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기술보고서

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사업자 간 UICC 이동성을 위한 VoLTE 단말규격(R4)과 GSMA IR 문서와의 비교 분석(기술보고서)

Comparison analysis between VOLTE Terminal Specification for UICC Portability(R4) and GSMA IR Documents (Technical Report)

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본 표준 발간 이전에 접수된 지식재산권 확약서 정보는 본 표준의 '부록(지식재산권 확약서 정보)'에 명시하고 있으며, 이후 접수된 지식재산권 확약서는 TTA 웹사이트에서 확인할 수 있습니다.
본 표준과 관련하여 접수된 확약서 외의 지식재산권이 존재할 수 있습니다.

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서 문

1 기술보고서의 목적

이 기술보고서의 목적은 TTA의 사업자 간 UICC 이동성 제공을 위한 VoLTE 단말 규격(TTAK.KO-06.0357) 4차 개정본에 대해 GSMA의 IR.92 및 IR.94로부터 직접 인용된 부분과 추가 및 수정 사항을 구분하여 명시하는 데에 있다.

2 주요 내용 요약

이 기술보고서는 사업자 간 UICC 이동성 제공을 위한 VoLTE 단말 규격 (TTAK.KO-06.0357) 4차 개정본의 각 본문 내용을 영어로 기술하고, GSMA IR.92 v.10.0 및 IR.94 v.11.0 본문 내용과 비교하여 수정 및 변경된 경우 파란 표기 및 추가된 경우에는 붉은 표기를 하여 두 표준간 차이를 나타낸다. GSMA IR.92 및 IR.94 본문 내용 중 삭제된 부분은 부록 II에 명시한다.

3 인용 표준과의 비교

3.1 인용 표준과의 관련성

이 기술보고서는 사업자 간 UICC 이동성 제공을 위한 VoLTE 단말 규격 (TTAK.KO-06.0357) 4차 개정본을 기반으로 한다.

3.2 인용 표준과 본 기술보고서와의 비교표

해당 사항 없음

Preface

1 Purpose

The purpose of this technical report is to provide the contents of GSMA IR.92 / IR.94 selected by the 4th revision of “Specification of VoLTE Terminal for UICC Portability between Mobile Operators(TTAK.KO-06.0357)” and to indicate the additions and modifications made.

2 Summary

This technical report is an English version of the 4th revision of “Specification of VoLTE Terminal for UICC Portability between Mobile Operators(TTAK.KO-06.0357)” and the color label is used to indicate the difference between TTA standard and GSMA IR documents; the blue characters for modifications and the red characters for additions. The removed contents of GSMA IR.92 and IR.94 are separately listed in Appendix II.

3 Relationship to Reference Standards

This technical report is based on the 4th revision of “Specification of VoLTE Terminal for UICC Portability between Mobile Operators (TTAK.KO-06.0357).

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(Comparison analysis between VOLTE Terminal Specification for UICC Portability and GSMA IR Documents (Technical Report))

1 Scope

This technical report is an English version of the 3rd revision of "Specification of VoLTE Terminal for UICC Portability between Mobile Operators(TTAK.KO-06.0357)" and the color label is used to indicate the difference between TTA standard and GSMA IR documents; the blue characters for modifications and the red characters for additions.

2 Reference Standards

GSMA IR 92, V10.0, 'IMS Profile for Voice and SMS'

GSMA IR 94, V11.0, 'IMS Profile for Conversational Video Service'

3 Terms Definition

None

4 Abbreviations

3GPP	3rd Generation Partnership Project
AM	Acknowledged Mode
AMR	Adaptive Multi-Rate
AMR-WB	Adaptive Multi-Rate Wideband
APN	Access Point Name
AVP	Audio Video Profile
AVPF	AVP Feedback Profile

CB	Communication Barring
CCM	Codec Control Message
CDIV	Communication Diversion
CDIVN	CDIV Notification
CFNL	Communication Forwarding on Not Logged-in
CFNRc	Communication Forwarding on Not Reachable
CID	Caller ID
CN	Core Network
CNAME	Canonical End-Point Identifier SDES Item
CS	Circuit Switched
CSFB	CS Fallback
CW	Communication Waiting
DRB	Data Radio Bearer
DRX	Discontinuous Reception
DTX	Discontinuous Transmission
eNB	eNodeB
EPS	Evolved Packet System
E-UTRAN	Evolved Universal Terrestrial Radio Access Network
FIR	Full Intra Request
FDD	Frequency-Division Duplexing
GBR	Guaranteed Bit Rate
GRUU	Globally Routable User agent URI
GSM	Global System for Mobile communications
HSPA	High-Speed Packet Access
ICS	IMS Centralized Services
ICSI	IMS Communication Service Identifier
IM	IP Multimedia
IMPU	IP Multimedia Public Identity
IMS	IP Multimedia Subsystem
IMS-AKA	IMS Authentication and Key Agreement
IMSI	International Mobile Subscriber Identity
IP	Internet Protocol

IPv4	Internet Protocol Version 4
IPv6	Internet Protocol Version 6
ISIM	IM Services Identity Module
LTE	Long Term Evolution
MMTel	Multimedia Telephony
MO	Managed Object
MRFP	Media Resource Function Processor
NACK	Negative Acknowledgment
MS	Mobile Station
MS-ISDN	Mobile Subscriber ISDN Number
MWI	Message Waiting Indication
NGBR	Non Guaranteed Bit Rate
PCC	Policy and Charging Control
PCO	Protocol Configuration Option
PCRF	Policy and Charging Rules Function
P-CSCF	Proxy - Call Session Control Function
PDN	Packet Data Network
PLI	Picture Loss Indication
PDP	Packet Data Protocol
PRD	Permanent Reference Document
PS	Packet Switched
QCI	Quality of Service Class Indicator
QOS	Quality of Service
RAB	Radio Access Bearer
RAT	Radio Access Technology
RFC	Request For Comment
RLC	Radio Link Control
RoHC	Robust Header Compression
RTCP	RTP Control Protocol
RTP	Real Time Protocol
SCC AS	Service Centralization and Continuity Application Server
SDES	Session Descriptor RTCP Packet

SDP	Session Description Protocol
SigComp	Signalling Compression
SIP	Session Initiation Protocol
SMSoIP	SMS over IP
SR	Sender Report
SRB	Signalling Radio Bearer
SR-VCC	Single Radio Voice Call Continuity
TAS	Telephony Application Server
TDD	Time-Division Duplexing
TFO	Tandem-Free Operation
TMMBN	Temporary Maximum Media Stream Bit Rate Notification
TMMBR	Temporary Maximum Media Stream Bit Rate Request
TrFO	Transcoder-Free Operation
UDP	User Datagram Protocol
UE	User Equipment
UICC	Universal Integrated Circuit Card
UM	Unacknowledged Mode
URI	Uniform Resource Identifier
VoIP	Voice Over IP
XCAP	XML Configuration Access Protocol
XML	eXtensible Markup Language

5 IMS Features

5.1 General

The following is an essential IMS profile required over the Gm reference point.

5.2 Support of Generic IMS Functions

5.2.1 SIP Registration Procedures

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

The UE must include IMS Communication Service Identifier (ICSI) value used to indicate the IMS Multimedia Telephony service, that being urn:urn-7:3gpp-service.ims.icsi.mmtel per 3GPP TS 24.173 [14], using procedures as defined in section 5.1.1.2.1 of 3GPP TS 24.229 [15]. In order to indicate video service, the "video" feature tag as defined in IETF RFC 3840 must be added to the "Contact" header. If the UE supports SMS over IP, it must include feature tag used to indicate SMS over IP service, that being +g.3gpp.smsip as defined in section 5.3.2.2 of 3GPP TS 24.341 [19].

If the UE supports only video service, the UE uses only the audio;video feature tag as defined in IETF RFC 3840 [xx].

The UE must use the "User-Agent" header to transfer the UE information with the format “User-Agent: TTA-VoLTE/3.0 device-info device-type operator”. The operator indicates the network operator to which the UE initially is subscribed.

```
User-Agent = “User-Agent” HCOLON VoLTE-client-val
VoLTE-client-val = "TTA-VoLTE/3.0" SP device-info SP device-type SP operator

device-info = device-model SLASH software-version
device-model = 1*16(alphanum / “-“ / “_” / “.”)
software-version = 1*16(alphanum / “-“ / “_” / “.”)
```

```
device-type = "Device_Type" SLASH value
value = "Feature_Phone" / "Android_Phone" / "iPhone" / "Android_PAD" / "iPad"
/ "Tablet_PC" / token

operator = "SKT" / "KT" / "LGU" / “OMD”[1]/ token
```

Example) User-Agent: TTA-VoLTE/3.0 LG-F320L/LMR1R140207 Device_Type/ Android_Phone LGU+

The UE must subscribe to the “registration event package” as defined in 5.1.1.3 of 3GPP TS 24.229[15]. If a "489 Bad Event" response is received from the network for a "subscribe" message, no registration event must take place.

During SIP registration, the UE must include the IMEI URN defined in 13.8 of 3GPP TS 23.003[2] in the "+sip.instance" field parameter of the Contact header.

All IMS public user identities provided in the IRS (Implicit Registration Set) for VoLTE by the IMS core network must be alias user identites and may not include a Tel URI. The following public user identity must be assigned to the implicit registration set used for VoLTE and it must be used by the UE when registering for VoLTE:

a) If ISIM is used and there is only one record, the public user identity in the first record of the Elementary File in the ISIM (see 3GPP TS 31.103 [44] section 4.2.4) is used for both normal calls and emergency calls. If there are two records, the first record is used for emergency calls and the second for normal calls; or

b) If there is no ISIM, the temporary public user identity derived from the IMSI (3GPP TS 24.229 [15]) is used.

---

[1] OMD(Open Market Device): used for open market devices

No other IRS[2] must be used for VoLTE.

The first URI in the P-Associated-URI received from the current SIP registration must be used as the "From" and "P-Preferred-Identity" values in emergency calls.

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

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

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

---

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

5.2.2. Authentication

The UE must follow the procedures defined in 3GPP TS 24.229 [15] and 3GPP TS 33.203 [45] for authentication with IMS Authentication and Key Agreement (IMS-AKA), Sec-Agree and IPsec. Support of integrity protection is required for both UE and network. The confidentiality protection is optional. When IPsec negotiation fails due to the network problem, Sec-Agree and IPsec are not used for the UE.

The UE must use 'alg=hmac-sha-1-96, prot=esp, ealg=aes-cbc' as the default value for the Security-Client. If there is no prot parameter, esp must be used as the default value in accordance with IETF RFC 3329 [85].

The UE must send the AKAv1-MD5 algorithm data to the "algorithm" parameter of the "REGISTER Authorization" during an un-protected registration.

The UE must support the procedures for ISIM-based authentication. If there is no ISIM present on the UICC (Universal Integrated Circuit Card), the UE must support the procedures for USIM-based authentication as defined in Annex E.3.1 of 3GPP TS 23.228[7] and Annex C.2 of 3GPP TS 24.229[15].

5.2.3. Addressing

The UE and IMS core network must support Public User Identities[3] as defined in section 13.4 of 3GPP TS 23.003[2].

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

- MSISDN represented as a Tel URI:

  (e.g. tel:+447700900123)

- MSISDN represented as a SIP URI:

  (e.g. sip:+447700900123@example.com;user=phone)

The UE must support the 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 [15]. When using the SIP URI of Request URI, the UE must set a dial string containing the local number and the “user=phone” parameter to user part. When using Tel URI, the “phone-context” parameter must be set.

If both Tel URI and SIP URI are available, Tel URI has priority.

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

The UE must process input numbers as shown in <Table 5-1>.

<Table 5-1> Dialing Plan for Input Number Processing

Case	User input number	Number	URI
A dial string that doesn't start with “+”	8522	local number	tel:8522;phone-context=home_domain_name
	01032167941		tel:01032167941;phone-context=home_domain_name

Case	User input number	Number	URI
	*23#		tel:*23#;phone-context=home_domain_name
A dial string that starts with "+"	+821032167941	global number	tel:+821032167941

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

5.2.4. Call Establishment and Termination

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

The UE must include "require;explicit" in the "Accept-Contact" header. However, when registering for VT only, a "audio;video" feature tag must be used in the Accept-Contact header instead of "require;explicit." The method including "require;explicit" in the

Accept-Contact header is applied to all items except for "Options."

The UE 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 a “video” media feature tag[4] in the Accept-Contact header of an INVITE request for a Video call, as described in IETF RFC 3840[73] and IETF RFC 3841[74]. The UE must include an ICSI in the P-Preferred-Service header in case of outgoing call set-up.

The UE must be able to send 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 sendrecv SDP attribute, or accept the video media in simplex mode by using the send only or recvonly SDP attribute. The video stream in a video call may be changed between simplex or duplex mode, or 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 descriptor is set to zero.

The UE must send an SDP offer containing the full codec for a re-INVITE request, if it supports the EVS or HEVC codec.

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[4] The video media feature tag provides a preference to video/audio-supported devices in the forking algorithm together with the ICSI of the feature tag.

5.2.5. Forking

The UE must be ready to receive responses generated due to a forked request and behave according to the procedures specified in IETF RFC 3261 [55], section 4.2.7.3 of 3GPP TS 23.228 [7] and 3GPP TS 24.229 [15] and section 4.7.2.1 of '3GPP Release 13 TS 24.628[71. The UE must be able to maintain at least forty parallel early dialogues until receiving the final response on one of them and the UE must support receiving media on one of these early dialogues.

5.2.6. Early-Media and announcements

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

Furthermore, the UE must render locally generated communication progress information, if:

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

5.2.7 SIP Session Timer

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

- for an initial SIP INVITE request, the UE must include a Supported header with the option tag "timer"
- if the UE receives a SIP 422 response to an INVITE request, the UE must follow the procedures of section 7.4 in IETF RFC 4028 [86];
- if the UE includes the "refresher" parameter in the Session-Expires header field of the SIP INVITE request, the UE must set the "refresher" parameter to "uac";

- if a received SIP INVITE request indicates support of the "timer" option tag, and does not contain the Session-Expires header field, the "refresher" parameter with the value "uac" in SIP 2xx response to the SIP INVITE request;and
- if a received SIP INVITE request indicates support of the "timer" option tag, and contains the Session-Expires header field without "refresher" parameter, the UE must include the "refresher" parameter with the value "uac" in the Session-Expires header field of the SIP 2xx response to the SIP INVITE request, and must set the delta-seconds portion of the Session-Expires header field of the SIP 2xx response to the SIP INVITE request to the value indicated in the delta-seconds portion of the Session-Expires header field of the SIP INVITE request.

5.3. Supplementary Services

5.3.1. Supplementary Services Overview

The UE must support supplementary services as defined in 3GPP TS 24.173 [14], with the constraints described in this section. Especially, UE must support the supplementary services listed in <Table 5-2>.

<Table 5-2> List of Supplementary Services

Supplementary Services
Originating Identification Presentation 3GPP TS 24.607[23]
Originating Identification Restriction 3GPP TS 24.607[23]
Communication Forwarding Unconditional 3GPP TS 24.604[20]
Communication Forwarding on not Logged in 3GPP TS 24.604[20]
Communication Forwarding on Busy 3GPP TS 24.604[20]
Communication Forwarding on not Reachable 3GPP TS 24.604[20]
Communication Forwarding on No Reply 3GPP TS 24.604[20]
Barring of All Incoming Calls 3GPP TS 24.611[26]
Barring of All Outgoing Calls 3GPP TS 24.611[26]
Barring of Outgoing International Calls 3GPP TS 24.611[26] (Note 1)
Barring of Outgoing International Calls – ex Home Country 3GPP TS 24.611[26] (Note 1)

Supplementary Services
Barring of Incoming Calls – When Roaming 3GPP TS 24.611[26]
Communication Hold 3GPP TS 24.610[25]
Communication Waiting 3GPP TS 24.615 [27]
Customized Alerting Tone 3GPP TS 24.182[82]
Voice message box
Voice-Video call switch

Note 1) Barring of International Calls is a 3GPP Release 9 feature.

5.3.2. Originating Identification Presentation

The UE must indicate an incoming call by using SIP URI or Tel URI of the "P-Asserted-Identity" header in the invite message. If multiple "P-Asserted-Identity" headers exist, the URI of the most significant header must be used.

If the user info of the Tel URI or SIP URI is global-number-digits starting with "+82," Originating Identification must be displayed by replacing "+82" with "0".

(e.g. "01012345678" is used as call number to indicate "+821012345678")

5.3.3. Originating Identification Restriction

The UE must support the SIP procedures as specified in 3GPP TS 24.607 [23]. The service configuration as defined in section 4.10 of 3GPP TS 24.607 [23] is optional.

5.3.4. Communication Diversion

The UE must support the SIP procedures as specified in 3GPP TS 24.604[20] for communication diversion. The UE must support the “History-Info” header used for communication forwarding as shown in <Table 5-3>.

<Table 5-3> History-Info Header for Communication Forwarding

Condition	Cause	Final History-Info data received
Unconditional	302	<tel:user3;cause=302>;index=1.1
Busy	486	<tel:user3;cause=486>;index=1.1
Not-registered	404	<tel:user3;cause=404>;index=1.1
No reply	408	<tel:user3;cause=408>;index=1.1
Not-reachable	503	<tel:user3;cause=503>;index=1.1

5.3.5. Communication Barring

The UE must support the SIP procedure as specified in 3GPP TS 24.611[26].

5.3.6. Communication Holding

The UE must maintain the call in progress if a “488 (Not Acceptable Here)” response is received for a re-invite for Communication holding.

5.3.7. Customized Alerting Tone

The UE must support the Customized Alerting Tone of the forking model as specified in 3GPP TS 24.182 [25]. The UE must support forking model responses even if the UE support the early-session model, according to the request of mobile operator.

5.3.8. Voice-Video Call Switch

The UE sends Voice-Video call switch capability data by using the “P-TTA-VoLTE-

Info: avchange" header in a (re-)INVITE message or 200 OK message.

5.3.9. Communication Waiting

The UE must use the terminal based method as defined in 3GPP TS 24.615 [27], and support Alert-Info with the value defined in 4.4.1 of TS 24.615[27]. The service must be configurable by network signaling. UE must send "486 (busy)" response if a video call is received during a voice call, or a voice call is received during a video call.

5.4 Call Set-up Considerations

5.4.1. Integration of resource management and SIP

5.4.1.1. Loss of PDN Connectivity

If the Packet Data Network (PDN) connectivity between the UE and network is lost[5], the network must terminate all ongoing SIP sessions related to this UE, according to the procedures in section 5.2.8 of 3GPP TS 24.229 [15].

If the UE discovers that PDN connectivity had been lost, for example during a TAU procedure, then the UE must attempt to re-establish the PDN connection. When the UE regains PDN and IP connectivity, if the IP address has changed or the IMS registration expired during the period of absence of IP connectivity when the UE must perform a new initial registration to IMS.

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[5] when the P-CSCF receives an abort session request from the Policy and Charging Rules Function (PCRF)

5.4.1.2. Loss of media bearer and Radio Connection

If a Guaranteed Bit Rate (GBR) bearer used for voice fails to get established, or is lost mid-session[6] [7], then the network must terminate the session associated to the voice stream according to the procedures in section 5.2.8 in TS 24.229 [15] (P-CSCF must be informed about loss of bearer by the PCRF).

If a SIP session includes media streams, and if a dedicated bearer for any media stream fails to get established, or is lost mid-session[8], the UE must modify, reject or terminate the SIP session that the dedicated media bearer is associated with, according to section 6.1.1 in 3GPP TS 24.229 [15]. The UE can act differently per media type.

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

5.4.2. Audio Media Considerations

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

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

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

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

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

The UE must include the SDP mandatory parameter as defined in <Table 5-4> during the Voice call.

<Table 5-4> SDP Mandatory Parameter for Audio Media

Parameter	Value	Description
o	<username>= [CTN] (Note 1)	
m	audio <port> RTP/AVP <fmt list>	Dynamic payload type used
b	AS:xx(Note 2)	
	RS:0	
	RR:2500(Note 3)	
a	rtpmap:<payload type> EVS/16000/1	EVS Primary mode
	fmtp:<payload type> br=9.6-24.4 (주5)	
	fmtp:<payload type> bw=nb-swb (주5)	
	rtpmap:<payload type> AMR-WB/16000/1	AMR-WB octet-aligned mode
	fmtp:<payload type> octet-align=1	
	rtpmap:<payload type> AMR-WB/16000/1	AMR-WB bandwidth efficient mode
	fmtp:<payload type>	
	rtpmap:<payload type> telephone-	Wideband DTMF

Parameter	Value	Description
	event/16000	
	fmtp:<payload type> 0-15	
	rtpmap:<payload type> AMR/8000/1	AMR-NB octet-aligned mode
	fmtp:<payload type> octet-align=1	
	rtpmap:<payload type> AMR/8000/1	AMR-NB bandwidth efficient mode
	fmtp:<payload type>	
	rtpmap:<payload type> telephone-event/8000	Narrowband DTMF
	fmtp:<payload type> 0-15	
	sendrecv | sendonly | recvonly | inactive	
	ptime:20	Packet time
	maxptime:<max packet time> (주4)	Maximum packet time
	candidate	5.4.4절 참조

Note 1) Originator's username: CTN

Note 2) AS uses the value defined in Annex K of 3GPP TS 26.114[35] according to IP version, packetization, and codec.

Note 3) RR value is a recommendation.

Note 4) Maxptime: 120 ms

Note 5) defalut bit-rate and audio bandwidth for SDP offer/answer of the EVS Primary mode. The values are subject to the UE conditions.

The UE does not contain a hf-only parameter for an SDP offer, and must be able to select hf-only=0, 1 or omission for an SDP answer.

5.4.3. Video Media Considerations

The UE supports the Session Description Protocol (SDP) offer/answer format for video media as specified in section 6.2.3 of 3GPP TS 26.114 [35], with the restrictions included in this standard.

The UE must include the SDP mandatory parameter as defined in <Table 5-5> in an SDP offer or answer according to target quality, video processing capacity or display resolution of a Video call. The “a=imageattr attribute[9]” defined in 3GPP TS 26.114[35] and IETF RFC 6236[75] must be used for resolution negotiation.

<Table 5-5> SDP Mandatory Parameter for Video Media

Parameter	Value	Description
m	video <port> RTP/AVP <fmt list>	Dynamic payload type used
b	AS	Parameter definition according to video resolution in <Table 6-2>
	RS:0	
	RR:2500	
a	rtpmap:<payload type> H265/90000	H.265 720p
	fmtp:<payload type> profile-id=1	
	fmtp:<payload type> level-id=93	
	fmtp:<payload type> sprop-vps= ; sprop-sps= ; sprop-pps=	
	imageattr:<payload type> send [x=720,y=1280] recv [x=720,y=1280]	
	rtpmap:<payload type> H264/90000	H.264 VGA Portrait
	fmtp:<payload type> profile-level-id=42C016; packetization-mode=1; sprop-parameter-sets=Z0LAFukDwKMg,aM4G4g==	

9 a=framesize attribute cannot be used for resolution negotiation.

Parameter	Value	Description
	imageattr:<payload type> send [x=480,y=640] recv [x=480,y=640]	
	rtpmap:<payload type> H264/90000	H.264 VGA Landscape
	fmtp:<payload type> profile-level-id=42C016; packetization-mode=1;	
	sprop-parameter-sets= Z0KAFukBQHsg,aM4G4g==	
	imageattr:<payload type> send [x=640,y=480] recv [x=640,y=480]	
	fmtp:<payload type> profile-level-id=42C00C; packetization-mode=1; sprop-parameter-sets=Z0LADekCg/I=,aM4G4g==	H.264 QVGA Landscape
	imageattr:<payload type> send [x=320,y=240] recv [x=320,y=240]	
	rtpmap:<payload type> H263/90000	H.263 QCIF Landscape
	fmtp:<payload type> profile=0; level=10	
	imageattr:<payload type> send [x=176,y=144] recv [x=176,y=144]	

Parameter	Value	Description
	sendrecv | sendonly | recvonly | inactive	
	framerate:<frame rate> (주1)	SDP 내 포함된 H.264, H.263의 video framerate 중 최대값
	candidate	5.4.4절 참조

Note1) Framerate attribute is included for the backward compatibility with legacy UE and it must not affect the UE's behavior. The UE must set the framerate according to <Table 6-2>.

<Table 5-5a> H.265 codec parameters

Resolution	Parameter	Value
VGA profile-id=1 level-id=90	sprop-vps	AAAAAUABDAL//wFgAAADAAADAAADAAADAFoAAJXKSBI=
	sprop-sps	AAAAAUIBAgFgAAADAAADAAADAAADAFoAAKAPCAKB/llcpMkjbDSxFVJiYZfZ9eb8s+vCiMTl2yA=
	sprop-pps	AAAAAUQBwaVYESA=
VGA Landscape profile-id=1 level-id=90	sprop-vps	AAAAAUABDAL//wFgAAADAAADAAADAAADAFoAAJXKSBI=
	sprop-sps	AAAAAUIBAgFgAAADAAADAAADAAADAFoAAKAFAgHh/llcpMkjbDSxFVJiYZfZ9eb8s+vCiMTl2yA=
	sprop-pps	AAAAAUQBwaVYESA=
720p profile-id=1 level-id=93	sprop-vps	AAAAAUABDAL//wFgAAADAAADAAADAAADAF0AAJXKSBI=
	sprop-sps	AAAAAUIBAgFgAAADAAADAAADAAADAF0AAKACglAtH+WVykySNsNLEVUmJhl9n15vyz68KlxOXbI=
	sprop-pps	AAAAAUQBwaVYESA=
1080p profile-id=1 level-id=120	sprop-vps	AAAAAUABDAL//wFgAAADAAADAAADAAADAHgAAJXKSBI=
	sprop-sps	AAAAAUIBAgFgAAADAAADAAADAAADAHgAAKADwIAQ5/llcpMkjbDSxFVJiYZfZ9eb8s+vCiMTl2yA
	sprop-pps	AAAAAUQBwaVYESA=

5.4.4. Candidate Attribute

If a NAT equipment exists between a VoLTE UE and a CSCF and performs an ALG function, the UE IP of SIP/SDP messages changes to the NAT IP. As a result CSCF cannot recognize the UE IP.

The transmission of UE IP and Port information is described in this section. The SDP candidate attribute as defined in IETF RFC 5245 [81] must be used.

The Syntax of the candidate as defined in RFC 5245 [xx] is as follows.

```
candidate-attribute = "candidate" ":" foundation SP component-id SP
transport SP
priority SP
connection-address SP ;from RFC 4566
port ;port from RFC 4566
SP cand-type
[SP rel-addr]
[SP rel-port]
*(SP extension-att-name SP
extension-att-value)
foundation = 1*32ice-char
component-id = 1*5DIGIT
transport = "UDP" / transport-extension
transport-extension = token ; from RFC 3261
priority = 1*10DIGIT
cand-type = "typ" SP candidate-types
candidate-types = "host" / "srflx" / "prflx" / "relay" / token
rel-addr = "raddr" SP connection-address
rel-port = "rport" SP port
extension-att-name = byte-string ;from RFC 4566
extension-att-value = byte-string
ice-char = ALPHA / DIGIT / "+" / "/"
```

The Candidate generation is as follows.

candidate: 1 component-id transport priority private-ip port typ candidate-types

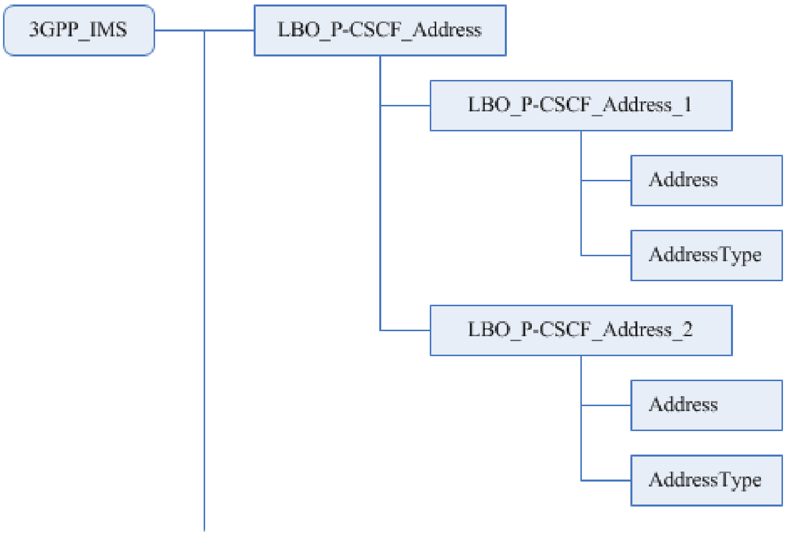
- component-id: A fixed number between 1 and 255 (1=RTP, 2=RTCP, fixed at 1)
- transport: Fixed as UDP
- priority: A fixed number between 1 and 2^31-1 (priority = (2^24)*(type preference) + (2^8)*(local preference) + (2^0)*(256 – component ID, exemplary value used)
- private-ip: Media processing IP (same IP as c line)
- port: Media processing port (same port as m line)

- typ: Fixed as typ
- candidate-types: Fixed as host

Example) a=candidate:1 1 UDP 2130706431 223.60.80.86 7010 typ host

5.5. Management Object

For IMS setting, the UE must support the 3GPP IMS Management Object (MO) defined in 3GPP TS 24.167 [68]. As LBO_P-CSCF-Address can be used in multiples, LBO_P-CSCF-Address X node (Figure 5-1) is defined as follows.



(Figure 5-1) Additional Definition of X Node of LBO_P-CSCF_Address

The Ext node of IMS MO must include the TTA VoLTE Management Object (MO) defined in <Table 5-6>.

An XML of TTA VoLTE MO is attached to this standard as a separate document.

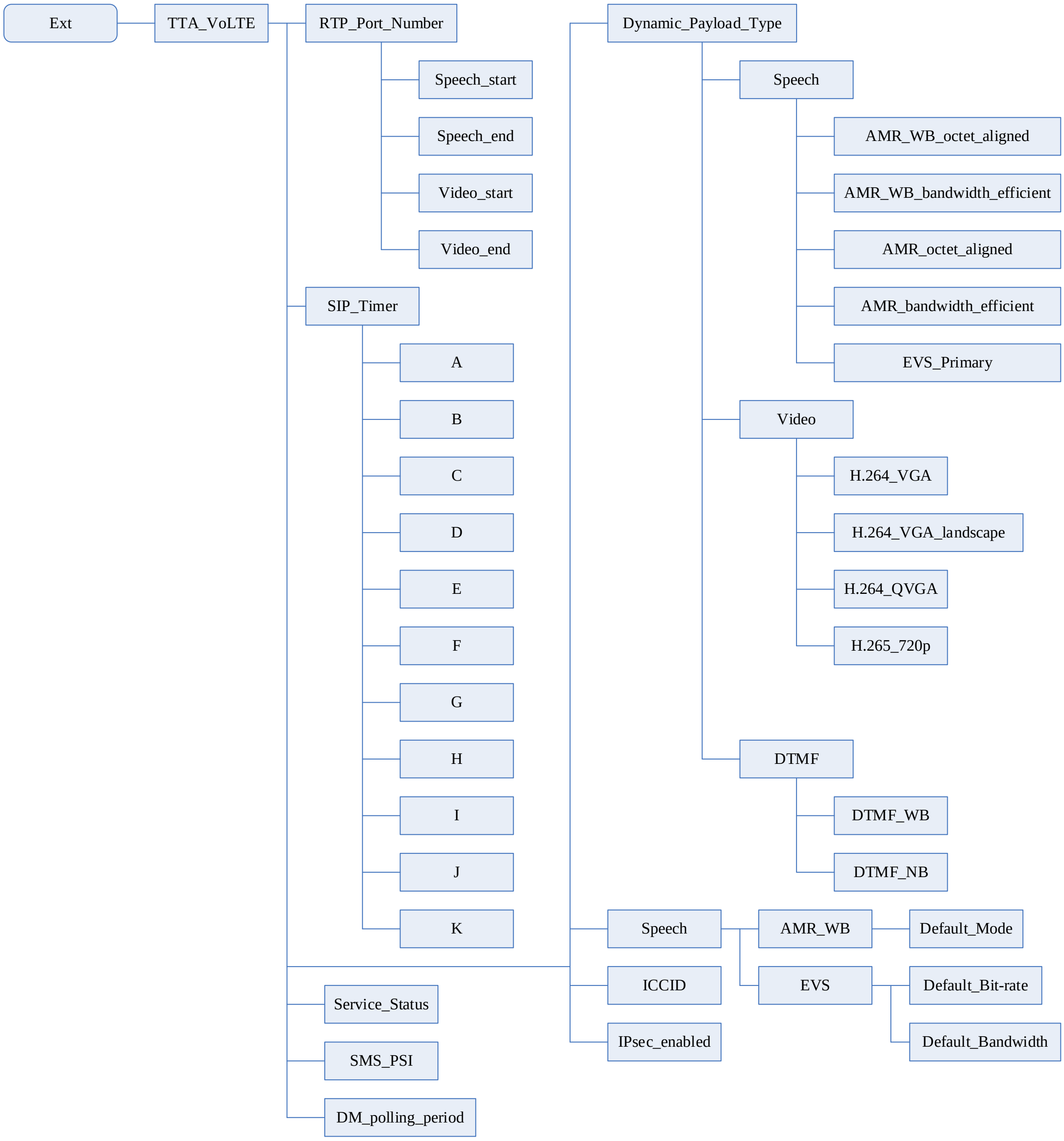
<Table 5-6> TTA VoLTE MO V2.0 Node

Name	Frequency	Format	Minimum access type	Description

Speech_start	One	int	Get, Replace	Minimum voice RTP port range
Speech_end	One	int	Get, Replace	Maximum voice RTP port range
Video_start	One	int	Get, Replace	Minimum video RTP port range
Video_end	One	int	Get, Replace	Maximum video RTP port range
A - K	One	int	Get, Replace	SIP timer A-K (millisecond)
AMR_WB_octet_aligned	One	int	Get, Replace	PT number
AMR_WB_bandwidth_efficient	One	int	Get, Replace	PT number
AMR_octet_aligned	One	int	Get, Replace	PT number
AMR_bandwidth_efficient	One	int	Get, Replace	PT number
H264_VGA	One	int	Get, Replace	PT number
H264_VGA_landscape	One	int	Get, Replace	PT number
H264_QVGA	One	int	Get, Replace	PT number
H265_1080p	One	int	Get, Replace	PT number
DTMF_WB	One	int	Get, Replace	PT number
DTMF_NB	One	int	Get, Replace	PT number
Default_Mode	One	int	Get, Replace	AMR-WB Default Mode (Note1)
Default_Bit-rate	One	chr	Get, Replace	EVS Primary Default Bit-rate (Note 2)
Default_Bandwidth	One	chr	Get, Replace	EVS Primary Default Bandwidth (Note 2)
Service_Status	One	bool	Get, Replace	If subscribed to VoLTE or not
SMS_PSI	One	chr	Get, Replace	SMS PSI string
DM_polling_period	One	int	Get, Replace	DM server polling period (second)
ICCID	One	chr	Get	ICCID of UICC card
IPsec_enabled	One	Bool	Get, Replace	IPsec On/Off

Note 1) Designation of AMR-WB default MODE (When there is no value, AMR-WB MODE8 is used). The default MODE must not affect SDP offer/answer.

Note 2) Designation of the default bit-rate and audio bandwidth of the EVS Primary mode. Default_Bit-rate and Default-Bandwidth must not have any influence on SDP offer/answer.



(Fig. 5-2) TTA VoLTE Management Object (MO) V2.0

6. IMS Media

6.1. General

This section describes the set of media functions as specified in 3GPP TS 26.114 [35].

6.2. Voice Media

6.2.1. Audio Codec

The UE must support the AMR voice codec as specified in 3GPP TS 26.071[29], 3GPP TS 26.090[31], 3GPP TS 26.073[30], and 3GPP TS 26.104[34] including all eight (8) modes and source rate controlled operations, as defined in 3GPP TS 26.093 [32]. The UE must be capable of operating with any subset of these eight (8) codec modes. The UE must support handling of CMR within RTP payload as specified in clause 7.5.2.1.2.2 of 3GPP Release 13 TS 26.114

If AMR-WB is offered, the UE must support the AMR-WB codec as specified in 3GPP TS 26.114[35], 3GPP TS 26.171[38], 3GPP TS 26.190[40], 3GPP TS 26.173[39], and 3GPP TS 26.204[42], including all nine (9) modes and source controlled rate operation 3GPP TS 26.193 [41]. The UE shall be capable of operating with any subset of these nine (9) codec modes. The UE must support handling of CMR within RTP payload as specified in clause 7.5.2.1.2.2 of 3GPP Release 13 TS 26.114 [35].

If EVS is offered, the UE must support the EVS codec as specified in 3GPP Release 12 TS 26.441[86], 3GPP Release 12 TS 26.442[87], 3GPP Release 12 TS 26.443[88], 3GPP Release 12 TS 26.445[89], 3GPP Release 12 TS 26.446[90], 3GPP Release 12 TS 26.447[91], 3GPP Release 12 TS 26.449[92], 3GPP Release 12 TS 26.450[93] and 3GPP Release 12 TS 26.451[94].

6.2.2. RTP Profile and SDP Considerations

6.2.2.1. RTP Profile

The RTP profile and AVP (Audio Video Profile) of voice media defined in IETF RFC 3551[57] must be used.

6.2.2.2. SDP Offer Considerations

If the AVP profile is used, the SDPCapNeg framework defined in IETF RFC 5939[64] must not be used in the SDP offer.

6.2.2.3. SDP Answer Considerations

The UE must be able to send the answer to the SDP offer which uses SDPCapNeg, and indicate the use of an RTP AVP profile[10].

6.2.2.4. SDP Session Modification Considerations

SDP media session modification must be only supported (session update during a call) when a re-invite (change of SDP IP address or port) is used during a call[11].

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

[11] Services that use a re-invite include call standby, Voice-Video call switch, ad-hoc conference and voice-video conference.

6.2.3. Data Transport

The UE must use RTP over UDP as specified in IETF RFC 3550[56] and IETF RFC 768[53], respectively, to transport voice and use symmetric RTP as defined in IETF RFC 4961[72].

6.2.4. RTCP Usage

The RTP implementation must include an RTP Control Protocol (RTCP) implementation according to IETF RFC 3550 [56]. The UE must use symmetric RTCP as specified in IETF RFC 4961[72].

The bandwidth for RTCP traffic must be described using the "RS" and "RR" SDP bandwidth modifiers at media level, as specified by IETF RFC 3556 [58]. Therefore, the UE must include the "b=RS" and "b=RR" fields in SDP and be able to interpret them. As shown in <Table 6-1>, the UE must be able to set the inactivity timer of RTP and RTCP. The requirements of RTCP usage are RS=0 & RR>0 during a call, and RS>=0 & RR>0 when on hold.

<Table 6-1> Audio RTP/RTCP Inactivity Timer Setting

	Inactivity timer use (Note 1)		Transmission interval
	During a call	On hold	
RTP	Used (10 sec)	-	-
RTCP	Not used	Not used	Under 5 sec

Note 1) Inactivity timer: terminated when no packets are received in a set period

The UE must set the sending frequency to a value calculated from the values of "RS" and "RR" SDP bandwidth modifiers according to the rules and procedures in IETF RFC 3556 [58]. The UE must support the transmission of RTCP packets formatted according to the rules in IETF RFC 3550 [56] and with the clarifications below:

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

RTCP compound packet format must be used. When sent, the compound packet must include one report packet and one source description (SDES) packet. When no RTP packets have been sent in the last two reporting intervals, a Receiver Report (RR) should be sent. Some implementations may send a Sender Report (SR) instead of a Receiver Report (RR), and this must be handled and accepted as valid.

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

Sender report (SR) and Receiver Report (RR) RTCP packet

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

Source description (SDES) RTCP packet

- Version and padding as described for SP packet
- The SDES item CNAME must be included in one packet[12].
- Other SDES items should not be used.

To be compatible and interwork with legacy UE, the UE must be able to receive all types of RTCP packets, according to the rules specified in IETF RFC 3550 [56].

6.2.5. AMR Payload Format Considerations

The Adaptive Multi Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) payload format defined in IETF RFC 4867[63] must be supported. The UE must support the band-efficient and octet-aligned format. Refer to IETF RFC 4867 [63] and section 5.4.2 for the SDP parameter. The UE must be able to 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 must request to receive one speech frame encapsulated in each RTP packet but must be able to accept any number of frames per RTP packet, up to the maximum limit of 6 speech frames per RTP packet[13].

6.2.6. EVS Payload Format Considerations

If EVS is supported, the UE must support the payload format specified in '3GPP TS

[12] 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.

[13] This means that the ptime attribute must be set to 20 and the max ptime attribute must be set to 120 in the SDP negotiation.

26.445'[89]'.

6.3. Video Media

6.3.1. Video Codec

The UE must support ITU-T Recommendation H.264 CBP (Constrained Baseline Profile) Level 2.2 as specified in section 5.2.2 of 3GPP Release 10 TS 26.114 [35]. The UE must also support ITU-T recommendation H.263 Profile 0, Level 10 as specified in section 5.2.2 of 3GPP Release 8 TS 26.114 [xx]. The UE must send an IDR (H.264) or I-frame (H.263) at an interval of two seconds or less.

The UE must send SPS, PPS data ahead of every IDR picture for 10 seconds from Video call setup to avoid possible video quality degradation due to SPS and PPS data loss on the network.

If H.265 is supported, the UE shall support 'ITU-T H.265' Main Profile, Main Tier Level 3.1 [97] specified in 5.2.2 of 3GPP TS 26.114 [35].

<Table 6-2> Parameter according to the Video Resolution

	codec	profile	level	Frame rate(fps)	Target bitrate (kbps)	b=AS value (IPv4)	b=AS value (IPv6)	profile-level-id
QCIF	H.263	0	10	7	48	64	66	-
QVGA	H.264	CBP	1.2	15	300	384	399	42C00C
VGA	H.264	CBP	2.2	15	512	639	653	42C016
VGA_ landscape	H.264	CBP	2.2	15	512	639	653	42C016
720p	H.265	Main	3.1	(주1)	(주1)	(주1)	(주1)	(주1)

Note 1) Determine after a trial (Apply the target bit-rate temporarily).

6.3.2. RTP Profile and Data Transport

The UE must use the RTP/AVP profile as specified in IETF RFC 3551 [57] for video media, and not use the RTP/AVPF profile.

The UE must comply with the SDP Offer Considerations in section 6.2.2.1 and SDP Answer Considerations in section 6.2.2.3.

6.3.3. RTCP Usage

The RTCP for Video media must comply with requirements for the Audio media RTCP as defined in paragraph 6.2.4. The UE must terminate video call if RTP inactivity timers of <Table 6-1> and <Table 6-3> expire.

<Table 6-3> Video RTP/RTCP Inactivity Timer Setting

	Inactivity timer (Note 1)		Transmission interval
	Call in progress	On hold	
RTP	Used (10 sec)	-	-
RTCP	Not used	Not used	5 seconds or less

Note 1) Inactivity timer: terminated when no packets are received in a set period

6.3.4. H.263 RTP Payload Format Considerations

The Payload format of H.263 defined in IETF RFC 2190 [83] must be supported.

6.3.5. H.264 RTP Payload Format Considerations

The Payload format of H.264 defined in IETF RFC 6184 [84] must be supported.

The UE must include packetization-mode=1 and sprop-parameter-sets in an SDP offer including H.264. The UE must be able to process responses to single NAL units and non-interleaved packetization.

6.3.6. H.265 RTP Payload Format Considerations

The UE shall support H.265 payload format defined in the IETF RFC 7798 [xx].

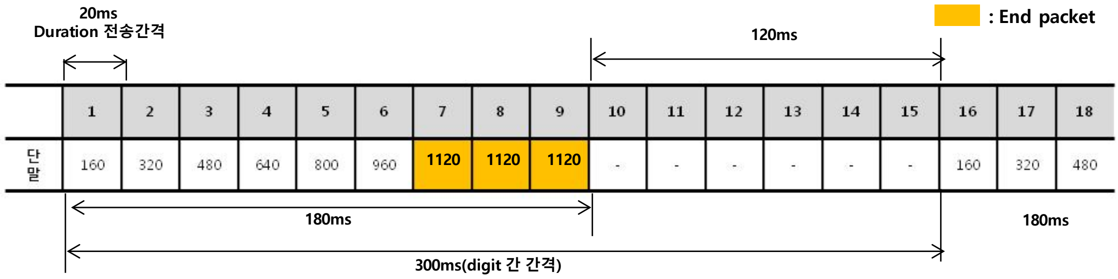
6.4. DTMF Event

The UE must support DTMF events as defined in Annex G of 3GPP TS 26.114[35] and the following event values.

<Table 6-4> DTMF Event Values

Item	Value	Description
DTMF duration	1120 (8k DTMF) 2240 (16k DTMF)	Sending 1120/2240 (accumulated) regardless of time pressed by user
End-of-Event frequency	3 times	Sending 3 times at intervals of 20 ms
Duration transmission interval	160 (8k DTMF) 320 (16k DTMF)	Increasing Duration transmission interval in one DTMF by 160/320
Volume	10	-10dBm

The interval between the first DTMF digit packets must be at least 300 ms or more.



(Fig. 6-1) 8k DTMF (Example)

7. Radio and Packet Core Feature Set

7.1. General

The LTE radio capabilities included in this standard are applicable to UE and network supporting FDD LTE only.

The UE that supports Conversational Video stream through a HSPA access must comply with radio and packet core feature set requirements defined in section 4, GSMA PRD IR.58 [76].

The UE must support the network acquisition functions required by mobile operators.

7.2. Robust Header Compression

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

7.3. LTE Radio Capabilities

7.3.1. Radio Bearers

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

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

The network must support the following combination of radio bearers:

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

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

EPS bearer usage is described in section 7.4.

7.3.2. DRX Mode of Operation

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

7.3.3. RLC Configurations

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

- Unacknowledged Mode (UM) for EPS bearers with QCI = 1
- Unacknowledged Mode (UM) for EPS bearers with QCI = 2
- Acknowledged Mode (AM) for EPS bearers with QCI = 5
- Acknowledged Mode (AM) for EPS bearers with QCI = 8/9

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

EPS bearer usage is described in section 7.4.

7.3.4. GBR and NGBR Services, GBR Monitoring Function

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

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

7.4. Bearer Management

7.4.1. EPS Bearer Considerations for SIP Signaling

The UE must operate using combined EPS/IMSI attachment during the attach request on an LTE network. The networks of each Mobile Operators must process the attachment as a combined attachment or EPS-only attachment according to their policy.

For SIP signaling, the IMS application must use the IMS well-known APN as defined in PRD IR.88 [67] and any other application must not use this APN.

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

If procedures in section 5.2.1 require the UE to register with IMS, and the PDN connection to the IMS well-known APN does not exist yet (e.g. when the PDN connection established during the initial attach is to an APN other than the IMS well-known APN then the UE must establish a PDN connection to the IMS well-known APN.

In addition, PDN Connection establishment can be caused by a SIP registration request. Sending a SIP registration request can cause PDN Connection establishment even if the IMS voice over PS Session indicator indicates that IMS voice over PS session is not supported.

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

7.4.2. EPS Bearer Considerations for Audio

For an IMS session request for a Voice call, a dedicated bearer for IMS-based voice must be created utilizing interaction with dynamic PCC. The network must initiate the creation of a dedicated bearer to transport the voice media.

The dedicated bearer for Voice must utilize the standardized QCI value (1) and have the associated characteristics as specified in 3GPP TS 23.203 [4]. Since the minimum requirement for the UE is the support of one UM bearer[14] which is used for voice (see section 7.3.1 and Annex B in 3GPP TS 36.331 [52]), the network must not create more than one dedicated bearer[15] for voice media. Therefore, the UE and network must be able to multiplex the media streams from multiple concurrent voice sessions.

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

7.4.3. EPS Bearer Considerations for Video

For an IMS session request for a Video call in E-UTRAN, each dedicated bearer resource for audio and video must be created by authorizing the flows utilizing dynamic PCC. The network must initiate the creation of dedicated bearer resources to transport

[14] A single bearer is used to multiplex the media streams from multiple concurrent voice sessions; this is necessary in some supplementary services.

[15] The sharing of a single GBR bearer for voice means that different QCI and/or ARP values are not possible for different voice streams.

the voice and the video media.

A dedicated bearer for conversational video streams can use a GBR or non-GBR bearer. If a GBR bearer is used it must utilize the standardized QCI value of two (2) and have the associated characteristics as specified in 3GPP TS 23.203[4].

For IMS session termination of a session using conversational media, the dedicated bearer resources must be deleted by withdrawing the authorization of the flows. The network must initiate the deletion of the bearer resources.

7.4.4. P-CSCF Discovery

The UE must perform SIP registration using a P-CSCF address in the following order of priority.

1. P-CSCF address via DM server
2. P-CSCF address via EPS

The UE must support P-CSCF discovery through a DM server.

The UE and packet core must support P-CSCF discovery via EPS. These are described in 3GPP TS 24.229 [15], Annex L.2.2.1 as optional features.

The UE must indicate a P-CSCF IPv6 address request and P-CSCF IPv4 address request when performing the following procedures (see section 7.3.1).

- During the initial attach when establishing PDN connection to the default APN,
- During the establishment of the PDN connection to the IMS well-known APN

The UE must use the P-CSCF addresses received during PDN connection establishment to the IMS APN, as defined in section 5.1 and 3GPP TS 24.229 [15].

8. Common Functionalities

8.1. IP Version

The UE and the network must support both IPv4 and IPv6 for all protocols that are used for the VoIP application (SIP, SDP, RTP, RTCP and HTTP). At PS attach, the UE must request the PDN type (IPv4v6) as specified in section 5.3.1.1 in 3GPP TS 23.401 [10]. If both IPv4 and IPv6 addresses are assigned for the UE, the UE must prefer the IPv6 address type when the UE discovers the P-CSCF.

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

8.2. Emergency Service

8.2.1. General

The UE must support emergency calls in the IMS domain. The UE must support the IMS emergency service as specified in 3GPP Release 9 TS 24.229[15], 3GPP Release 9 TS 23.167[3] and Chapter 6 and Annex H in 3GPP Release 9 TS 24.301[17].

The UE references the URN defined in <Table 8-1> and refers to 3GPP TS 24.229 [xx] Annex L for emergency calls without the URN.

<Table 8-1> Emergency Call URN Table

Emergency number	URN	Description
111	sos.country-specific.kr.111	National Intelligence Service
112	sos.police	National Police Agency
113	sos.country-specific.kr.113	Spy report
117	sos.country-specific.kr.117	School Violence report
118	sos.country-specific.kr.118	Cyber Terrorism report
119	sos.fire	National Emergency Management Agency
122	sos.marine	National Maritime Police
125	sos.country-specific.kr.125	Contraband report (Customs Service)

After the emergency PDN establishment, the UE attempts an emergency call without emergency registration, and immediately release emergency PDN at the end of the emergency call.

If there is no UICC in the UE, only a voice emergency call shall be provided and the UE shall use the IMEI to attempt an emergency call through an available WCDMA carrier. If WCDMA is not supported, the UE must do emergency attach to any available LTE system, and initiate emergency call without emergency registration. After the emergency call is finished, the UE must attempt to detach from the network. In this case, from P-Preferred-Identity, Contact header must indicate “Anonymous” as originating line identity with home domain “Anonymous.invalid”.

If there is a UICC in the UE, the UE shall provide a voice emergency call when a video emergency call is attempted for a non 112 and 119 call. The video emergency call shall only be provided for 112 and 119 calls.

The P-CSCF address must be requested during Emergency PDN request to use separate P-CSCF address for emergency service. If the P-CSCF address for emergency is provided by the network, it must be used for emergency service. If the P-CSCF address for emergency is not provided by the network, the normal P-CSCF address must be used for emergency service.

If the UE has already completed IMS registration, the UE must use the target number in the "From" header for the caller ID. If IMS registration has not been performed, the UE performs the outgoing emergency call in WCDMA mode. But when the emergency outgoing call is attempted in the airplane mode, the UE must disable airplane mode and support the emergency outgoing call in the IMS domain. After the termination of emergency call, the airplane mode must be kept disabled. If registration fails, the UE must fallback to WCDMA to originate the emergency call.

The UE must support the 3GPP IM CN subsystem XML body[16] as specified in section 7.6 of 3GPP TS 24.229[15].

8.3 Voice Calls and Smart Congestion Mitigation

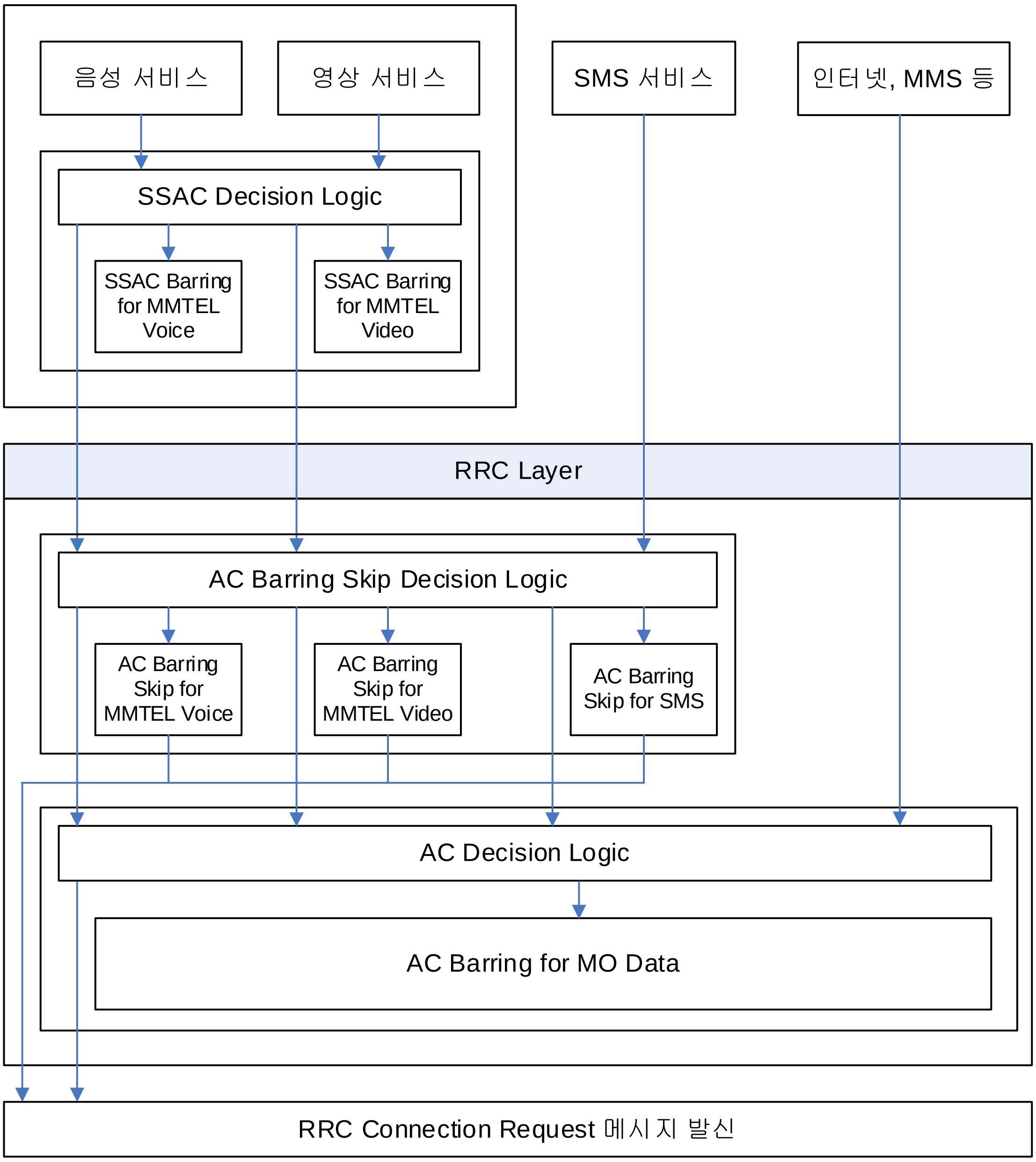
8.3.1 General Considerations

The UE must support SSAC (Service Specific Access Control) and SCM (Smart Congestion Mitigation), for control of network overload when accessing the network for VoLTE, Video Communications and SMS over IP services, as specified in 5.3.3 of 3GPP 36.331 [51] and J.2 of 3GPP 24.173 [14].

[16] This body is used to re-direct emergency calls to the CS domain.

8.3.2 Operation

The UE must be able to distinguish VoLTE, Video Communications, SMS over IP and internet data traffic when accessing the network as described in the figure 8-1. SIP messages related to Emergency Call/Registration must not be subject to this feature.



(Figure 8-1) Operation Flow of Smart Congestion Mitigation

9. SMS

The network must support either SMS over NAS or SMS over IP according to GSMA

IR.92 [79] and the UE must support both SMS over NAS and SMS over IP.

9.1. SMS Operation Mode Selection

The SMS operation mode in the terminal must be selected either SMS over NAS or SMS over IP as designated by the DM server.

9.2. SMS over NAS

The UE and network must support procedures as specified in 3GPP TS 23.272[9], 3GPP TS 23.221[6], and 3GPP TS 24.301[17]. The UE and network must perform in accordance with 3GPP TS 23.040[78], 3GPP TS 23.038[77], and WCDMA SMS specifications.

9.3. SMS over IP

The requirement for SMS transmission based on the IMS is defined in this section. The UE must support the SMS over IP procedures as specified in 3GPP TS 24.341 [19] according to the SMS specifications of each mobile operator for SMS over IP message transmission.

The UE sets the request URI by referencing PSI information sent by the DM server.

The UE must implement the roles of a SM-over-IP sender and a SM-over-IP receiver, according to the procedures in section 5.3.1 and 5.3.2 of 3GPP TS 24.341[19]. The status report capabilities, delivery reports, and notification of having memory available, according to section 5.3.1.3, 5.3.2.4 and 5.3.2.5 of 3GPP TS 24.341[19] must be supported.

10. MMS

The UE and network must perform the following processing according to items of OMA MMS 1.2 specification.

<Table 10-1> Common Specifications of MMS Items

Item	Common specification[1]	Note
MMBox	X	
Streaming RETRIEVAL	X	
SENDING MESSAGE	O	
Notification	O	
Retrieving an MM	O	
Forward Without Download	X	
Delivery Report	O	
Read Report	O	
Store-Update MM in MMBox	X	
Viewing MM Infromation from MMBox	X	
Uploading MM to MMBox	X	
Deleting MM from MMBox	X	
PIM (vCard, vCalendar)	O	Comply with section 7.1.2 PIM in OMA MMS 1.2 CONF
Video	O	
DRM	X	
Speech Audio	O	
Presentation (SMIL)	O	
Text	O	
Content-Name, Content-ID, Content-Location (Korean file name supported)	O	OMA MMS 1.3 Specification for only this item

Note 1) Items marked with “O” must be implemented in the UE development.

부 록 Ⅰ

GSMA IR.92 및 IR.94 삭제 항목

(The contents of GSMA IR.92 v10.0 & IR.94 v11.0 not selected in TTAK.KO-06.0357/R4)

Ⅱ-1. General Considerations

The TTA specification tries to focus as much as possible on UE's requirement and hence the changes "UE and IMS Core network" from "UE" from GSMA IR.92 v7.0 to GSMA IR.92 v9.0 are not adopted.

Editorial corrections and notes are not listed in this document including this annex.

XCAP is not used.

The mechanism to enable/disable preconditions is not adopted (i.e. no preconditions mechanism is adopted)

SIP URI in alphanmueric format is used.

Ⅱ-2. List of contents

Contents in GSMA PRD	Contents in TTAK_KO-06-0357_R3
IR92 2.2.1 SIP Registration Procedures	5.2.1 SIP Registration Procedures
UE and IMS core network must support network-initiated de-registration as defined in 3GPP TS 24.229.	
The UE must include the audio media feature tag, as defined in IETF RFC 3840, in the Contact header field of the SIP REGISTER request, using procedures of 3GPP TS 24.229	
The UE must set the URI of the From header field of the REGISTER request for user-initiated reregistration or for user-initiated deregistration to the public user identity which was used in the URI of the From header field of the REGISTER request that created the binding being refreshed or being removed. The UE must set the URI of the To header field of the REGISTER request for user-initiated reregistration and for user-initiated deregistration to the public user identity that was used in the URI of the To header field of the REGISTER request that created the binding being refreshed or being removed.	
For backwards compatibility the network must support all formats of URIs compliant with 3GPP TS 24.229.	
must immediately re-attempt an initial registration (as described above) when another P-CSCF is used.	

<table>
<tr><td>IR92 2.2.2 Authentication</td><td>5.2.2 Authentication</td></tr>
<tr><td colspan="2">This includes support for the P-Associated Uniform Resource Identifier (URI) header to handle barred IP Multimedia Public Identities (IMPUs)</td></tr>
<tr><td colspan="2">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. If the UE supports the Generic Authentication Architecture procedures specified in 3GPP TS 24.623, 3GPP TS 33.222 and 3GPP TS 24.109, then the UE must construct the Bootstrapping Server Function (BSF) address as defined in section 16.2 of 3GPP TS 23.003.</td></tr>
<tr><td colspan="2">The UE must support receiving 2xx response to the HTTP request without being challenged by 401 Unauthorized response</td></tr>
<tr><td>IR 92 2.2.3 Addressing</td><td>5.2.3 Addressing</td></tr>
<tr><td colspan="2">Alphanumeric SIP URIs</td></tr>
<tr><td colspan="2">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</td></tr>
<tr><td colspan="2">The UE and network must support geo-local numbers. The UE must set the “phone-context” parameter according to section 7.2A.10.3 in 3GPP TS 24.229.
Example of phone-context for geo-local number: if the visited network has MCC = 234, MNC = 15, and the home network has MCC = 567, MNC = 26, the “phone context” parameter is set to the string “234.15.eps.ims.mnc026.mcc567.3gppnetwork.org”</td></tr>
<tr><td colspan="2">The UE and network have the option to support geo-local numbers. If the UE supports geo-local numbers, it must set the "phone-context" parameter as with home local numbers, but prefixed by the "geo-local." string, according to the Alternative 8 in Section 7.2A.10.3 in 3GPP TS 24.229.</td></tr>
<tr><td colspan="2">The support of Globally Routable User agent URIs (GRUUs) by UE or network is not required.</td></tr>
<tr><td>IR 92 2.2.4 Call Establishment and Termination</td><td>5.2.4 Call Establishment and Termination</td></tr>
<tr><td colspan="2">The UE must be able to accept a SIP INVITE request without a Session Description Protocol (SDP) offer and the UE must include an SDP offer with audio media in the first non-failure reliable response to a SIP INVITE request without SDP offer.</td></tr>
<tr><td colspan="2">The UE must include the audio media feature tag, as defined in IETF RFC 3840, 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.</td></tr>
<tr><td colspan="2">The usage of preconditions is discussed in Section 2.4.</td></tr>
<tr><td colspan="2">If the user rejects an incoming call by invoking User Determined User Busy (UDUB) as described in 3GPP TS 22.030 [85], then the UE must send a SIP 486 (Busy here) response to the network.</td></tr>
<tr><td>IR 92 2.2.5 Forking</td><td>5.2.5 Forking</td></tr>
</table>

If the originating UE needs to release an early dialog, the UE must send a BYE request within the early dialog to be released, in accordance with section 15 in IETF RFC 3261, 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.	
The IMS core network can support sending and the UE must support receiving a SIP CANCEL request including a Reason header field with values of: • SIP; cause=200; text="Call completed elsewhere" • SIP; cause=603; text="Declined" for forked calls as defined in 3GPP Release 12 TS 24.229 .	
IR 92 2.2.6 The use of Signalling Compression	**(Section deleted)**
IR 92 2.2.7 Early media and announcements	**5.2.6 Early-Media and announcements**
The UE must behave as specified in section 4.7.2.1 of 3GPP Release 13 TS 24.628	
The UE must also maintain an early media authorization state per dialog as described in IETF RFC 5009	
IR 92 2.2.8 SIP Session Timer	**5.2.8 SIP Session 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;	
it is recommended that the UE does not include the "refresher" parameter in the Session-Expires header field of the SIP INVITE request.	
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	
IR 92 2.3.1 Supplementary Services Overview	**5.3.1 Supplementary Services Overview**
Terminating Identification Presentation Terminating Identification Restriction Barring of Outgoing International Calls - When Roaming Message Waiting Indication Ad-Hoc Multi Party Conference	
IR 92 2.3.2 Supplementary Service Configuration	**(Section deleted)**
IR 92 2.3.3 Ad-Hoc Multi Party Conference	**(Section deleted)**
IR 92 2.3.5 Message Waiting Indication	**(Section deleted)**
IR 92 2.3.7 Terminating Identification Restriction	**(Section deleted)**
IR 92 2.3.8 Communication Diversion	**5.3.4 Communication Diversion**
For CDIV service activation, deactivation, and interrogation (XCAP operations), the UE and IMS core network must support the XML rules for Call Forwarding Unconditional and the conditions, actions and elements listed in Table 2.2.	

<table>
<tr><td colspan="2">The UE and IMS core network shall support the XML rules as described in 3GPP TS 24.604 [20] section 4.9.1. The UE must support the History-Info header for identification of diverting parties at the terminating side and of diverted-to parties at the originating side. At the terminating side, a History-Info entry shall be used for the identificatio of the diverting party only if another History-Info entry exists that has assigned the next index in sequence AND includes a cause value. At the originating side only History-Info entries including a cause value shall be used for presentation of the diverted-to party.</td></tr>
<tr><td colspan="2">Table 2.2 Supported conditions, actions and elements in CDIV</td></tr>
<tr><td colspan="2">In addition to the requirements in section 2.3.2, when configuring settings for the Communication Diversion supplementary service the UE must configure only one of the following in an XCAP request:
- Communication diversion supplementary service activation, no-reply-timer or both.
- For the communication diversion services supported in this PRD, elements of one <rule> element for communication diversion supplementary service only.</td></tr>
<tr><td>IR 92 2.3.9 Communication Barring</td><td>5.3.5 Communication Barring</td></tr>
<tr><td colspan="2">For service activation, deactivation, and interrogation (XCAP operations), the UE and IMS core network must support the XML rules for Barring of All Incoming Calls, Barring of All Outgoing Calls and the conditions listed in Table 2.3. UE and IMS core network shall support the XML rules as described in 3GPP TS 24.611 [26] section 4.9.1.3.</td></tr>
<tr><td colspan="2">Table 2.3 Supported conditions in CB</td></tr>
<tr><td colspan="2">In addition to the requirements in section 2.3.2, when configuring settings for the Communication Barring supplementary service the UE must modify only one of the following in an XCAP request:
- Incoming communication barring supplementary service activation
- Outgoing communication barring supplementary service activation
- For the communication barring services supported in this PRD, elements of one <rule> element for communication barring supplementary service only.</td></tr>
<tr><td>IR 92 2.4.1 SIP Preconditions Considerations</td><td>(Section deleted)</td></tr>
<tr><td>IR 92 2.4.2.1. Loss of PDN Connectivity</td><td>5.4.1.1. Loss of PDN Connectivity</td></tr>
<tr><td colspan="2">(e.g. when the P-CSCF receives an abort session request from the Policy and Charging Rules Function (PCRF))</td></tr>
<tr><td colspan="2">This will trigger the network to initiate a new SIP signalling bearer in conjunction with the PDN connection establishment.</td></tr>
<tr><td>IR 92 2.4.3. Voice Media Considerations</td><td>5.4.2. Audio Media Considerations</td></tr>
<tr><td colspan="2">If multiple audio bandwidths are offered by the UE for speech communication, then the codec preference order must be as specified in clauses 5.2.1.5 and 5.2.1.6 of 3GPP Release 12 TS 26.114</td></tr>
<tr><td colspan="2">Unless otherwise preconfigured by the home operator, if a dedicated bearer for the media does not exist, the UE must consider itself not having local resources. If the UE has no local resources, the UE must not send media. See also section L.2.2.5.1B in 3GPP TS 24.229.</td></tr>
</table>

IR 92 2.4.4 Multimedia Considerations	(Section Deleted)
IR 92 2.5 SMS over IP	9.3 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, and the following functions: • answering of routing information query and obtaining the routing information according to the procedures in section 5.3.3.3 in 3GPP TS 24.341; and • transport layer interworking according to sections 5.3.3.4 of 3GPP TS 24.341.	
IR 92 3.1 General	6.1 General
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).	
IR 92 3.2.1 Codecs	6.2.1 Audio Codec
When transmitting using the AMR codec, the AMR-WB codec or the EVS AMR-WB IO mode codec, then the UE must be capable of aligning codec mode changes to every frame border, and must also be capable of restricting codec mode changes to be aligned to every other frame border, for example as described for UMTS_AMR_2 in 3GPP TS 26.103 based on the SDP offer-answer negotiation. The UE must also be capable of restricting codec mode changes to neighbouring codec modes within the negotiated codec mode set based on the SDP offer-answer negotiation.	
When receiving using the AMR codec, the AMR-WB codec or the EVS AMR-WB IO mode codec, then the UE and the entities in the IMS core network that terminate the user plane must allow codec mode changes at any frame border and to any codec mode within the negotiated codec mode set. As an exception, entities in the network that provide Circuit Switched (CS) interworking and apply Transcoder-Free Operation (TrFO) of Tandem-Free Operation (TFO) shall accept codec mode changes in accordance with the capabilities at the CS network side.	
Entities in the IMS core network that terminate the user plane supporting speech communication and supporting TFO and/or TrFO shall support: - AMR speech codec modes 12.2, 7.4, 5.9 and 4.75 as described in 3GPP TS 26.071, 3GPP TS 26.090, 3GPP TS 26.073, and 3GPP TS 26.104.	
Entities in the IMS core network that terminate the user plane supporting wideband speech communication and supporting TFO and/or TrFO shall support: - AMR-WB speech codec modes 12.65, 8.85 and 6.60 as described in 3GPP TS 26.171, 3GPP TS 26.190, 3GPP TS 26.173, and 3GPP TS 26.204.	
Entities in the IMS network that provide transcoding-free interworking to the CS network shall be capable of requesting the UE to restrict codec mode changes to be aligned to every other frame border and also be capable of requesting the UE to restrict codec mode changes to neighbouring codec modes within the negotiated codec mode set.	
IR92 3.2.2.4 SDP Bandwidth Negotiation	(Section deleted)

IR92 3.2.4 RTCP Usage	6.2.4 RTCP Usage
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.	
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.	
RTCP is controlled on a per session basis by the SDP offer/answer exchange as defined in 3GPP TS 26.114 [35] with the following clarifications: • If the UE receives an SDP offer that contains “b=RS” attribute set to zero, then the UE must set the “b=RS” attribute to zero in an SDP answer to that SDP offer. If the UE receives an SDP offer that contains “b=RR” attribute set to zero, then the UE must set the “b=RR” attribute to zero in an SDP answer to that SDP offer. If the UE receives an SDP offer that contains both "b=RR" and "b=RS" attributes set to zero, then the UE must not send RTCP packets and must consider RTCP to be disabled for the session. • If the UE received an SDP answer containing zero values in both of the “b=RS” and “b=RR” attributes, then (regardless of the values assigned to these attributes in the corresponding SDP offer) the UE must not send RTCP packets and must consider RTCP to be disabled for the session. • The UE must accept receiving RTCP packets for a session that the UE considers RTCP to be disabled. The UE is not required to process these received RTCP packets.	
IR92 3.2.5 Speech Payload Format Considerations	6.2.5 AMR Payload Format Considerations
IMS media gateway not supporting redundancy may limit the maxptime attribute to 80 in the SDP negotiation.	
The UE and the entities in the IMS core network that terminates the user plane must be able to sort out the received frames based on the RTP Timestamp and must remove duplicated frames, if present. If multiple versions of a frame are received, for example, encoded with different bit rates, then the frame encoded with the highest bit rate should be used for decoding.	
RTCP-APP must not be used for Codec Mode Requests (CMR).	
IR92 3.2.6 Jitter Buffer Management Considerations	(Section deleted)
IR92 3.2.7 Front End Handling	(Section deleted)
IR92 4.3.1 EPS Bearer Considerations for SIP Signalling and XCAP	7.4.1 EPS Bearer Considerations for SIP Signalling

<table>
<tr><td colspan="2">The UE must and the network can support T3396 IE in PDN Connectivity Reject and the UE must start the ESM back-off timer according to the value indicated by the T3396 IE as specified in 3GPP Release 12 TS 24.301. If the T3396 IE is not included in a PDN Connectivity Reject, and the PDN Connectivity Reject is for a standalone PDN CONNECTIVITY REQUEST, then the UE must apply a default value of 12 minutes for the ESM back-off timer for the cause values #8 "operator determined barring", #27 "missing or unknown APN", #32 "service option not supported", and #33 "requested service option not subscribed" as described in section 6.5.1.4.3 of 3GPP Release 12 TS 24.301.</td></tr>
<tr><td colspan="2">For XCAP requests, the UE must be preconfigured or provisioned by the home operator with the Network Identifier part of the APN for Home Operator Services to be used for these requests (see GSMA PRD IR.88 for more information).</td></tr>
<tr><td>IR92 4.3.2 EPS Bearer Considerations for Voice</td><td>7.4.2 EPS Bearer Considerations for Audio</td></tr>
<tr><td colspan="2">When the UE has an ongoing conversational voice call, the UE must follow the procedures for access domain selection related to “Persistent EPS bearer context” as specified in sections 5.5.3.2.4 and 5.5.3.3.4.3 of Release 10 of 3GPP TS 24.301, sections 5.1.3.1 and L.2A.0 of Release 10 3GPP TS 24.229, and section 8.2 of Release 10 3GPP TS 24.237.</td></tr>
<tr><td>IR92 4.4 P-CSCF Discovery</td><td>7.4.4 P-CSCF Discovery</td></tr>
<tr><td colspan="2">• during the initial attach when establishing PDN connection to the IMS well-known APN;
• during the attach procedure for emergency bearer services; and
• during the establishment of the PDN connection for emergency bearer services when already attached.</td></tr>
<tr><td colspan="2">when accessing non-emergency services, and must use the P CSCF addresses received during PDN connection establishment for emergency bearer services when accessing emergency services</td></tr>
<tr><td colspan="2">If the UE receives a Modify EPS Bearer Context Request message containing a list of P-CSCF addresses that do not include the address of the currently used P-CSCF, the UE must acquire a P-CSCF different from the currently used P-CSCF and initiate a new initial registration as described in section L.2.2.1C 3GPP Release 12 TS 24.229.</td></tr>
<tr><td>IR92 5.1 IP Version</td><td>8.1 IP Version</td></tr>
<tr><td colspan="2">and XCAP/HTTP</td></tr>
<tr><td colspan="2">and section 6.2.2 of 3GPP TS 24.301</td></tr>
<tr><td colspan="2">If only an IPv4 address or only IPv6 address is assigned by the network to the UE then the network must send ESM cause #50 "PDN type IPv4 only allowed" or #51 "PDN type IPv6 only allowed", respectively, to the UE and the UE must not request another PDN connection to the APN utilised in the initial attach or PDN connection establishment for the other IP version, as specified in section 6.2.2 of 3GPP Release 12 TS 24.301</td></tr>
</table>

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.	
IR92 5.2.1 General	8.2.1 General
When the UE has an ongoing emergency call, the UE must follow the procedures for access domain selection related to “Persistent EPS bearer context” as specified in sections 5.5.3.2.4 and 5.5.3.3.4.3 of Release 10 of 3GPP TS 24.301 and section 8.2 of Release 10 3GPP TS 24.237.	
Recognizing that some network operators will continue a parallel CS network whilst their IMS network is deployed, and that support of emergency calls with CS support may be a local regulatory requirement, emergency calls in the CS domain are addressed in Annex A.	
UEs and networks compliant with this profile must implement support for the 3GPP IM CN subsystem XML body as defined in section 7.6 of 3GPP TS 24.229.	
The SUPL enabled UE sends the emergency SUPL messages related to the UE detectable emergency session within the PDN connection for emergency bearer services. The SUPL enabled UE sends the emergency SUPL messages related to the non UE detectable emergency session within the PDN connection to the IMS well known APN. The UE selects the bearer to be used based on the TFTs of the bearers of the PDN connection. QCI of the selected bearer is provided by the network.	
If the UE: • receives the Emergency Service Support indication during EPS attach or tracking area updating procedures; • attempts an emergency registration with IMS; • receives a SIP 3xx, 4xx (except 401), 5xx or 6xx response to the emergency REGISTER request; and • is still in a tracking area that has received the Emergency Service Support indication; then the UE must perform the procedures defined in subclause 5.1.6.8.2 of 3GPP Release 9 TS 24.229.	
IR92 5.2.2 Interactions between supplementary services and PSAP callback	(Section deleted)
5.3. Roaming Considerations	(Section deleted)
5.4. Accesses in addition to E-UTRAN	(Section deleted)
5.5. Data Off and Services Availability	(Section deleted)
IR94 2.4.2 Video Media Considerations	5.4.3 Video Media Considerations
Coordination of Video Orientation (CVO) as specified in 3GPP Release 12 TS 26.114 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 Release 12 TS 26.114.	

Unless pre-configured otherwise by the home operator, if a dedicated bearer for the media does not exist, the UE must consider itself not having local resources. If the UE has no local resources, the UE must not send media. See also section L.2.2.5.1B in 3GPP TS 24.229.	
IR94 3.3.1 Video Codec	6.3.1 Video Codec
COMPLETELY REPLACED BY CONTENTS IN 6.3.1	
IR94 3.3.2 RTP Profile and Data Transport	6.3.2 RTP Profile and Data Transport
COMPLETELY REPLACED BY CONTENTS IN 6.3.2	
IR 94 3.3.3 RTCP Usage	6.3.3 RTCP Usage
COMPLETELY REPLACED BY CONTENTS IN 6.3.3	
IR94 3.3.4 RTP Payload Format Considerations for Video	6.3.4/6.3.5/6.3.6 H.263/H.264/H.265 RTP Payload Format Considerations
COMPLETELY REPLACED BY CONTENTS IN 6.3.4~6.3.6	

부 록 II-1

지식재산권 확약서 정보

해당사항 없음

※ 상기 기재된 지식재산권 확약서 이외에도 본 표준이 발간된 후 접수된 확약서가 있을 수 있으니, TTA 웹사이트에서 확인하시기 바랍니다.

부 록 Ⅱ-2

시험인증 관련 사항

Ⅰ-2.1 시험인증 대상 여부

해당사항 없음

Ⅰ-2.2 시험표준 제정 현황

해당사항 없음

부 록 II-3

본 기술보고서의 연계(family) 표준

I-3.1 사업자 간 UICC 이동성 제공을 위한 VoLTE 단말 규격 (TTAK.KO-06.0357)

본 기술보고서는 사업자 간 UICC 이동성 제공을 위한 VoLTE 단말 규격 제4차 개정본의 내용을 영문으로 표기한다.

부 록 II-4

참고 문헌

해당사항 없음

부 록 II-5

영문기술보고서 해설서

1 IMS 기능

본 절에서는 Gm 참조점(Reference Point)에서 필수적으로 요구되는 IMS 프로파일을 제시한다.

2 IMS 미디어

본 절은 '3GPP TS 26.114'[35]에서 규정된 일련의 미디어 기능을 정의한다.

3 Radio and Packet Core Feature Set

본 절은 LTE 무선 기능에 대한 요구사항으로 FDD LTE 만을 고려한다. HSPA access를 통한 대화형 비디오 서비스를 지원하는 UE는 'GSMA PRD IR.58 [76] 4 절에 정의된 radio 및 packet core feature set에 대한 요구 사항을 만족해야 한다.

4 공통기능

본 절은 IP 버전, 긴급 호 서비스에 대한 요구사항을 정의한다.

5 SMS

본 절은 SMS 동장 방식에 대해 정의한다. SMS over SGs 및 SMS over IP 동작을 위한 요구사항을 정의한다.

6 MMS

본 절은 OMA MMS 1.2' 규격 항목별 지원 필수 여부를 정의한다.

부 록 II-6

기술보고서의 이력

판수	채택일	기술보고서번호	내용	담당 위원회
제1판	2016.08.31	제정 TTAR-06.0171	TTAK.KO-06.0357 3차 개정본에 대한 영문본 및 GSMA IR 문서와의 비교 작성	모바일응용서비스 프로젝트그룹 (PG910)
제2판	2018.x.xx	개정 TTAR-06.0171/R1	TTAK.KO-06.0357 4차 개정본에 대한 영문본 및 GSMA IR 문서와의 비교 작성	모바일응용서비스 프로젝트그룹 (PG910)