for any media stream fails to get established, or is lost in mid-session, then the UE must, based on its preferences, modify, reject or terminate the SIP session that the media QoS flow is associated with, according to section 6.1.1 of 3GPP TS 24.229 [8]. The UE can act differently per media type.

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

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

P-CSCF Discovery

The UE and the packet core must support the procedures for P-CSCF discovery via 5GS. These are described in Annex U.2.2.1 of 3GPP TS 24.229 [8], as option II for P-CSCF discovery.

The UE must indicate P-CSCF IPv6 Address Request and P-CSCF IPv4 Address Request when performing the following procedures (see also section 4.9)

1. during the establishment of the PDU session to the IMS well-known APN/DNN; and
2. during the establishment of the PDU session for emergency services.

The UE must use the P-CSCF addresses received during a PDU session establishment to the IMS well-known APN/DNN when accessing to non-emergency services and must use the P-CSCF addresses received during the PDU session establishment when accessing the emergency services as defined in section 5.1 of this document and 3GPP TS 24.229 [8].

If the UE receives a PDU SESSION MODIFICATION COMMAND message containing a list of P-CSCF addresses that do not include the address of the currently used P-CSCF, then the UE must acquire a P-CSCF different from the currently used P-CSCF, and initiate a new initial registration as described in section U.2.2.1C 3GPP TS 24.229 [8].

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

Interworking with E-UTRAN connected to EPC

In order to offer its customers a seamless session continuity and various deployment scenarios to obtain the IMS voice service, the operator may wish to complement the NG-RAN and 5GC by using E-UTRAN and EPC (see also 3GPP TS 23.501 [4] and GSMA PRD IR.92 [2]). This section describes the additional features that need to be implemented for the UEs and the networks that wish to support such a deployment scenario.

The UE and the network must support the interworking with E-UTRAN connected to EPC for single registration mode with N26 interface as specified in section 4.11 of 3GPP TS 23.502 [5], 3GPP TS 24.501 [7], 3GPP TS 24.301 [29] and 3GPP TS 38.413 [30]. Interworking with E-UTRAN connected to EPC with N26 interface requires the support of the idle mode mobility procedures as specified in section 4.11.1.3 of 3GPP TS 23.502 [5] and the handover procedures as specified in section 4.11.1.2 of 3GPP TS 23.502 [5].

If the UE establishes a PDN connection to the IMS well-known APN via E-UTRAN connected to EPC, using the PDN type IPv4v6, the network can override the PDN type requested by the UE to be limited to a single address PDN type (IPv4 or IPv6, respectively), and set the ESM cause value to #50 "PDN type IPv4 only allowed", or #51 "PDN type IPv6 only allowed", respectively as specified in section 6.2.2 of 3GPP TS 24.301 [29]. In this case the UE must not request another PDN connection to the IMS well-known APN for the other IP version during the existence of the PDN connection.

The UE and the network must support the EPS fallback procedure for IMS voice with N26 interface as specified in section 4.13.6.1 of 3GPP TS 23.502 [5], 3GPP TS 38.331 [31] and 3GPP TS 38.413 [30].

The present version of this PRD in the context of interworking is restricted to system change between 5GS (NR (AN) / 5GC) and EPS (E-UTRAN / EPC).

PDU Session to the IMS well-known DNN

Note 1: This version of the document profiles the establishment of a single PDU session to the IMS well-known APN/DNN.

The UE and the network must support the PDU session establishment procedure as specified in section 6.4 of 3GPP TS 24.501 [7] and section 5.15.5.3 of 3GPP TS 23.501 [4].

Note 2: If the Requested NSSAI contains no S-NSSAI, then the AMF selects the S-NSSAI.

The UE must set the requested PDU Session Type as specified in section 5.8.2.2.1 of 3GPP TS 23.501 [4].

The UE must select the SSC mode as specified in section 5.6.9.3 of 3GPP TS 23.501 [4].

Note 3: If the UE does not include SSC mode, then the SMF selects the SSC mode for the PDU session.

The UE must support both IPv4 and IPv6 for all protocols that are used. At PDU session establishment to the IMS well-known APN/DNN, the UE must set the PDU Session type as specified in section 5.8.2.2.1 of 3GPP TS 23.501 [4] and section 6.2.4 of 3GPP TS 24.501 [7].

If the UE has used the PDU Session type IPv6 during PDU session establishment and receives an ESTABLISHMENT REJECT message with the 5GSM cause value #50 "PDU session type IPv4 only allowed", then the UE must initiate another PDU session establishment with the PDU Session type IPv4 to the same APN/DNN as specified in section 6.4.1.4 of 3GPP TS 24.501 [7].

If the UE has used the PDU Session type IPv4v6 during PDU session establishment, and if both IPv4 and IPv6 addresses are assigned by the network to the UE, then, the UE must prefer the IPv6 address type when the UE discovers the P-CSCF. If only an IPv4 address or only IPv6 address is assigned by the network to the UE, then the network must send 5GSM cause #50 "PDU session IPv4 only allowed" or #51 "PDU session IPv6 only allowed", respectively, to the UE, and the UE must not request another PDU session to the APN/DNN utilised in the PDU session establishment for the other IP version as specified in section 6.4.1.3 of 3GPP TS 24.501 [7].

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

Note 4: 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.

UE Route Selection Policy

The UE must support UE Route Selection Policy (URSP) rules as specified in section 6.6.2 of 3GPP TS 23.503 [6].

Note: The UE can be pre-configured with URSP rules.

4.2 UE Configuration Update procedure for transparent UE Policy delivery

The UE must and the network can, support UE Configuration Update procedure for transparent UE Policy delivery as specified in section 4.2.4.3 of 3GPP TS 23.502 [5] using NAS transport procedure as specified in section 5.4.5 of 3GPP TS 24.501 [7].

Note: The UE Configuration Update procedure for transparent UE Policy delivery is executed as a part of UE Policy Association Establishment and UE Policy Association Modification procedures defined in sections 4.16.11 and 4.16.12 of 3GPP TS 23.502 [5].

5 Common Functionalities

5.1 Emergency service

The UE and the network must support the IMS emergency services as specified in 3GPP TS 24.229 [8] and Annex H of 3GPP TS 23.167 [80].

The UE must support:

a) the Emergency Services natively over 5GS,
b) the Emergency Services with a handover and/or a redirection to an E-UTRAN cell at QoS Flow establishment for IMS Emergency Services and

Note: Redirection (RRC connection release and redirection to a LTE frequency) may not always meet the service requirement for emergency service as the session establishment delay may be longer. Handover provides a more reliable solution.

- the Emergency Services Fallback procedure as specified in 3GPP TS 23.501 [4] and 3GPP TS 23.502 [5].

The UE must support the emerg-reg timer defined in table 7.8.1 of 3GPP TS 24.229 [8] and the related procedure defined in section 5.1.6.1 of 3GPP TS 24.229 [8]. The operator can configure the UE with the emerg-reg timer parameter as specified in Annex C.3.

If the UE supports SR-VCC for emergency session for handover from E-UTRAN connected to EPC as described in Annex A.5 of GSMA PRD IR.92 [2], the UE must include the SIP instance ID as specified in section 7.2.1 of 3GPP TS 24.237 [12].

In case the VPMN supports other emergency numbers than 112 and 911, then the network must provide the Extended Emergency Number List to the UE as specified in 3GPP TS 24.501 [7], unless all the local emergency numbers are provided in the Emergency Number List where each number is associated with a single emergency service category value as specified in section 10.5.4.33 of 3GPP TS 24.008 [71]. The SUPL enabled UE sends the emergency SUPL messages related to the UE detectable emergency session within the user plane of the Emergency PDU session. The SUPL enabled UE sends the emergency SUPL messages related to the non UE detectable emergency session within the user plane of the PDU session to the IMS well-known APN/DNN.

Roaming Considerations

This profile has been designed to support IMS voice roaming. For more information on the IMS voice roaming models see GSMA PRD NG.113 [55].

Accesses in addition to NG-RAN

UEs that support cellular (e.g. NG-RAN) and Internet access via non-cellular access must use:

1. the cellular access as transport of the Gm reference point; and
2. the cellular access as transport of the Ut reference point, unless preconfigured otherwise by the home operator with the AccessForXCAP parameter as specified in Annex C.3.

Data Off and Services Availability

General

The UE and the network must support 3GPP PS Data Off. When 3GPP PS Data Off is activated by the user, the UE and the network must not send via any PDU session any IP packet of any service other than 3GPP PS Data Off Exempt Services. This applies to both non-IMS based services and SIP-based IMS services.

The UE must support to be provisioned with the list of SIP-based 3GPP PS Data Off Exempt Services as per section 5.67 of 3GPP TS 24.167 [67] and section 5.7 5.8 of 3GPP TS 24.275 [68].

The UE must support to be provisioned with the list of non-IMS 3GPP PS Data Off Exempt Services as per section 5.10i of 3GPP TS 24.368 [69] and section 5.11 of 3GPP TS 24.424 [70].

A UE must report its 3GPP PS Data Off status, and each change in its 3GPP PS Data Off status as specified in 3GPP TS 24.229 [8] and in sections 6.2.10 and 6.4.2.2 of 3GPP TS 24.501 [7].

The UE must support to be provisioned for data off for the MMTEL Call Composer service via the configuration parameter PRE AND POST CALL DATA OFF (see Annex C.3) as specified in section 2.8.1.5 of GSMA PRD RCC.07 [73].

If the 3GPP PS Data Off status changes from "inactive" to "active" the UE must release all IMS sessions/dialogs that fulfil the conditions in section U.3.1.5 of 3GPP TS 24.229 [8].

If MMTel Video Service is not in the list of 3GPP PS Data Off Exempt Services, or if MMTel Video Service in the list of 3GPP PS Data Off Exempt Service is configured not to be a 3GPP PS Data Off Exempt Service (see Annex C.3), and if 3GPP PS Data Off is active, the use of MMTel Video Service is disabled.

Note: The UE can disconnect PDU sessions that are not required by 3GPP PS Data Off Exempt Services.

Supplementary Service Settings Management

The UE must be able to perform supplementary service settings management as described in section 2.3 regardless of whether 3GPP PS Data Off is active. For this reason, the operator must configure "IMS Supplementary Service configuration via the Ut interface using XCAP" as 3GPP PS Data Off Exempt Service, see Annex C.3.

Voice/Video Calls and SMS over IP

The UE must be able to initiate and to receive Voice/Video Calls and SMS over IP as described in the main body of this document regardless whether 3GPP PS Data Off is active. For this reason, the operator must configure "MMTel Voice", "MMTel Video" and "SMS over IP" as 3GPP PS Data Off Exempt Services, see Annex C.3.

RCS Messaging Services / MSRP based Enriched Calling Services

When configured to allow RCS Messaging services and MSRP based Enriched Calling services, the UE must be able to support these services as described in sections 2.5 and 2.6 regardless of whether 3GPP PS Data Off is active. To enable this, the UE shall support the RCS Data Off configuration parameters as defined in Annex A.1.14 of GSMA PRD RCC.07 [73].

HTTP Content Server

For the MMTEL Call Composer service, RCS Messaging services and MSRP based Enriched Calling services, the UE must support the procedures for file uploading and file downloading to the HTTP Content Server. The network must provide a HTTP Content Server if the MMTEL Call Composer service is enabled or if any of the RCS Messaging services or MSRP based Enriched Calling services that require an HTTP Content Server are enabled.

The procedures for the file upload to the HTTP Content Server are defined in steps 1, 2, 3 and 4 of section 3.2.5.3.1.1 of GSMA PRD RCC.07 [73] and section 3.2.5.3.1.2 of GSMA PRD RCC.07 [73]. The operator can specify the HTTP Content Server for file uploading via the configuration parameter FT HTTP CS URI, see Annex C.3.

The procedures for the file downloading from the HTTP Content Server are defined in sections 3.2.5.3.2 and 4.1.15.4 of GSMA PRD RCC.07 [73]. The operator can specify the

HTTP Content Server for file downloading via the configuration parameter FT HTTP DL URI, see Annex C.3.

Annex A USSI

A.1 Introduction

A.1.1 Overview

The support of USSI is optional for both the UE and the Network. This Annex describes the additional functionalities that need to be implemented for the UEs and networks that support USSI. The scope includes the following aspects:

1. IMS basic capabilities [Annex A.2]
2. Media negotiation [Annex A.3]
3. Functionality that is relevant across the protocol stack and subsystems [Annex A.4]

UEs and networks supporting USSI must be compliant with all of the normative statements in this Annex.

In this version of the PRD, only voice-capable UEs and networks are considered.

A.2 IMS Feature Set

A.2.1 General

This Annex lists the additional mandatory capabilities compared to the main body that are required over the Gm reference point.

A.2.2 Support of generic IMS functions

A.2.2.1 General

The UE and the network must fulfil the requirements for addressing as specified in section 4.5.4.1 of 3GPP TS 24.390 [87].

Note: Geo-local numbering is not used in USSI.

A.2.2.2 SIP Registration Procedures

A UE must perform a SIP Registration as specified in 2.2.1 and must include a g.3gpp.nw-init-ussi media feature tag in the Contact header field as specified in 3GPP TS 24.390 [87].

A.2.3 USSI Considerations

A.2.3.1 General

The following sub-sections provide considerations for the UE and USSI AS.

A.2.3.2 UE initiated USSI Request

The UE must support the invocation and operation of user initiated USSI as defined in sections 4.5.4.1 and 4.5.3 of 3GPP TS 24.390 [87].

The USSI AS (USSI Application Server) must support the actions defined in section 4.5.4.2 of 3GPP TS 24.390 [87].

USSI AS initiated Request

The USSI AS may support the invocation and operation of the network initiated USSI as defined in section 4.5.5.1 of 3GPP TS 24.390 [87].

The UE must support the actions defined in section 4.5.5.2 of 3GPP TS 24.390 [87].

A.3 SDP negotiation

The UE and the IMS core network must support the SDP negotiation to not use media resources for UE initiated USSI and network initiated USSI as described in sections 4.5.2 and 4.5.2A (respectively) of 3GPP TS 24.390 [87].

A.4 Common Functionalities

A.4.1 Data Off

The UE must fulfil the requirements specified in section 5.4.1.

The UE must be able to initiate and receive USSI messages as described in this Annex regardless whether 3GPP PS (Packet Switched) Data Off is active i.e. the UE continues to use (does not disconnect) the PDN connection via the IMS well-known APN/DNN. For this reason, the operator must configure "USSI_exempt" as 3GPP PS Data Off Exempt Services, see Annex C.3.

Annex B Features needed in certain regions

B.1 General

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

B.2 Global Text Telephony over IP

In some regions, there are regulatory requirements that allows deaf/hearing impaired people to use text-based communication known as Teletype Writer (TTY) to other users and government offices (e.g. to provide equal access to emergency services to all users). In this document, the evolution of the legacy CS-based TTY in IMS is referred to as Global Text Telephony over IP (GTT-IP).

Note 1: GTT-IP is also referred to as Real Time Text in some standards. The use of GTT-IP in this section infers both GTT-IP and Real Time Text.

The following requirements outline how the GTT-IP service should be implemented in regions where required.

The UE must include the text media feature tag, as defined in IETF RFC 3840 [18], in the Contact header field of the SIP REGISTER request, in the Contact header field of the SIP INVITE request, and in the Contact header field of the SIP response to the SIP INVITE request, using the procedures defined in 3GPP TS 24.229 [8].

GTT-IP messages must use ITU-T Recommendation T.140 [52] real-time text according to the rules and procedures specified in 3GPP TS 26.114 [16] with the following clarifications:

- The call with GTT-IP component must contain both "text" and "audio" media RTP streams negotiated using existing SDP offer/answer procedures. The text media stream may be released mid-call, resulting in the call becoming a voice call. The “audio” media stream may not be released mid-call.
- Note 2: The implementation of calls with single "text" media is not supported.
- For real-time text, RTCP reporting must be turned on by setting the SDP bandwidth modifiers “RS” and “RR” as specified in section 3.5.3.
- The sampling time used must be 300 ms.
- Change of the sampling time (rate adaptation) is not required.
  For an IMS session request for a call with GTT-IP component (originating and terminating), a QoS flow for the T.140 text media must be created by the network using interaction with dynamic PCC. The network must initiate the creation of a dedicated QoS flow to transport the text media.

For a scenario when GTT-IP component is added or removed during the session, the existing QoS flow must be modified to add or to remove text media using interaction with dynamic PCC.

As stated in annex E.4 of 3GPP TS 26.114 [16], the QoS flow for a call with GTT-IP component may use:

- a Non-GBR QoS flow with a 5QI value of 6, 8 or 9; or

- a GBR QoS flow with a 5QI value of 1.
  For networks residing in regions where regulatory requirements include requirements for end-to-end text latency, the usage of a GBR QoS flow with a 5QI of 1 for transport of both audio and real-time text RTP streams is recommended.

For IMS session release of a call with GTT-IP component, the QoS flow must be deleted by the network using interaction with dynamic PCC. The network must initiate the deletion of the QoS flow.

Annex C MNO provisioning and Late Customization

C.1 General

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

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

C.2 Configuration Methods

C.2.1 Remote Client Configuration for MNO provisioning

The UE and the network must support remote client configuration as specified in GSMA PRD RCC.07 [73] in order to support MNO provisioning (see also Table C.2-1).

C.2.2 Late Customization

The UE must support late customization as specified in GSMA PRD TS.32 [75] for the parameters that are in Table C.2-1 and can support late customization for the remaining parameters of GSMA PRD TS.32 [75].

C.3 Configuration Parameters

Table C.3-1 contains the configuration parameters with their default values, that must be supported by the UE and the network. The UE must use the default value for each parameter in Table C.2-1 unless configured differently as described in Annex C.2.

Note: The parameters in Table C.3-1 are a subset of parameters defined in section 3.9 of GSMA PRD TS.32 [75].

Parameter	Default value	Defined in	See also section
IMS	0-Enabled	Section 5.13 of 3GPP TS 24.305 [9] as /<X>/IMS	2.2.1
RCS VOLTE SINGLE REGISTRATION	1 (single registration)	Section A.3.2 of GSMA PRD RCC.07 [73] (/<X>/rcsVolteSingleRegistration) Where <X> corresponds to the <X> node below the Ext node of the IMS sub tree defined GSMA PRD RCC.15 [88]	2.2.1.1
RegRetryBaseTime	30 Sec	Section 5.35 of 3GPP TS 24.167 [67] (/<X>/ RegRetryBaseTime)	2.2.1.6
RegRetryMaxTime	1800 sec	Section 5.35 of 3GPP TS 24.167 [67] (/<X>/ RegRetryMaxTime)	2.2.1.6

Parameter	Default value	Defined in	See also section
Media_type_restriction_policy (Audio and/or Video over 5G allowed while roaming on 5G core (5GC) via NG-RAN IP-CAN)	Voice and video allowed	Sections 5.43 and 5.48 of 3GPP TS 24.167 [67] (interior node /<X>/Media_type_restriction_policy and leaf/<X>/Roaming) and 3GPP TS 24.229 [8]	2.2.1.7, 2.2.10, 3.3.2.1
Media_type_restriction_policy (Voice and/or Video over 5G allowed)	Voice and video allowed	Section 5.43 of 3GPP TS 24.167 [67] (interior node /<X>/Media_type_restriction_policy) and 3GPP TS 24.229 [8]	2.2.1.7, 2.2.10, 3.3.2.1
SMSoIP_usage_policy (When to use SMSoIP)	2 - SMSoIP irrespective of IMS voice support	Section 5.71 of 3GPP TS 24.167 [67] (/<X>/ SMSoIP_usage_policy)	2.2.1.7
FT HTTP CS USER	No default	Section A.2.4 of GSMA PRD RCC.07 [73] (/<X>/Messaging/FileTransfer/ftHTTPCSUser)	2.2.2.3
FT HTTP CS PWD	No default	Section A.2.4 of GSMA PRD RCC.07 [73] (/<X>/Messaging/FileTransfer/ftHTTPCSPwd)	2.2.2.3
Policy_on_local_numbers (Local number type for voice and video calls)	1 -Home-local number	Section 5.62 of 3GPP TS 24.167 [67] (/<X>/ Policy_on_local_numbers)	2.2.3.2
Timer_T1	2 sec	Section 5.10 of 3GPP TS 24.167 [67] (/<X>/Timer_T1)	2.2.4
Timer_T2	16 sec	Section 5.11 of 3GPP TS 24.167 [67] (/<X>/Timer_T2)	2.2.4
Timer_T4	17 sec	Section 5.12 in 3GPP TS 24.167 [67] (/<X>/Timer_T4)	2.2.4
Reliable 18x policy (Sending SIP 18x reliably)	1 – Indicates that the SIP 18x responses (other than SIP 183 response) are to be sent reliably	Section 5.56 of 3GPP TS 24.167 [67] (/<X>/ Reliable_18x_policy /<X>/ Send_18x_Reliably) and 3GPP TS 24.229 [8]	2.2.4

Parameter	Default value	Defined in	See also section
Precondition_disabling_policy (SIP Preconditions used)	0 – the UE is allowed to use the precondition mechanism	Section 5.60 of 3GPP TS 24.167 [67] (/<X>/Precondition_disabling_policy) and section 5.1.5A of 3GPP TS 24.229 [8]	2.2.5
CAPABILITY DISCOVERY MECHANISM	2 - OFF	Section A.2.5 of GSMA PRD RCC.07 [73] (/<X>/CapDiscovery/defaultDisc)	2.2.10
XCAP Root URI	No default	3GPP TS 24.623 [26]	2.3.2
Conf_Factory_URI (Conference Factory URI)	None	Section 5.4 of 3GPP TS 24.166 [81] (/<X>/Conf_Factory_URI)	2.3.3
FromPreferred	0 - From header field; is not used for determination of the originating party identity in OIP service	3GPP TS 24.607 [33] and section 5.4 of 3GPP TS 24.417 [91] (/<X>/FromPreferred)	2.3.12
COMPOSER AUTH	0 – Indicates that Call Composer service is disabled	Section 2.1.2 of GSMA PRD RCC.20 [72] (/<X>/Services/composerAuth) The value "3" indicate that both the MMTEL Call Composer service and the MSRP Call Composer service are enabled. The value "2" indicates that the MMTEL Call Composer service is enabled. The value “1” indicates that the MSRP Call Composer service is enabled. If the value “1” is configured and the UE only supports MMTEL Call Composer, then the UE must behave as defined for a value of “0”. If the value “3” is configured and the UE only supports MMTEL Call Composer, the UE must behave as defined for a value of “2”. If any other value is configured, the UE must behave as defined for a value of "0".	2.3.13, 2.6

Parameter	Default value	Defined in	See also section
SMS_Over_IP_Networks_Indication	1 – SMS service is preferred to be invoked over the IP networks	Section 5.28 of 3GPP TS 24.167 [67] (/<X>/SMS_Over_IP_Networks_Indication)	2.4
RateSet for AMR	0,2,4,7 ("mode-set = 0,2,4,7" included in the SDP answer)	Section 15.2 of 3GPP TS 26.114 [16] (/<X>/Speech/<X>/RateSet) with (/<X>/Speech/<X>/Codec= "amr")	3.2.2.2
RateSet for AMR-WB	Undefined (no mode-set parameter included in the SDP answer)	Section 15.2 of 3GPP TS 26.114 [16] (/<X>/Speech/<X>/RateSet) (/<X>/Speech/<X>/Codec= "amr-wb")	3.2.2.2
ICM/INIT_PARTIAL_REDUNDANCY_OFFSET_RECV	Undefined (ch-aw-recv not included in SDP offer)	Defined in section 17.2 of 3GPP TS 26.114 [16] /<X>/Speech/<X>/ ICM/INIT_PARTIAL_REDUNDANCY_OFFSET_RECV	3.2.2.3
EVS/Br	5.9-24.4	Section 15.2 of 3GPP TS 26.114 [16] (/<X>/Speech/<X>/EVS/Br) and section 5 of 3GPP TS 26.441 [56] (Table 1)	3.2.2.3
EVS/Bw	nb-swb	Section 15.2 of 3GPP TS 26.114 [16] (/<X>/Speech/<X>/EVS/Bw) and section 5 of 3GPP TS 26.441 [56] (Table 1)	3.2.2.3
ToConRef (Network Identifier part of the HOS APN)	Internet APN	Section 5.9 of 3GPP TS 24.424 [70] (/<X>/XCAP_conn_params_policy/<X>/ XDM_MO_ref) and 3GPP TS 24.623 [26]	4.5.1
Default_QoS_Flow_usage_restriction_policy (Voice and Video Media on 5QI=5 QoS flow)	Voice and video prohibited	Section 5.79 of 3GPP TS 24.167 [67] (interior node /<X>/ Default_QoS_Flow_usage_restriction_policy) and 3GPP TS 24.229 [8]	4.5.1
emerg-reg	10 sec	Section 5.61 of 3GPP TS 24.167 [67] (/<X>/Timer_Emerg-reg)	5.1
AccessForXCAP	1 – 3GPP accesses only	3GPP TS 24.424 [70] and section 5.2.1.3 of 3GPP TS 24.623 [26] (/<X>/AccessForXCAP)	5.3

Parameter	Default value	Defined in	See also section
PRE AND POST CALL DATA OFF	1 – indicates that Call Composer services are cellular data off exempt services	Section A.2.2 of GSMA PRD RCC.07 [73] (/<X>/Services/Ext/DataOff/preAndPostCallDataOff)	5.4.1
SS_XCAP_config_exempt	1 - Indicates that the SS configuration via XCAP is a 3GPP PS data off exempt service	Section 5.11 of 3GPP TS 24.424 [70] (/<X>/3GPP_PS_data_off/SS_XCAP_config_exempt).	5.4.2
MMTEL_voice_exempt	1 - Indicates that the MMTEL voice is a 3GPP PS data off exempt service	Section 5.7 of 3GPP TS 24.275 [68] (/<X>/3GPP_PS_data_off/MMTEL_voice_exempt)	5.4.3
MMTEL_video_exempt	1 - Indicates that the MMTEL video is a 3GPP PS data off exempt service	Section 5.8 of 3GPP TS 24.275 [68] (/<X>/3GPP_PS_data_off/MMTEL_video_exempt)	5.4.3
SMSoIP_exempt	1 - Indicates that the SMS over IP is a 3GPP PS data off exempt service	Section 5.67 of 3GPP TS 24.167 [67] (/<X>/3GPP_PS_data_off/SMSoIP_exempt)	5.4.3
FT HTTP CS URI	As defined in section A.1.4 of GSMA PRD RCC.07 [73]	Section A.2.4 of GSMA PRD RCC.07 [73] (/<X>/Messaging/FileTransfer/ftHTTPCSURI)	5.5

Parameter	Default value	Defined in	See also section
FT HTTP DL URI	No default	Section A.2.4 of GSMA PRD RCC.07 [73] (/<X>/Messaging/FileTransfer/ftHTTPDLURI)	5.5
USSI_exempt	1 - Indicates that USSI is a 3GPP PS data off exempt service	Section 5.4B of 3GPP TS 24.391 [82] (/<X>/3GPP_PS_data_off/USSI_exempt)	A.4
Device_management_over_PS	1 - Indicates that the device management over PS is a 3GPP PS data off exempt service	Section 5.10i of 3GPP TS 24.368 [69] (/<X>/3GPP_PS_data_off/Exempted_service_list/<X>/Device_management_over_PS)	C.2

Table C.2-1 Configuration parameters and their default values

Annex D Document Management

D.1 Document History

Version	Date	Brief Description of Change	Approval Authority	Editor / Company
1.0	25/09/2019	New PRD (5GJA#6)).	First Draft	George Foti <george.foti@ericsson.com>
1.0r01	15/01/2020	Implemented CR 1002, CR 1003, CR 1004, CR 1005.		CR 1002 : Ericsson - Resolving editor's note Annex C.3 CR 1003 : Ericsson - Video Level Alignment CR 1004 : China Mobile - Revision for reference description CR 1005 : GSMA - Updating of EPS Fallback
1.0r03	15/01/2020	Implemented CR 1006,CR 1007, CR 1008, CR 1009.		CR 1006 : Ericsson - QoS Flow Details for MRSP CR 1007 : Ericsson - PANI in SIP ACK CR 1008 : Nokia - Scope Clarification for RAN-NG Support CR 1009 : Nokia - Scope Clarification for Interworking with EPC
1.0r04		Implemented CR 1010, CR 1011, CR 1012, CR 1013, CR 1015.		CR 1010: GSMA - Local ring tone – align with IR.92 CR1190 . CR 1011: Huawei – Clarification for MMTEL Call composer service CR 1012: GSMA - Restore Missing Paragraph CR 1013: Ericsson - Several Corrections CR 1015 : Nokia - Update Section Reference to TS.32 in Annex
1.0r06	27/03/2020	Implemented CR 1016		CR 1016: HWA - Removing Unexpected call composer elements

Other Information

Type	Description
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Editor / Company	George Foti

Feedback

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