THE VOICE OF 5G AND LTE FOR THE AMERICAS

The Evolution of 3GPP Communications Services

VoLTE and RCS Technology Evolution and Ecosystem

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EXECUTIVE SUMMARY

Voice over LTE (VoLTE) is being rapidly deployed in many parts of the world and is on its way to becoming mainstream in many operator networks. Video over LTE (ViLTE) is a complementary conversational video service specified by GSMA and is also being adopted worldwide, albeit at a slower pace. The expectation is that in a few years, both VoLTE and ViLTE will form the backbone of IP-based telecommunications just as voice was an integral part of 2G and 3G networks.

VoLTE enables High Definition (HD) call quality, helping operators compete against over-the-top (OTT) VoIP providers. In addition, VoLTE typically offers lower delay and higher capacity compared to OTT VoIP services, which utilize best-effort bearers that in turn adversely affect service quality.

VoLTE implementation can take different paths based on factors such as each operator’s spectrum availability, voice strategies, technology architecture, business objectives and market conditions. As operators determine their VoLTE deployment strategy, they may choose an evolutionary approach that starts with Circuit Switched Fallback (CSFB) and, when IP Multimedia Subsystem (IMS) is deployed, introduce SR-VCC once VoLTE is launched prior to ubiquitous LTE coverage. Other operators may choose to achieve ubiquitous LTE coverage prior to launching VoLTE service.

Similar to VoLTE, Rich Communications Services (RCS) Messaging takes advantage of IMS control capabilities, encompassing a rich multimedia service portfolio, including multimedia messaging, chat, file transfer, privacy, buddy lists and presence.

This paper discusses the key aspects of VoLTE/RCS coexistence and evolution:

- **Introduction** – Provides an overview of VoLTE and RCS Messaging and how standards and interoperability are evolving.

- **VoLTE Considerations** – Discusses the impacts and considerations for VoLTE in terms of service performance, roaming and regulatory aspects and explains the concepts of enhanced (eSRVCC), video (vSRVCC) and reverse (rSRVCC) voice call continuity features when an operator does not have ubiquitous LTE coverage, but wants to offer VoLTE and ViLTE services.

- **VoLTE and RCS Coexistence** – Discusses IMS as a platform for managing coexistence.

- **Overview of Enhanced Voice Services (EVS) Codec** – Discusses details of the EVS codec, including the various modes such as wideband (WB), super-wideband (SWB) and full-band (FB) and the ability of EVS to support a range of bitrates.

- **Regulatory Aspects of VoLTE in U.S. and Canada** – Provides a detailed overview of regulatory aspects of VoLTE such as e911 location requirements, text-to-911, emergency and alerting services, lawful interception and government priority services.

- **VoLTE Roaming** – Discusses both the Local Breakout (LBO) and the newly approved S8 Home Routed (S8HR) architecture models. The billing and charging impacts, emergency and lawful interception aspects of both approaches are also discussed.

- **Evolution of Communication and Messaging: A Path to RCS** – Discusses recommendations and best practices for RCS Messaging implementations and service offerings, the relation with other communication services such as OTT and WebRTC services and finally, RCS as a platform for innovation via API exposure.
INTRODUCTION

INTRODUCTION TO VOLTE AND RCS

Operators deploying VoLTE will tend to also deploy RCS Messaging as one step in their migration to an all-IP network. RCS adds IP-based messaging services that are an evolution to legacy SMS and MMS messaging.

The drivers and the timeline for the deployment of RCS Messaging vary widely by operator. Some have chosen to deploy RCS Messaging prior to VoLTE as a strategy to maintain messaging service relevance for their subscribers who are exposed to OTT applications enabled by the smartphone revolution. When VoLTE, including video, is initially introduced, it provides Quality of Service (QoS) and regulator-required services such as E-911. VoLTE also is complemented by CS services for specific scenarios. Depending on market drivers, RCS-enabled voice over Wi-Fi calling can also be used to complement VoLTE and existing CS services. The drivers and timelines of this evolution make it necessary for vendors to support different coexistence combinations, which depend on each operator’s network architecture and business goals.

RCS introduces an important all-IP communication evolution enabler in capability discovery. The end user will always be aware of who in their contact book is able to communicate using new services. This awareness avoids the frustration of trial and error and enhances the user experience in ways that encourage usage. It also gives operators the freedom to introduce new services at their desired pace and maintain the maximum service interoperability.

EVOLUTION OF COMMUNICATIONS SERVICES ENABLED BY VOLTE AND RCS

VoLTE and RCS Messaging drive the evolution of telephony, messaging and contact management (address book) services, enabling operators to offer new and enriched communication experiences to consumers, enterprise and government users.

Operators can deploy and launch VoLTE and RCS Messaging services utilizing the same device and network infrastructure. This approach ensures an efficient service and network-evolution path enabling continuous development of new communication services utilizing IMS.

The present and ongoing evolution of VoLTE and RCS Messaging will enable delivery of advanced communication services on all types of IP devices and broadband networks. This evolution also maintains interoperability and reachability with legacy communication services.

IMS delivers a modular and extendable service framework that simplifies the development and launch of new communication services such as VoLTE and RCS Messaging. Core IMS functionality and infrastructure will be used to deliver several communication services utilizing one common base.

Building on today’s global reachability on voice and SMS, the inter-operator connections, known as network-to-network interfaces (NNIs), will be extended to support new communication services. The evolved communication services form the basis for new, innovative third-party and OEM-specific applications.

An operator should be able to implement VoLTE and RCS Messaging features at its own pace, but at the same time avoid trial and error that undermines the customer experience. RCS avoids this problem by providing a capability-discovery mechanism, which shows users which communication services and applications that another person also has the capability of using. Another advantage is that it enables seamless communication between users.
STANDARDS EVOLUTION

OVERVIEW

The deployment of LTE networks generated interest in IMS-based telephony for mobile networks defined as VoLTE. A simple profile for IMS-based telephony service over LTE was envisioned by the One Voice industry initiative taking into consideration IMS MultiMedia Telephony Service (MMTel) standards defined in 3GPP Release 7 and specified in TS 24.173. One Voice also took into account the Supplementary Services Specifications developed by ETSI TISPAN, which were transferred to 3GPP.

The One Voice specification was submitted to the GSM Association (GSMA) for further enhancement as IMS-based VoLTE solution. Based on the One Voice specification, GSMA developed the Permanent Reference Document (PRD) IR.92. Also known as IMS Profile for Voice and SMS, IR.92 specifies a minimum mandatory set of features, defined in 3GPP Release 8, that wireless devices and networks require for the implementation of VoLTE as an interoperable, high-quality, IMS-based telephony service over LTE. While VoLTE is suitable to replace existing CS voice, GSMA PRD IR.92 also profiles capabilities to support the CS-IMS voice transition.

During this time, standards bodies such as the Internet Engineering Task Force (IETF) and the Open Mobile Alliance (OMA) were developing enabling technologies such as Presence, Converged IP Messaging (CPM) and other supporting enablers to be used as the next generation messaging framework. It became clear that there was a need to evolve the existing, xMS-based messaging services to provide a richer user experience and allow the development of new services based on it.

RCS started as an industry initiative by several major operators and OEMs in 2007. They wanted to explore next generation, all-IP multimedia communications that would enable the rapid adoption of mobile applications and services, providing an interoperable, convergent and rich communication experience based on IMS architecture. When this initiative grew too big due to overwhelming industry interest, it moved to GSMA to leverage its organizational supports and project management skills to make it a true global project. In GSMA, the Global Specification Group under Network 2020 Program is responsible for the development and maintenance of relevant RCS specifications.

THE HISTORY OF RCS DEVELOPMENT:

- 2007-06 Started as an industry initiative
- 2008-07 Moved into GSMA EMC (the current PSMC)
- 2008-12 GSMA released RCS 1.0
- 2009-06 GSMA released RCS 2.0
- 2009-12 GSMA released RCS 3.0
- 2010-06 First NA RCS workshop (GSMANA #51) in Miami
- 2010-12 GSMA released RCS 4.0 (all-IP release)
- 2012-04 GSMA stopped existing RCS development and formed RCE to focus on RCS 2.0 deployment
- 2011-06 Second NA RCS workshop (GSMANA #54) in Montreal
• 2011-08 CT-AGO formed NA RCS TF
• 2012-01 GSMA formed RCCTF
• 2013-01 Third NA RCS workshop (GSMANA #59) in St. Pete
• 2013-03 GSMA released RCS 5.0
• 2013-11 GSMA released RCS 5.1v4 (NA RCS profile base)
• 2014-09 Fourth NA RCS workshop (GSMANA #64) in Atlanta
• 2016-03 GSMA released RCS 6.0
• 2016-03 Fifth NA RCS workshop (GSMANA #69) in Anchorage

VOLTE AND VILTE

GSMA PRD IR.92 addresses IMS’s basic capabilities and supplementary services for telephony, and how these service capabilities can be set up in a UE via Ut interface using XCAP. It also addresses related real-time media negotiation, transport and codecs, the latest update being the inclusion of EVS as one of the supported codecs for VoLTE.

Based on its alignment with 3GPP, IMS procedures as defined in 3GPP TS 24.229, GSMA PRD IR.92 also profile requirements for SMS over IMS and SMS over SGs for a cohesive SMS over IP solution (SMSoIP), according to procedures defined in 3GPP TS 24.341. Moreover, GSMA PRD IR.92 lists essential Evolved Universal Terrestrial Radio Access Network (E-UTRAN) and Evolved Packet Core (EPC) features required for service interoperability. GSMA PRD IR.92 provides a comprehensive list of standards required for defining VoLTE service.

IMS telephony can be deployed over LTE without relying on any legacy infrastructure (packet or CS). Even so, GSMA PRD IR.92 considers additional features that can be implemented for the UEs and networks that may wish to support concurrent CS coverage by their IMS network. This is referred to as IMS Centralized Services (ICS) and provides service consistency for all mobile subscribers. Distinct from this is the critical requirement for service continuity for subscribers who are on a VoLTE call but leave LTE coverage due to travel or movement within large buildings. The service continuity is fulfilled by Single Radio Voice Call Continuity (SR-VCC) or its enhanced version, eSRVCC.

GSMA PRD IR.92 provides the basis for the definition of other services such as GSMA PRD IR.94 as IMS Profile for Conversational Video Service, as well as GSMA PRD IR.58 as IMS Profile for Voice over HSPA (VoHSPA). IR.92 also specifies region-specific features such as Global Text Telephony (GTT) and network interactions required to support the transfer of text media (Real Time Text), such as guidelines for dedicated bearer setup and teardown.

Similar to GSMA PRD IR.92, PRD IR.94 defines a minimum mandatory set of features required to implement and guarantee an interoperable, high-quality IMS-based conversational video service over LTE, commonly referred as Video over LTE (ViLTE).

Regulatory aspects such as those for emergency services are specified in GSMA PRD IR.92, which refers to 3GPP Release 9 IMS Emergency Services, namely TS 24.229, TS 23.167 and TS 24.301. Besides addressing emergency services for VoLTE, PRD IR.92 serves as reference for ViLTE, according to GSMA PRD IR.94, as well as for VoHSPA, according to GSMA PRD IR.58. However, GSMA PRD IR.94 has been updated to conform to procedures defined in TS 23.167 in Release 11 in support of conversational video services. In addition, PRD IR.92 points to similar requirements applicable to the CS domain.
RCS

As the main RCS specification, GSMA PRD RCC.07 defines the capabilities needed for RCS services and the standards based technical realizations. For IP-based voice services, GSMA PRD RCC.07 endorses the services defined in GSMA PRD IR.92, IR.94 and IR.51, with defined service tags for capability discovery.

In addition, for non-multimedia telephony services it endorses OMA-developed service enablers and relevant 3GPP specs for technical realization. Figure 1 illustrates the set of services specified in RCS 6.0.

The suite of services specified in GSMA PRD RCC.07 can also be supported with an RCS RESTful API model, which is captured in a set of requirements included in GSMA PRD RCC.13, and standardized by OMA. Based on these requirements, OMA defines a set of lightweight, Web-friendly APIs exposing RCS capabilities to Internet-based developers. RCS-related APIs also allow mobile operators and other service providers to expose useful information and capabilities to application developers.

WebRTC and RCS can be integrated to complement each other, such as to enable vertical solutions for telemedicine, customer service and video surveillance.

To provide a level of future proofing, the RCS framework includes service extensibility. This allows operators to use RCS as a platform to develop new services in a timely manner, with no impact on their existing RCS services. If needed, these service extensions can be added to the main RCS specs for global adoption.
For inter-service provider aspects, GSMA PRD IR.92, IR.94 and RCC.07 refer to GSMA PRD IR.88, IR.65, IR.90 and RCC.54. These specifications document IMS-NNI-specific details associated with the IMS-based services:

- GSMA PRD IR.88 defines LTE and Evolved Packet Core (EPC) roaming guidelines required for interworking and definition of mobile network capabilities when customers roam. Consequently, IR.88 provides technical roaming guidelines for VoLTE using the local breakout (LBO) and S8HR options in the LTE roaming architecture. In the LBO roaming scenario, IMS traffic is broken out from the packet data network gateway (P-GW) in the visited network traversing the IMS-NNI to the functions in the home network and taking advantage of optimized media routing methods defined in 3GPP standards. In the S8HR roaming architecture, the IMS PDN is anchored on the P-GW in the HPMN with the IMS traffic traversing the S8 interface between the VPMN and HPMN (hence the name). Since all the IMS network elements reside in the HPMN, an IMS-NNI may not be needed with the current solution being developed by 3GPP for handling emergency calls and Lawful Intercept (LI) for roaming situations. GSMA PRD IR.88 also covers relocation to CS voice and SMS services using CSFB as defined in 3GPP TS 23.272 when VoLTE or SMSoIP is not available. Furthermore, GSMA PRD IR.88 specifies capabilities to facilitate roaming for IMS-based services, such as VoLTE, and is based on the IMS Access Point Name (APN).

- GSMA PRD IR.65, or IMS Roaming and Interworking Guidelines, expands on details related to network-level roaming and interworking between two IMS networks and use of the IMS NNI as defined in 3GPP TS 29.165. PRD IR.65 also specifies operational requirements for VoLTE/ViLTE and RCS associated with routing of signaling and media over IMS, considering that it should be as optimal as done for CS communications. VoLTE roaming guidelines are specified for both roaming models (LBO and S8HR)

- GSMA PRD IR.90 defines interworking guidelines for RCS. GSMA PRD IR. 90 helps service providers select the optimal solution for service interconnection at the IMS NNI based on the possible service alternatives associated with RCS. This further provides general service interoperability aspects related to RCS as input to GSMA PRD IR.65. PRD IR.90 also defines the interworking that service providers could use for setting up mutual agreement for IMS-based service interconnection.

- GSMA PRD RCC.54 defines guidelines for interconnection between service providers for the purposes of interchanging VoLTE, ViLTE and RCS Messaging inter-service-provider traffic. The current version of the PRD is based on a set of features or profile defined for North American deployments. Interconnection aspects for other regional RCS profiles will be captured in future versions. The document supports interconnection via direct bilateral connectivity and connectivity via internetwork packet exchange (IPX). However, the current scope addresses guidelines only for service providers rather than IPX providers. GSMA PRD IR.34 facilitates guidance on procedures for IPX providers associated with the inter-service-provider IP backbone.

In addition to the specifications mentioned above, other GSMA interworking guidelines – such as PRDs IR.67 and IR.77 – are used to address issues related to addressing, quality of service (QoS) and security, all of which rely on 3GPP standards.

The verification of the inter-service-provider aspects is documented in GSMA PRD RCC.51, which defines end-to-end testing of VoLTE, ViLTE and RCS Messaging over IMS. It also covers testing of NNI based on
service interconnection recommendations specified in GSMA PRD IR.90 considering the IP and SIP connectivity, RCS media connectivity, as well as addressing and routing, including E.164 number mapping (ENUM) and DNS functionality. The set of tests defined in GSMA PRD RCC.51 requires service providers to set up a mutual agreement populating the “Interworking Form for IMS-based Services” defined in GSMA PRD IR.90, and the “Interworking Template Agreement” specified in GSMA PRD AA.69. The GSMA PRD RCC.58 specifies the RCS end-to-end test cases for the North America RCS profile, which aligns with North America RCS Implementation Guidelines defined in GSMA PRD RCC.59.

VOLTE AND RCS TECHNOLOGY MATURITY AND ECOSYSTEM

RCS

South Korea launched the world’s first commercial VoLTE services in 2012. The same year, MetroPCS launched it in select U.S. markets), followed by some European and other North American operators. One South Korean operator said it had 1 million Joyn subscribers by mid-February 2013 (within 50 days of launch). According to GSMA, approximately 47 wireless carriers in 34 countries launched RCS services as of 1Q 2016.

A universal RCS profile, with the noticeable inclusion of Google among the supporters, made the news in early 2016. Future Android devices with embedded client support for GSMA’s enriched calling is seen as a strong endorsement because it aligns operator and device capabilities in same direction. The RCS ecosystem is further strengthened by Google’s upcoming release of open client source code and APIs. Adopting the expected universal RCS client will enable users to access RCS services such as group chat and high-resolution photo sharing, among others. That will enhance SMS, which is used by more than 4 billion people worldwide.

A key difference in the RCS ecosystem is the arrival in North America of embedded RCS clients, which are now deployed as standard with a number of Android handsets. This changes the game because users are presented with a native messaging application that just happens to be RCS. The subscriber adoption rates are now higher due to the success for these new devices.

RCS is not very different from OTT in the type of services per-se, which are well known and used by a wide range of multimedia. However, there are a few key differentiators:

1. The possibility of RCS enriched calling making use of telco-provided reliable bearers (i.e., traditional CS calling and VoLTE), which guarantee QoS.
2. The availability of lawful intercept interfaces and procedures.
3. Single IMS registration and the ability to leverage QoS for RCS over OTT applications. This capability is especially useful when many applications are competing for IP bandwidth.
4. With native clients able to access the SIM card, there is no need for additional user authentication. Instead, the RCS service “just works” out of the box. Another key benefit is the fact that they are default clients that the user can simply start using without any additional interactions.

A noticeable RCS trend is the increase in the size of media shared between the users. When file transfers reach GB range, operators will be forced to implement HTTP-based file transfers rather than the standard MSRP-based one.
RCS also enables support for multiple devices and allows operators to leverage their IMS investments to deliver advanced messaging capabilities today and creates enablers for future rich communication services.

RCS specifications have evolved rapidly since the first RCS-e services were launched and branded as Joyn. The current specifications are mature enough for operators to launch and deploy common base messaging technologies, such as standalone messaging and chat-based services with presence used for capability discovery. There are two different technologies to enable messaging services: OMA CPM and SIMPLE IM. The agreement among most North American operators is to deploy RCS IP messaging based on OMA CPM.

North American operators have been driving toward making RCS the default native experience for messaging services because Joyn’s OTT downloadable and preloaded clients did not fare well compared to other OTT clients. RCS as the default messaging client, paired with VoLTE to share the same IMS registration and IMS stack, will continue to create efficiencies on the device itself and on the network. This approach also will simplify the deployment of RCS services.

Regional marketing requirements and deployment schedules led to adoption of different version of RCS specs that endorse different OMA service enablers. European operators were under pressure to combat an OTT messaging invasion, which resulted in an early RCS deployment based on RCS 2 specifications. These offer a mix of legacy and IP-based messaging and OTT-like IP voice services. Meanwhile, North America decided early on that all-IP messaging and QoS-controlled IP voice services should be offered to replace similar existing services based on RCS 5.1.

RCS 2.0 endorses an older OMA SIMPLE IM messaging enabler, while RCS 5.1 endorses the newer OMA CPM messaging enabler. OMA CPM is an evolution of OMA SIMPLE IM, which offers many essential features unavailable with OMA SIMPLE IM. These include multi-device support, legacy messaging interworking and network-based message storage. European operators have agreed to migrate their SIMPLE IM deployment to CPM.

North America RCS (NARCS) focuses on how RCS could best be implemented to meet the region’s unique history, challenges and sought-after use cases. By comparison, the European market has a strong emphasis on RCS5.3 / 6.0 feature alignment, along with the Crane profile for devices. NARCS remains adhered to RCS 5.1 / 5.2 and defines its own regional profiles as guidance for OEMs and application developers. But it also looks at NNI implementation details as a means to ensure future interconnection among interested MNOs.

**VOLTE**

From both a standards and a network implementation point of view, VoLTE is being deployed throughout the world with more evolution to follow. With key Radio Access Network (RAN) features such as Transmission Time Interval (TTI) bundling and Robust Header Compression (RoHC), VoLTE is able to achieve the link budget necessary to match existing CS voice services.

According to SNS Research’s estimates, VoLTE service revenue will grow at a CAGR of 36 percent between 2015 and 2020. By the end of 2020, VoLTE subscribers will account for nearly $120 billion in revenue. Although traditional voice services will constitute a major proportion of this figure, over 12 percent of the revenue will be driven by video calling and supplementary services.

SRVCC will provide a seamless transition to VoLTE, and eSRVCC will support mid-call features during the seamless transition to other radio access technologies. QoS support will give VoLTE calls precedence over
non-Guaranteed Bit Rate (GBR) services. Also, features such as packet bundling provide added efficiency without significantly compromising latency.

Some chipset and OEM vendors have their own IMS stack implementations and their own network architectures. These differences mean portability and roaming between operators might prove to be challenging. Though most of the LTE handsets launched since 2014 have VoLTE capability, operators typically disable the feature until their networks are fully optimized and ready to support that service. The device ecosystem has grown substantially since the latter half of 2014. Apple’s release of iPhone 6 in 2015 was a big push to the whole ecosystem, especially in the North American region, where the U.S. alone has more than 100 million iPhone users. It leaves no option for operators that are not yet VoLTE ready to upgrade their networks for newer devices.

With basic VoLTE becoming widespread, operators have started to offer value-added features on top of IMS by integrating non-SIM devices with the user’s primary subscription. Multi-device calling capability is one such feature, where the same number can be used for calling on any of the user’s registered devices. Another flavor of the service allows calls over Wi-Fi through the same device by offloading from cellular and addressing indoor coverage issues, but still utilizing the same IP core network and calling features. Two Tier 1 U.S. operators intend to offer one-number usage with multiple devices. With iOS 9.0’s release in 2015, voice and messaging can be extended to multiple devices, not necessarily iPhones.

From a solution point of view, roaming architecture and interfaces have been standardized so customers experience the same service when they roam (i.e., Local Breakout for VoLTE). This means that QoS can be maintained and billing information can be communicated from visited to home networks as needed. The architecture also supports lawful interception and emergency calls while on the visited network.

**VOLTE CONSIDERATIONS**

**ENHANCED SINGLE RADIO VOICE CALL CONTINUITY (eSRVCC)**

When a customer’s phone has established a voice over IMS call and then moves to an area where LTE coverage is not provided, the eSRVCC procedure is invoked to continue the voice session. In preparation for this, the serving network has to upgrade several network components and implement a new network node called Access Transfer Control Function (ATCF) and the related gateway Access Transfer Gateway (ATGw).

When the network supports eSRVCC, the ATCF is inserted in the signaling path of the voice call. The ATCF is responsible for ensuring that the call control is transferred to the correct Mobile Switching Center (MSC) Server once the phone is in the GSM/EDGE Radio Access Network/Universal Terrestrial Radio Access Network (GERAN/UTRAN) access and interacts with the MSC Server enhanced to support SRVCC through a SIP-based interface. In case the MSC supports IMS Centralized Services (ICS), the interface is named as I2. The ATCF also controls the Access Transfer Media Gateway. Similar to the ATCF, which anchors the signaling, the ATGw anchors the media for the duration of the call. In other words, the voice media traverses the ATGw both while the phone is in LTE coverage and when it has moved to legacy CS networks. Figure 2 illustrates this network architecture.
Finally, the ATCF interfaces with the Serving Call Session Control Function (S-CSCF) so that the regular signaling path towards the destination can be established.

It is beyond the scope of this document to describe further the architecture details of how the continuity of a voice call is realized. However, it is important to note the following aspects:

- By anchoring the signaling for the call in the ATCF, the B party is not affected by the execution of an eSRVCC procedure, which is completely local.
- When the call is transferred to the CS domain, the P-CSCF is removed from the call path.
- In 3GPP Release 10 realizing eSRVCC, the ATCF does not possess the functionality to create call detail records (CDRs). That functionality is added in 3GPP R11.

SINGLE RADIO VIDEO CALL CONTINUITY (vSRVCC)

vSRVCC addresses video session continuity from E-UTRAN to UTRAN-CS for calls that are anchored in the IMS and when the UE is capable of transmitting/receiving on any one of those access networks at a time. The term vSRVCC is introduced for Single Radio Video Call Continuity to differentiate it from Single Radio Voice Call Continuity (SRVCC). vSRVCC procedures are defined in 3GPP Release 11. To enable video call transfer from E-UTRAN to UTRAN-circuit-switched network, IMS/EPC is evolved to pass relevant information to the CS domain, and all the EPC interfaces are enhanced for video bearer-related information transfer.

The vSRVCC architectures for 3GPP E-UTRAN to 3GPP UTRAN/GERAN reuse the session transfer function of SRVCC, where an MSC Server enhanced for SRVCC should be further enhanced for vSRVCC by providing following extra functions:
• Request radio resources for Bearer BS30 (64 kbps video) when it receives the Sv request with a vSRVCC indication from the MME.

• Negotiating with the SCC AS for the last active session to determine if it should perform SRVCC or vSRVCC procedure.

The high-level concepts for (v)SRVCC from E-UTRAN to UTRAN are:

• The UE uses one voice and one video media component over the associated QCI=1 and vSRVCC marked PS bearers for bearer identification reasons.

• The MME first receives the handover request from E-UTRAN. It then triggers the vSRVCC procedure with the MSC Server enhanced for vSRVCC via the Sv reference point with vSRVCC related information.

• The MSC Server enhanced for vSRVCC then interacts with IMS, initiates the session transfer procedure to IMS and coordinates it with the CS handover procedure to the target cell. If SRVCC with priority is supported, the IMS session transfer procedure and the CS handover procedure are performed with priority handling according to the priority indication received from MME.

• The MSC Server enhanced for vSRVCC then sends PS-CS Handover Response to MME, which includes the necessary CS HO command information for the UE to access the UTRAN.

• If the SCC AS indicates current active session is voice and video, the MSC Server requests UTRAN radio resources for BS30 bearer and continues with the vSRVCC procedure.

• When the UE receives the HO Command indicating a TS 11 or BS30 bearer, it knows whether it should start the CS 3G-324M video codec negotiation or SRVCC.

Figure 3 illustrates these processes.
REVERSE SRVCC FROM UTRAN/GERAN TO E-UTRAN (rSRVCC)

Reverse SRVCC allows a CS voice call to be handed over to LTE/HSPA as an IMS voice session from UTRAN/GERAN to E-UTRAN/HSPA (rSRVCC). The solution is aimed on enhancing user experience by providing better service, optimizing network usage and enhancing network coverage. It is expected that if coverage-triggered rSRVCC handovers are expected to occur, they should not be frequent.

Providing Users with Better Service

As packet services are better provided over E-UTRAN/HSPA, the operator deploys rSRVCC to make sure that users get service on E-UTRAN/HSPA as soon as E-UTRAN/HSPA becomes available. Typically, this is when the EUTRAN/HSPA cell quality is better than a given threshold.

Optimizing Network Usage

The operator may want to minimize CS core network usage and optimize the radio network usage, so it chooses to handover calls to PS as soon as E-UTRAN/HSPA becomes available. Typically this occurs when the EUTRAN/HSPA cell quality is better than a given threshold.

Enhancing Coverage

The operator wants to enhance its radio coverage by adding the possibility to hand over calls to E-UTRAN where GERAN/UTRAN coverage is getting weak. One possible reason is that the E-UTRAN cell quality is getting better than the GERAN/UTRAN cell quality. Another is that the GERAN/UTRAN cell quality gets worse than a given threshold, while the E-UTRAN cell quality is better than another one.

To support SRVCC from GERAN to E-UTRAN/HSPA, GERAN specifications are evolved to enable mobile stations and BSS to support seamless service continuity when a mobile station hands over from GERAN...
circuit-switched access to EUTRAN/ HSPA for a voice call. To support SRVCC from UTRAN to EUTRAN/ HSPA, UTRAN specifications are evolved to enable the RNC to perform rSRVCC handover and to provide relative UE capability information to the RNC.

At a high level, the procedures involved in rSRVCC are:

- The MSC Server first receives the handover request from UTRAN/GERAN with the indication that this is for CS to PS SRVCC handling. It then triggers the CS to PS SRVCC procedure, with the target MME/SGSN enhanced with CS to PS SRVCC via the enhanced Sv reference point.
- The MME enhanced with CS to PS SRVCC triggers PS HO in target E-UTRAN/UTRAN (HSPA).
- If the target MME/SGSN does not have the UE context, then the target MME/SGSN retrieves the UE context from the source SGSN/MME.
- The MSC Server then initiates the session transfer procedure to IMS and coordinates it with the PS handover procedure to the target cell.
- The MME then sends CS-PS handover Response to MSC Server, which includes the necessary CS to PS HO command information for the UE to access the E-UTRAN/UTRAN (HSPA)

**SRVCC FOR EMERGENCY SESSIONS**

With the introduction of emergency services in IMS, the need arises for emergency sessions to be transferred to CS similar to normal voice sessions in SRVCC. The UE initiates the IMS emergency sessions, which are anchored in serving PLMN. For facilitating session transfer (SRVCC) of the IMS emergency session to the CS domain, the IMS emergency session needs to be anchored in the serving IMS (i.e., in the visited PLMN when roaming).

The E-UTRAN initiates the SRVCC procedure as specified for regular voice over IMS session with few exceptions. The MME is aware that this is an emergency session and relays that status to the MSC Server enhanced for SRVCC. The MSC Server then initiates the IMS service continuity procedure with the locally configured E-STN-SR to the serving IMS as opposed to using the STN-SR provided by the MME for normal voice over IMS sessions. When handover of the emergency session has been completed, the MME/SGSN or the MSC Server may initiate location continuity procedures for the UE.

3GPP has not defined multiple E-STN-SRs configuration in the network so far. This implies that the serving PLMN is assumed to have a single logical EATF.

**E2E SERVICE PERFORMANCE (VOICE & VIDEO)**

The ultimate goal for operators that are launching VoLTE is to be able to phase out the legacy CS network such as HSPA/UMTS, GSM or CDMA2000. This means the baseline performance target for VoLTE has already been set: Its end-to-end performance shall be equal or better than that of the CS voice in all aspects, including UE battery drain, voice quality, mobility, call retainability and call features/functions. Early operator VoLTE deployments indicate that the following aspects are key for VoLTE/ViLTE to achieve this goal:

- Current drain during idle and active call.
- End-to-end path latency.
• Mobility management.
• QoS, radio resource allocation, transport, core.
• Voice and video codec.

Enhancements in these areas can significantly improve the end-to-end performance of VoLTE compared to VoIP alternatives such as OTT services. Due to the interrelation between VoLTE and ViLTE services, technology features implemented for VoLTE also apply to conversational ViLTE.

The main components affecting current consumption during voice calls are cellular radio and voice codec components:

• Current consumption during a call can be reduced with optimization in voice codec components such as the integration of the VoLTE audio codec in the device’s modem. Such enhancement lowers current consumption compared with OTT VoIP applications that perform frequent background ‘keep-alive’ activity in the application processor that can reduce battery life.

• Current consumption can be further optimized during a call and during user inactivity periods. The use of Discontinuous Transmission (DTX) and Discontinuous Reception (DRX) during a voice call can achieve considerable energy savings. DTX/DRX mechanisms allow a device to shut down cellular radio transmission and reception between voice packets to reduce energy consumption. In addition, packet aggregation between time intervals permits longer interruptions in transmission for further energy savings.

End-to-end path delay is critical for the voice quality. In a CS network, to ensure good voice quality, mouth-to-ear delay typically needs to be below 300ms. Because channel resources in the radio and core network are dedicated for CS voice and codec delays are fixed, the main variation of the path delay for CS domain may mainly come from the transport network. VoLTE, on the other hand, is by nature a LTE packet data service that competes with all other data services for radio resources. Although the radio network assigns higher QCI classes for VoLTE (QCI 1) and ViLTE (QCI 2), it still doesn’t eliminate the possibility of delay and collision in a loaded network. In addition, delay jitter caused by packet loss, retransmission or node processing delay requires the device and network to implement jitter buffer to overcome the path delay variations. This in turn leads to longer end-to-end path delay.

The VoLTE mobility scenario varies from operator to operator. For operators that are planning to deploy LTE in several phases, VoLTE to CS domain service continuity (SRVCC) will be critical. Early VoLTE developments have led to serval enhancement in the 3GPP standard to improve the performance of SRVCC (eSRVCC):

• Support of mid-call feature during SRVCC handover (eSRVCC, Rel10).

• Support of SRVCC PS-CS transfer of a call in alerting phase (aSRVCC, Rel10).

• Single Radio Video Call Continuity for 3G-CS (vSRVCC, Rel11)

Proper use of scheduling methods of packets over the radio interface can optimize the allocation of radio resources:

• Dynamic scheduling (DS) enables efficient use of radio resources associated with data applications. However, the dynamic allocation of radio resources has drawbacks because control channel information must be transmitted along with application data in both uplink and
downlink directions. Semi Persistent Scheduling (SPS) can be used to overcome this deficiency based on allocation of predefined radio resources for VoLTE. Due to the fixed allocation of radio resources, the SPS method is not well suited for mixed VoLTE and data traffic. Hence, in network deployments with mixed VoLTE and data usage, the packet aggregation described earlier can be used to compensate for the associated with the DS method.

- Use of RoHC can considerably compress the header size in voice packets and reduce the user plane traffic over the air interface. The high compression ratio obtained not only increases network capacity, but also provides coverage improvements for VoLTE compared to VoIP alternatives.

- TTI bundling can also be used to compensate for transmission errors in poor radio conditions. TTI bundling improves the reliability of VoLTE packet transmission in the uplink direction by sending a few redundant versions of the same set of bits in consecutive time intervals.

- Traffic between a device and the network is carried over bearers, which can have different QoS characteristics. When a VoLTE device attaches to the network for the first time, it will be assigned default non-GBR bearer for required IMS signaling. When a VoLTE call is set up, a dedicated GBR bearer is established for the voice connection. Such dedicated GBR bearers improve performance by ensuring a low latency and low jitter connection.

- With the selection of a dedicated bearer, VoLTE takes into account network design and dimensioning for QoS differentiation. By comparison, some VoIP alternatives run on the assigned bearer for Internet access, where differentiated QoS cannot be guaranteed. Although a VoLTE service requires additional signaling for setting up a dedicated GBR bearer, the effect on the associated network load is negligible.

Perceived voice quality depends on the audio codec used, as well as reduction of transmission impairments such as jitter, bit errors and packet loss:

- VoLTE originally was launched with AMR-WB support to improve voice quality needed to enable HD voice.

- The EV codec was launched in early 2016. It provides a super wideband experience and further improves HD call quality.

- The EVS codec’s new channel-aware mode makes the voice more resilient to the poor radio condition and thus effectively improves customer perceived VoLTE coverage.

**RAN PERFORMANCE INDICATORS**

Customer experiences with 2G and 3G voice services set QoS expectations that VoLTE must meet or exceed. By monitoring LTE RAN performance indicators, the EPC and IMS should provide operators with enough visibility into the VoLTE user experience. Operators also should monitor dedicated VoLTE indicators for accessibility, retainability, mobility integrity (speech quality) and LTE cell capacity.

KPIs are defined in 3GPP TS 32.450, including accessibility, retainability and mobility performance indicators distinguishing between the various quality-of-service class indicators (QCI). Thus operators should have visibility for each of these defined performance indicators for QCI1 bearer, which is the voice packets stream, and for QCI5 bearer, which is used for IMS control signaling. Monitoring an aggregate result for all various QoS bearers rather than monitoring the VoLTE bearers may mask VoLTE-related issues, especially when data and VoLTE usage are not homogenous within the network.
Note that with video streaming services that have a guaranteed bit rate in use, availability of the KPIs for QCI2, in addition to QCI1 and QCI5 bearers, is also recommended. The same recommendation goes for a use case where a non-GBR bearer is uniquely used for video streaming. In those cases, availability of the KPIs for the specific QCI is recommended.

It is also recommended to provide the operator with visibility of each of the raw counters used for calculating the performance indicators. That’s because this information will enable the operator to identify the root cause for network issues needed to be handled. Besides the relevant performance indicators defined in 3GPP TS 32.450, there are additional performance indicators to be measured and reported for operators. Those would enable operators to provide a suitable QoE for VoLTE users.

When it comes to mobility performance indicators, the success rate of the intra-LTE (inter-frequency and intra-frequency) handover (HO) procedure is defined in 3GPP TS 32.450. But the inter-RAT HO procedure, known as SRVCC, is not defined. With SRVCC being a new procedure, it is recommended that the operator have visibility to both of its sub-procedures, namely HO preparation (attempts and success) and HO execution (attempts and success). It is therefore recommended that SRVCC performance monitoring adopt the definition of mobility counters and mobility KPI counters as it is available in 3GPP TS 32.450, with the stipulation that the cell the UE is handed off to is a non LTE RAN cell.

Another aspect of mobility is that with voice being a delay-sensitive service, and with no voice packet retransmission at higher layers (GSMA IR. 92 sets voice as an unacknowledged mode DRB), tuning the mobility parameters to avoid early/late handovers is essential. By providing the operator with visibility into mobility counters (attempts and success of HO preparation and execution) and into mobility success rate KPIs uniquely for each of the LTE cell’s neighbors (both LTE cells and legacy RAN cells), the operator would be able to identify local mobility issues and fix them in order to optimize mobility performance.

Furthermore, visibility into the distribution of the different VoLTE HO failures types/causes is recommended. Operators also should have visibility into the distribution of interruption time, both for intra and inter-frequency HO, including SRVCC, for devices with an active VoLTE session (reflects user QoE during HO).

LTE offers the Radio Resource Control (RRC) re-establishment procedure to reduce the number of HO failures, avoid radio link failures and handling other issues. As a result, monitoring the number of RRC re-establishment attempts, success and failures per cause for active QCI1 is important. With this information, operators can be aware of potential degradation in VoLTE QoE that may not be fully visible otherwise. For example, a successful RRC re-establishment procedure doesn’t guarantee good QoE during the procedure itself.

Another important performance indicator is VoLTE integrity, which reflects voice quality during the call. While download/upload (DL/UL) throughputs are good metrics for data integrity, packet error loss rate, mouth-to-ear delay and jitter are suitable metrics for integrity of the voice service. These metrics, taking into account the voice codec rate in use, affect the mean opinion score (MOS), which is a standard method for evaluating voice quality.

The operator should have visibility of packet error loss rate, mouth-to-ear delay and jitter per VoLTE call and per direction (DL and UL). There are available definitions per QCI (and hence for VoLTE, which uses QCI1 and QCI5) for packet delay and packet drop rate in TS 32.425. There is also a definition for IP latency per QCI in TS 32.450. The operator should also have visibility to the used codec rate correlated with the different performance indicators. Some LTE eNodeB (eNB) vendors may choose to process this information per call and provide the operator a total estimation of the number of VoLTE calls meeting a certain QoE.
There is a list of other performance indicators that are recommended for operators to monitor in order to optimize VoLTE service performance and quality in their networks. One of them is VoLTE usage: the number of simultaneous VoLTE calls within a certain period of time. This parameter could also be measured in Erlang units or VoLTE usage during busy hour. The number of simultaneous VoLTE calls identifies both actual VoLTE usage and VoLTE load in the network. In the RAN, this can be measured, for example, by the number of simultaneous active ERABs of QCI1 and the duration (maximum and average) of a QCI1 session.

PDCCH usage is another recommended performance indicator to be monitored related to load and to VoLTE usage. With voice traffic patterns of small packets every 20 msec, and no SPS in use, the number of symbols for PDCCH is expected to increase. Sometimes it may even become the bottleneck of the number of supported users in the LTE sub-frame. It is therefore recommended to monitor and potentially have the ability to view the VoLTE downlink control information within a frame, in addition to the number of symbols within the PDCCH. Complimentary information would be the number of VoLTE calls, which are scheduled in an SPS manner.

The last recommended performance indicator is related to retainability: the number of E-RABs (active and non-active) that are normally released per QCI. This counter, together with number of active E-RABs with data in a buffer that were abnormally released per QCI (one of the counters in the retainability formula in 3GPP TS 32.450), provides the operator the VoLTE (QCI 1 bearer) call drop ratio, in addition to knowing the average duration of a call session. As E-RAB release can be triggered both by eNB and by MME, it is preferable to have dedicated counter for releases triggered by MME and for releases triggered by eNB for both normal and abnormal E-RAB release. It is also recommended to provide the operator with the distribution of the abnormal VoLTE E-RAB releases per cause.

The recommended VoLTE RAN performance indicators introduced above have been selected out of a wider range of performance indicators. These indicators benefit operators in adjusting their network to achieve a certain VoLTE QoE for their subscribers. Figures 4 and 5 provide a comprehensive list of customer perception KPIs and Network KPIs from ETSI.1

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As VoLTE deployments progress around the world, it is assumed that with the proper visibility to the operators, VoLTE-related network challenges can be optimized, and VoLTE service can be established as
the main voice solution on LTE networks. Until this happens, and as long as other voice alternatives such as CSFB are still in use, it is recommended to additionally monitor the number of CSFB attempts, as well as the CSFB success rate. This information enables estimates of the potential capacity for VoLTE and how many calls are deviated to CS. Examples include helping identify potential LTE coverage holes, helping improve VoLTE effective coverage and improving the user experience while being deviated to CS.

**LINK BUDGET AND VOICE QUALITY**

VoLTE deployments have grown significantly over the past few years, with approximately 40 VoLTE systems deployed worldwide. The GSA says 111 operators in 52 countries are investing in VoLTE and that 246 models of VoLTE-capable devices, including 224 smartphones, have been announced. All this activity has occurred despite the challenges VoLTE faced to provide the ubiquitous coverage that 2G/3G voice services have conditioned customers to expect. As detailed in a previous release of this white paper, VoLTE faced several challenges that were not present in 3G CS voice. These include:

1. VoLTE targeted the use of higher bit rate HD voice codecs such as AMR-WB. These reduce coverage because more voice information bits need to be sent. Thus fewer redundancy/channel coding bits can be sent, which degrades receiver sensitivity.

2. VoLTE has additional IP header overhead compared to CS 2G or 3G voice (7 bytes of IP overhead for VoLTE on the radio link. This extra IP overhead for VoLTE compared to 3G CS voice again means more bits to transmit, reducing receiver sensitivity.

3. VoLTE also has no Soft Handoff (SHO) like in 3G, putting it at a disadvantage in terms of link layer performance and fading margin requirements. SHO provides an additional diversity gain by transmitting/receiving signals simultaneously from multiple cells, which helps combat fast Rayleigh fading. It also reduces the required margin that must be included in the link budget due to longer term shadow fading.

4. LTE uses hybrid ARQ (HARQ) for voice packet transmission, which results in non-continuous transmission for a device. On the other hand, CS 3G allows cell-edge devices to transmit voice frames continuously at maximum power. The non-contiguous transmission of VoLTE devices introduces a penalty to its coverage, which is limited by the uplink.

Despite these challenges, VoLTE deployments are growing thanks to several aspects of LTE design that help make up for the shortcomings compared to 3G CS voice discussed above. These aspects include:

1. To mitigate the duty cycle limitation, TTI bundling was defined for VoLTE. This allows a bundle of four consecutive sub-frames to be transmitted on UL for a voice frame.

2. On the uplink, which generally is the limiting link for coverage, LTE provides good orthogonal between users. By comparison, 3G UMTS/WCDMA users are separated by not-perfectly-orthogonal scrambling codes.

3. LTE is an OFDM-based technology. This allows for Inter-Cell Interference Coordination (ICIC) in the frequency domain and dynamic Frequency Selective Scheduling (FSS) techniques not possible in 3G WCDMA-based systems.

4. The LTE HARQ operation enables better link efficiency and time-domain diversity in combating fading.
5. Many LTE networks have been deployed in lower radio frequency bands than 2G/3G voice (e.g., 700 MHz in the U.S. and 800 MHz in Europe/Middle-East/Asia). At lower frequencies, signals can propagate further, leading to improved cell coverage.

6. SRVCC has been used to drop down to 3G voice in areas where VoLTE coverage is challenged.

7. The Explicit Congestion Notification (ECN) enables codec mode control when radio conditions are under continually varying conditions or congestion limits cell resources. This gives the operator a flexible tool to trade off between voice quality and peak load.

VoLTE deployments have taken advantage of the above techniques in order to overcome the challenges of VoLTE as compared to 3G CS voice.

To give VoLTE a clear quality benefit compared to 2G/3G networks, most VoLTE deployments use the highest codec mode of WB AMR (23.85 kbps). In most regions, VoLTE will come as an evolution of 2G/3G networks. Thus the seamless interworking between VoLTE and CS voice is key for many operators. Therefore, operators are also introducing WB AMR in 2G and 3G networks, although only with the three lower modes of WB AMR (6.60 + 8.85 + 12.65 kbps). Transcoding Free Operation (TrFO) between WB AMR in CS and in VoLTE is of utmost importance for a consistent HD voice user experience. As a result, TrFO-compatible Multi-Mode-sets have been defined, like mode-set=0,1,2 for CS voice and mode-set=0,1,2,8 for VoLTE, adding the highest codec mode only for VoLTE → VoLTE calls. Meanwhile, VoLTE → CS voice calls will be restricted to the lower modes, but they will still be substantially better than NB AMR calls.

To further improve voice quality, coverage and battery life, Release 12 introduced the EVS codec, which offers new features and improvements for VoLTE and 3G UMTS CS systems. The EVS codec provides better compression efficiency and significantly improved quality (beyond earlier 3GPP AMR-WB codec's HD Voice) over clean and noisy speech and music, including support for WB, super-wideband (SWB) and full-band (FB) content. The EVS codec operates at a broad range of bitrates, as shown in Table 1. It can efficiently code content at various audio bandwidths (where NB is up to 4 kHz, WB is up to 8 kHz, SWB is up to 16 kHz, and FB is up to 20 kHz), at an algorithmic delay of 32 ms. EVS also supports the source bit rates in Table 2.

| Source codec bit-rate  
<table>
<thead>
<tr>
<th>(kbit/s)</th>
<th>Audio bandwidths supported</th>
<th>Source Controlled Operation Available</th>
</tr>
</thead>
<tbody>
<tr>
<td>5,9 (SC-VBR)</td>
<td>NB, WB</td>
<td>Yes (Always On)</td>
</tr>
<tr>
<td>7,2</td>
<td>NB, WB</td>
<td>Yes</td>
</tr>
<tr>
<td>8,0</td>
<td>NB, WB</td>
<td>Yes</td>
</tr>
<tr>
<td>9,6</td>
<td>NB, WB, SWB</td>
<td>Yes</td>
</tr>
<tr>
<td>13,2</td>
<td>NB, WB, SWB</td>
<td>Yes</td>
</tr>
<tr>
<td>13,2 (channel aware)</td>
<td>WB, SWB</td>
<td>Yes</td>
</tr>
<tr>
<td>16,4</td>
<td>NB, WB, SWB, FB</td>
<td>Yes</td>
</tr>
<tr>
<td>24,4</td>
<td>NB, WB, SWB, FB</td>
<td>Yes</td>
</tr>
<tr>
<td>32</td>
<td>WB, SWB, FB</td>
<td>Yes</td>
</tr>
<tr>
<td>48</td>
<td>WB, SWB, FB</td>
<td>Yes</td>
</tr>
<tr>
<td>64</td>
<td>WB, SWB, FB</td>
<td>Yes</td>
</tr>
<tr>
<td>96</td>
<td>WB, SWB, FB</td>
<td>Yes</td>
</tr>
<tr>
<td>128</td>
<td>WB, SWB, FB</td>
<td>Yes</td>
</tr>
</tbody>
</table>

Table 1. EVS Codec Source Bit rates from 3GPP TS 26.441
The EVS codec introduced three key features for VoLTE and 3G UMTS CS: super-wideband coding², music coding³ and channel-aware advanced error resiliency coding⁴. These features are described below. A detailed 3GPP Technical Report (TR 26.952⁵) on the subjective and objective evaluation of the EVS codec has been published.

**Super-Wideband Coding**

Earlier 3GPP conversational codecs are limited to compression of narrowband or wideband signals. EVS is the first 3GPP conversational codec to offer super-wideband coding up to 16 kHz from bitrates starting at 9.6 kbps in combination with features such as discontinuous transmission (DTX). For the ACELP-based speech coding, the larger audio bandwidth is achieved by bandwidth extension technology, namely a time-domain bandwidth extension (TBE) technology is used during speech. The TBE algorithm uses a nonlinear harmonic modeling technique that incorporates principles of time-domain envelope-modulated noise mixing. At 13.2 kbps, the super-wideband coding of speech uses as low as 1.55 kbps for encoding high band content. For music, the coding of higher bandwidth is integrated within the respective MDCT-based framework. The result is higher efficiency across all types of content, in particular for speech.

**Music Coding**

Earlier generations of 3GPP codecs for voice services, such as AMR and AMR-WB, are based on the principles of speech coding. EVS is the first 3GPP codec to deploy content-driven switching between speech and audio compression leading to significantly improved coding of generic content (e.g., speech/music and mixed content). Two improved variants of MDCT coding are implemented in EVS: the low/high rate high quality MDCT coding and TCX coding. Depending on the characteristics of the input signal, the EVS encoder selects one of the MDCT variants for coding music content.

**Channel Aware – Advanced Error Resiliency Coding**

Multiple measures have been taken to provide a built-in, highly robust frame loss concealment to mitigate the impact of packet loss in mobile systems. Inter-frame dependencies in the core coding have been minimized to arrest error propagation and thereby ensure fast recovery after lost packets. The channel-aware coding at 13.2 kbps extends even further the error robustness performance through source/channel-controlled transmission of partial redundant information of previous frames. The EVS Channel Aware mode

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⁵ 3GPP TR 26.952, “Universal Mobile Telecommunications System (UMTS); LTE; Codec for Enhanced Voice Services (EVS); Performance characterization,” http://www.3gpp.org/DynaReport/26952.htm
introduces a novel technique for transmitting redundancy in-band as part of the codec payload in a constant bitrate stream, and is implemented for WB and SWB at 13.2 kbps.

EVS is capable of switching between:

- Active Speech – using linear predictive coding techniques based on ACELP.
- Generic and inactive content – supporting Voice Activity Detection (VAD), Discontinuous Transmission (DTX) and Comfort Noise Generation (CNG).

Through the features discussed above, EVS is expected to provide noticeable improvements in the audio experience, improved spectral efficiency and enable new services and applications compared to the current AMR-WB and AMR-NB codecs.

**VOLTE ROAMING – TECHNICAL AND COMMERCIAL CONSIDERATIONS**

**OVERVIEW**

For inter-operator EPC roaming between PLMNs, 3GPP TS 23.401 defines the roaming architecture for 3GPP access. For LTE data roaming, the traffic is home routed: The application at the home network controls the user session, both the user’s signalling and media traffic are routed to home network and the user’s traffic is billed by data usage.

As discussed above, there are two VoLTE roaming models that operators can implement: Local Breakout (LBO) and S8-Home Routed (S8HR). In the LBO model, the entire enhanced packet core network (EPC) and P-CSCF are located in the visited network for a VoLTE roaming user. The roaming user attaches in a visited network, with the IMS PDN anchored on a PGW in the visited network, and registers for IMS services in its home IMS network. The P-CSCF then interacts with the S-CSCF and TAS in the home network for SIP signalling and setting up a VoLTE session. Diameter signalling is exchanged between the visited network’s MME and the home network’s HSS via the S6a interface. The interface between the V-PCRF and the H-PCRF (S9 interface) is optional. The LBO model allows the MNO to keep the same billing model used in current CS networks both for VoLTE and SMS (minute of use for voice and event based for SMS) as recommended in GSMA PRD IR.88 and IR.65.

The VoLTE basic call flows are in accordance with 3GPP specifications for E-UTRAN/EPC, IMS and PCC. (Please refer to 3GPP TS 23.401, TS 23.228, and TS 23.203 respectively, and relevant stage 3 specifications for further detailed information.) The general VoLTE roaming architecture with LBO is shown in Figure 6 (from 3GPP TS. 23.228).
For VoLTE roaming with LBO, the control signalling is always routed to the home network Application Server (AS). For media traffic, there are options with home routed and visiting network routed. Figures 7 and 8 show the high-level logical diagrams for both cases.

Figure 7. IMS Roaming Architecture with LBO and VoLTE MO Call Media Home Routed.
With P-CSCF located in the visited network, the call/session control is by default in the HPLMN (S-CSCF and TAS Telephony Application Server). After the VPLMN route, the user initial SIP INVITE message to the HPLMN, 3GPP TS 23.228 Annex M describes two signaling routing options: HPLMN routes the signaling to the far end and loopback to VPLMN.

Interworking between H-PCRF and V-PCRF can be used to allow H-PCRF in the home network, where it provides PCC rules to the V-PCRF in the visited network. The deployment of the S9 interface is determined on a bi-lateral operator agreement. It is also possible to use static policy control where the local configuration data is stored in the V-PCRF, in which case the S9 interaction does not occur. The V-PCRF in the visited network provisions PCC rules in the PDN GW in the visited network for bearer handling. Media exchanged between the UE and the far end is routed either between the PDN GW in the visited network and the far end, or between the PDN GW in the visited network via the TrGW in the home network, then routed to far end. For both routing cases, local breakout model is used, whereby the full EPC and P-CSCF are located in the visited network for the roaming user.

In loopback case, the SIP session signaling is looped back from the home network to the visited network, while the media is anchored in the visited network. From that point, both the signaling and media are routed to the terminating party’s home network as per current roaming CS routing principles.

In addition to the architecture in Figure 8, the Transit & Roaming Function (TRF) is required within the visited network to handle the looped back routing between the visited and home networks of the originating party prior to forwarding the session request to the home network of the terminating party. TRF is also one of the nodes in VPLMN generating CDRs for VoLTE roaming traffic.

In LBO’s case, Optimal Media Routing (OMR) is used at the originating VPMN side to route the media to the terminating home network. That way, the originator’s home network does not need to maintain the media path for the call. Note that the signalling shall always be conveyed to the home network.

For a mobile terminating VoLTE call roaming case (i.e., the destination user is roaming outside of home network), the signaling procedure is generally reciprocal with the MO roaming case in Figure 8. This maintains the same business model as CS voice.
In the S8HR VoLTE roaming architecture, the PGW, PCRF and the IMS core network are all located in the roaming user’s home network. IMS services are home routed using the well-known IMS APN via the S8 interface, similar to LTE data roaming traffic. SIP signaling and RTP media use the same IMS APN established with the HPLMN, each with specific QCI. The HPLMN has full control of call routing and can execute VoLTE service logic for its users in the VPLMN. The VPLMN supports the required capabilities to support VoLTE for inbound roamers (i.e., it supports the IMS Voice over PS (IMSVoPS) supported indication and the specific QCIs required for SIP signaling (QCI=5) and voice media (QCI=1). The IMS APN configuration in the user’s profile in the HSS must be set to “VPLMN address not allowed.”

Figure 9 illustrates the S8HR roaming architecture.

**IMPACT ON ENHANCED SRVCC**

In the enhanced SRVCC architecture, the ATCF/ATGW anchors the call in the VPLMN. This makes the access change transparent to the HPLMN.

In VoLTE roaming with LBO routing, the eSRVCC has no impact because TRF anchors the call for its entire duration, including after an SRVCC event. The VPLMN also may correlate the ATCF CDR with the TRF CDR.

In VoLTE roaming with S8, the MSC in the VPLMN interfaces with the home IMS network using ISUP/BICC. As a result, eSRVCC, SRVCC with MSC-server assisted mid-call feature, SRVCC in alerting phase and SRVCC in pre-alerting phase are not supported.

Pre-Release 10 SRVCC without the use of an ATCF/ATGW is supported for S8HR architecture. However, the user experience may be affected due to the latency of the handover. It should also be noted that reverse SRVCC (rSRVCC), which is the handover from the CS to PS domain, is also not supported in an S8HR roaming architecture.
DIAMETER SIGNALLING

Roaming introduces the added complexity of an additional capability required to provide a PMN-edge Diameter function to reduce the export of network topologies and support scalability and resilience. This Diameter Edge Agent is described in GSMA PRD IR.88, section 3.1.3.

The Diameter Edge Agent needs to discover which peer to route messages to, for a specific application to enable Diameter routing between networks. This may be done by manual configuration of Diameter nodes within each node. However, a dynamic discovery mechanism (NAPTR query) would allow for a simpler, scalable and robust deployment. GSMA PRD IR.88, section 3.1.3.4 describes the peer discovery mechanism in further detail.

The roaming scenario requires Diameter messages to be routed between the respective Diameter Edge Agents. As in the single network scenario, all Diameter-enabled elements in a network route their requests and responses via the local Diameter Agent. For the roaming scenario, the Diameter Agent routes the requests/responses via the local Diameter Edge Agent based on the realm identity in the message. A Diameter Edge Agent may apply topology hiding to reduce the export of topology information across a network boundary.

For additional security between the Diameter Edge Agents, see section 6.5.1 of GSMA PRD IR.88.

OPTIMAL MEDIA ROUTING

As a network option, it is also possible to utilize Optimal Media Routing (OMR), as recommended in GSMA PRD IR.65 and described in 3GPP TS 29.079 in the roaming scenario. The purpose of OMR is, when possible, to maintain the media path in the visited network and avoid needlessly tromboning the media path via the home network. Note that the signaling should always be conveyed to the home network.

In the VoLTE roaming scenario with LBO, OMR occurs for IMS sessions, including call cases where the roaming user is roaming in VPLMN, the media is routed by the VPLMN to terminating HPLMN directly. If OMR is applicable, the IBCF frees off the media resources in the TrGW during session establishment. This means there is subsequently no need for the IBCF to communicate with the TrGW during session teardown. See annex C of 3GPP TS 29.079 for message example flows illustrating OMR.

Optimal Media Routing is not applicable when S8HR roaming is used because the voice call is routed and handled by the home network.

TRAFFIC MANAGEMENT AND POLICY

Dynamic or static policy control may be applied between the home and visited networks. For dynamic policy control when LBO is used, Rx messages are sent between the visited network PCRF and home network PCRF via the optional S9 interface.

In terms of DiffServ marking, TrGWs are present in the end-end media path and are thus able to modify the DSCPs under the control of the IBCF. See GSMA PRD IR.34 for traffic class mappings.

SESSION BORDER CONTROL

For the roaming scenario, both signaling and media traverse the network boundary between the visited and home networks via network-side session border controllers (SBCs). These nodes perform the role of an IMS IBCF/TrGW and provide capabilities such as topology hiding, firewall and NAT traversal.
It is also possible for an access and network-side SBC to be co-located in a single physical element. In this case, the message flows documented in section 5 may be simplified as a single element that would provide the IMS P-CSCF/IMS-ALG/IMS-AGW and IBCF/TrGW functions. This enables messages between the P-CSCF and IBCF (in the control plane) and IMS-AGW and TrGW (in the media plane) to be internal to that element.

**EMERGENCY CALL AND LAWFUL INTERCEPT**

A UE-detected emergency call is always subject to local breakout, regardless of the VoLTE roaming model used between MNOs. Such calls are routed to the local PSAP using procedures standardized in 3GPP TS 23.167 and 3GPP TS 24.229. The UE uses the emergency service URN (See RFC 5031), and the E-CSCF routes the call to the nearest PSAP. However, as discussed above when the S8HR roaming model is used, there may not be an IMS NNI implemented between the VPLMN and the HPLMN. In this case, the IMS emergency session cannot be authenticated using standard authentication procedures outlined in 3GPP TS 23,167. 3GPP is currently in the process of standardizing a solution for this scenario, and a proposal to use GIBA-like procedures is captured in 3GPP TR 23.749 v1.2.

Non-UE-detected emergency sessions pose certain challenges when the S8HR roaming model is used. Because the UE is unable to detect the dialed digits as emergency numbers, procedures such as Emergency Registration or Emergency PDN connectivity request are not initiated. This results in the call being routed to the HPLMN just like any other non-emergency call. Then the IMS entities in the HPLMN have to detect and handle such calls. Proposed solutions are captured in 3GPP TR 23.749 v1.2. Once a solution is finalized and approved by 3GPP working groups, the respective normative specifications will be updated.

For legal interception in the local breakout roaming scenario, the Access SBC (P-CSCF, IMS – ALG, IMS – AGW) in the visited network is a mandatory point of intercept (see 3GPP TS 33.107 section 7A). However, legal interception requirements when S8HR roaming model is used are still being investigated by 3GPP SA3-LI working group at the time of writing this document.

**CHARGING**

For VoLTE calls in the roaming scenario with local break-out with the P-CSCF in the visited network, the VPMN is service aware. There is scope for service-based charging to be deployed between the visited and home networks and to apply flow-based charging mechanisms. The VPLMN can apply differentiated charging rules based on the QCI of the bearers used for VoLTE, ViLTE or RCS messaging. It also can use the P-CSCF CDRs for service-based usage and exchange such records with the VPLMN using Transferred Account Procedure (TAP) records for wholesale charging for inbound roamers.

The HPLMN has visibility into the SIP signaling used for different IMS sessions and can use the CDRs generated by the TAS or the RCS Messaging server for retail charging. When the S8HR roaming model is used, the VPLMN is not service aware, and differentiated charging is not possible at the wholesale level. Retail charging by the HPLMN can follow the same principles as used for the LBO scenario. That’s possible because the HPLMN is completely service aware and can charge the user for IMS services as the business case warrants. For example, volume-based charging can be accomplished for ViLTE, duration-based charging can be accomplished for VoLTE and event-based charging can be accomplished for RCS messaging.
EMERGENCY SERVICES

Emergency services are citizen-to-authority communications that are typically associated with emergency or life-threatening situations. These emergency calls are typically routed to the Public Safety Answering Point (PSAP) serving the current location of the mobile device that initiated the emergency call.

While wireless networks have evolved to IP-based communications such as VoLTE and RCS, the emergency services community has also been working on an evolution to an IP-based network. The Emergency Services Network (ESInet) is an all-IP, SIP-based network. Similarly, the PSAPs are also migrating from legacy circuit switched to all-IP (i.e., NENA i3 PSAP).

“ATIS Standard for Implementation of 3GPP Common IMS Emergency Procedures for IMS Origination and ESInet/Legacy Selective Router Termination” was created to handle emergency calls. Figure 10 illustrates the migration of emergency calls from a legacy CS architecture to an all-IP Emergency Services Network (ESInet). The top portion of the figure shows the traditional circuit switched based emergency call, originating in the legacy TDM network and routed to a legacy PSAP via Selective Router (SR).

Another configuration supported is when calls originate from legacy TDM networks and are routed to a NENA (i3) PSAP via SIP/IP Based Emergency Services Network (ESInet). Yet another phase toward IP migration is when calls originate in the IMS Originating network and can be routed to either a legacy PSAP or to a NENA i3 PSAP via ESInet. ATIS is developing a standard to migrate from SIP-based ESInet to a 3GPP common IMS-based ESInet, as depicted in Figure 10. The goals are to take advantage and share common nodes (such as LRF) in the IMS originating network and IMS-based ESInet.

ATIS and TIA recently published “Joint ATIS/TIA Native SMS to 9-1-1 Requirements and Architecture Specification,” which provides support for SMS text to 911 over mobile networks. This feature lets users communicate with a PSAP operator using SMS.
The text message contains “911” as the destination address. The SMS text message from the caller is sent to the SMSC using the existing SMS transport mechanism. The SMSC forwards the text message to a new platform specifically developed to support SMS to 911: The Text Control Center (TCC). The TCC establishes a dialogue with the PSAP (it identifies the appropriate PSAP based on caller location) and sends the SMS text message in the form of TTY Baudot tones (legacy PSAP) or MSRP (NENA i3 PSAP).

As a transition from legacy PSAP to NENA i3 PSAP, the SMS text messages can also be delivered to IP-based PSAPs (referred to as Transitional PSAP) via Web interface. In the reverse direction, the TCC receives the text messages from the PSAP (TTY/MSRP) and converts the text to SMS message, which it then forwards to the mobile device through the SMSC. Figure 11 illustrates this process.

![Figure 11. Text to 911 Involving Legacy and NENA i3 PSAPs](image)

Note that if the emergency text message contains multimedia, only text is delivered to PSAP. If there are multiple destinations, then other destination information is removed before forwarding the text to PSAP.

**E9-1-1 LOCATION REQUIREMENTS**

In traditional wireline architecture, the emergency caller had a fixed and thus dispatchable address associated with his or her location. The nomadic nature of mobiles created a need to be able to locate emergency callers using geolocation: latitude and longitude. Finding a mobile emergency caller inside a building where GPS signal is weak or not reachable has been a challenge. An active ATIS project aims to deliver a dispatchable location for emergency calls originated by mobile devices inside the building using other means beside GPS.

Figure 12 is a high-level architecture diagram that shows the use of NEAD/NEAM/External data sources to determine the indoor location of a caller during emergency.
The National Emergency Address Database (NEAD) is a countrywide database that utilizes MAC address of Wi-Fi access points or public device address of Bluetooth beacons and maps them to a dispatchable location. The NEAD stores geo-coded location information associated with the dispatchable location (i.e., geo-coded location of the Wi-Fi access points or the Bluetooth beacons). The NEAD responds to queries for dispatchable location information from a serving core network in support of individual emergency calls.

The National Emergency Address Manager (NEAM) supports OAM&P aspects for the NEAD. The NEAM provisions the NEAD with a dispatchable location against the Wi-Fi MAC address or the Bluetooth beacon public device addresses. The NEAM supports identification and authentication of external data sources and validation of received civic location information before provisioning it in the NEAD.

Additional aspects of this project are still ongoing in ATIS and various CTIA groups. For example, one of the CTIA sub-working groups is developing a test plan. It also is identifying cities that provide a good representation of the entire country so those can serve as test sites.

**ALERTING SERVICES**

Alerting services are authority-to-citizen broadcast of alert messages to inform the population, via their mobile devices, of imminent threats to life or property. In the U.S., the broadcast of alerting messages is called Wireless Emergency Alerts (WEA). WEA alerts carry text messages with up to 90 displayable characters. WEA alerts are broadcast over GSM, cdma2000, UMTS and LTE networks.

Wireless operators elect whether or not to participate in WEA. All major U.S. wireless operators support WEA.

The ability of the user to receive the broadcast WEA alert is a function of the mobile device capabilities and the configuration options selected by the user. There are three types of alerts: Presidential, Imminent Threat and AMBER. Users can opt out of Imminent Threat Alerts and AMBER Alerts but cannot opt out of Presidential Alerts.

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6 WEA is the name of the alerting service that was adopted by the FCC and wireless industry shortly after it became deployed and available to the public. In the FCC Notice of Proposed Rulemaking (NPRM) and the three associated FCC Reports and Orders (R&O), this alerting service is called Commercial Mobile Alert Service (CMAS). CMAS is also the term that is used in the ATIS standards, the Joint ATIS/TIA standards, and the 3GPP standards.
ATIS recently completed a standard on “CMAS International Roaming (WTSC-LB-096)” to allow users to receive wireless emergency alerting while roaming. One of the challenges of WEA international roaming is the possibility of WEA broadcasts in different languages. The devices should be able to receive the alerts broadcast in the local languages.

**LAWFUL SURVEILLANCE**

Standards for the support of lawful surveillance are being developed in both 3GPP and ATIS working groups. ATIS standards are used for the networks deployed in North America.

In 3GPP, there is currently a study evaluating the feasibility of having an LI solution when “S8 Home Routed (S8HR)” based approach is used as the roaming architecture for VoLTE roamers. With S8HR, since there is no IMS node present in the visited network for IMS message interception, the current LI capabilities will not be able to support the LI of voice calls originated or received by the inbound roamers, thus requiring a new solution. Different approaches are under discussion as part of the study.

**ACCESSIBILITY FOR INDIVIDUALS WITH DISABILITIES**

Accessibility for individuals with disabilities must be considered for 4G services such as VoLTE and RCS Messaging.

In a CS environment, accessibility is provided through the use of TTY (known as teletypewriter or text telephony) devices. Individuals with disabilities use TTY devices to send and receive text messages that are transmitted over the CS network in the form of Baudot tones. As operators transition to an all-IP environment, there is an opportunity to make use of an internal keyboard on the screen of a wireless device rather than having the user carry a bulky standalone TTY device.

In 2013, the FCC Emergency Access Advisory Committee (EAAC) recommended the use of a standardized Real Time Text (RTT) service as networks transition toward all-IP. The disability community (including deaf and hard of hearing) was overwhelmingly in favor of RTT as a service versus continuing to perpetuate TTY. However, they stressed that TTY must continue to be supported as long as CS networks exist because legacy TTY service may be the only option for certain users. ATIS standards work is ongoing to support RTT as a replacement of the traditional TTY devices.

**GOVERNMENT PRIORITY SERVICES**

Government priority services are authority-to-authority communications. In the U.S., Wireless Priority Service (WPS) and Government Emergency Telecommunication Service (GETS) are two such services administered by the Office of Emergency Communications (OEC) of the Department of Homeland Security (DHS).

During emergencies, wireless networks can experience congestion due to increased call volumes and/or damage to wireless network facilities. As a result, the ability of national security and emergency preparedness (NS/EP) personnel to communicate can be hindered. WPS improves the communications capability of the NS/EP personnel by allowing communication requests from WPS-enabled NS/EP personnel to be placed at the top of the queue for the next available radio resource.

During emergencies, the public telephone network also can experience congestion due to increased call volumes and/or damage to network facilities. GETS provides NS/EP personnel priority access and

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7 https://www.dhs.gov/wireless-priority-service-wps
8 http://www.dhs.gov/government-emergency-telecommunications-service-gets
prioritized processing in the local and long distance segments of the Public Switched Telephone Network (PSTN), greatly increasing the probability of call completion. GETS is intended to be used in an emergency or crisis situation when the PSTN is congested and the probability of completing a normal call is reduced.

To obtain end-to-end priority services, WPS and GETS are used in conjunction. Specifically, the NS/EP personnel would use WPS to obtain access to GETS.

In 3GPP, the priority services related functions are defined under the name Multimedia Priority Services. 3GPP Release 10 enhanced the MPS to support priority handling in EPS and IMS. These enhancements enable the network to support end-to-end priority treatment for MPS call/session origination/termination, including the Non Access Stratum (NAS) and Access Stratum (AS) signaling establishment procedures at originating/terminating network side, as well as resource allocation in the core and radio networks for bearers. Priority treatment is applicable to IMS-based multimedia services, priority EPS bearer services and CS fallback.

The MPS enhancements captured in 3GPP TR 23.854 have triggered updates to the relevant 3GPP specifications for E-UTRAN, PCC, IMS, CSFB, etc. to improve support for E-UTRAN MPS.

REGULATORY - CANADA

Many Canadian regulatory features are similar to U.S. ones. For example, Canada supports E9-1-1, lawful intercept and is currently developing a mobile alerting feature. The following sections highlight some of the differences and provide links to additional information. All these regulatory voice features must be supported whether using CS voice or VoLTE/RCS. Unless otherwise noted, it is fair to assume that VoLTE/RCS support for these Canadian regulatory features is similar to what is done for the respective U.S. regulatory feature.

The Canadian Radio-television and Telecommunications Commission (CRTC) is an administrative tribunal that regulates and supervises broadcasting and telecommunications in the public interest.9

EMERGENCY SERVICES

In Wireless E9-1-1 Phase 1 (August 2003), the CRTC mandated that all Canadian mobile operators implement a form of E9-1-1 service whereby the telephone number and cell site/sector information of wireless E9-1-1 callers would be automatically conveyed to the E9-1-1 call center or PSAP. The CRTC required operators to provide this service in all areas where wireline E9-1-1 service is available and to notify their customers regarding the availability and limitations of their particular wireless E9-1-1 service.

Wireless E9-1-1 Phase II10 (February 2009) includes a number of additional enhancements. Phase II provides PSAPs with Phase I information, plus more accurate longitudinal and latitudinal (X, Y) information regarding the location of each wireless E9-1-1 caller. Some of the new Phase II requirements are:

- Preservation and use of wireless Phase I E9-1-1 voice routing as the basis for wireless Phase II E9-1-1 call handling capabilities.
- Assignment of a unique class of service identifier for wireless Phase II E9-1-1 calls.

9 [http://www.crtc.gc.ca/eng/acrtc/acrtc.htm](http://www.crtc.gc.ca/eng/acrtc/acrtc.htm)
• Absence of filtering of wireless E9-1-1 data provided to PSAPs by either Canadian mobile operators or incumbent local exchange carriers (ILECs) with which they are interconnected for the provision of wireless E9-1-1 caller information to the PSAPs.

• Automatic enabling of cell phone GPS functionality where GPS technology is being used.

• Use of the mobile location protocol (MLP) interface standard for the transfer of data between mobile operators and ILECs.

• Transmission of X, Y coordinates by mobile operators and ILECs in the MLP defined format, using World Geodetic System 1984 map projection datum.

• Provision of continuous mobile operator call center support to PSAPs wherever wireless Phase II E9-1-1 call location information is available,

• Setting threshold values for specific location system parameters that provide PSAPs with a caller's wireless Phase II E9-1-1 location information.

• Deployment of wireless Phase II E9-1-1 features in two stages:
  
  o Provision of X, Y coordinate information and location system parameters that PSAPs use to determine a caller's location (stage 1).
  
  o Provision of mid-call location updates, plus provision of wireless Phase II E9-1-1 service for roamers and unsubscribed handsets, to be deployed as technology solutions become available (stage 2).

Canada also supports text messaging to 9-1-1 in ways that are quite different from U.S. text messaging to 9-1-1. In Canada, the service is designed for hearing- or speech-impaired individuals and requires the individual to pre-register for this messaging to 9-1-1 service. The registered Canadian user would then dial 911, which would then lead to an SMS session with a 9-1-1 operator at the PSAP.

Canada is currently still evaluating the needs and timing of deployments for NG 9-1-1.

**CANADIAN LAWFUL INTERCEPT**

Canada’s criminal code sets out the provisions for the law enforcement community to obtain judicial authorization to conduct electronic surveillance of private communications for criminal investigations. Public Safety Canada releases an annual report on the Use of Electronic Surveillance.11 Canada tends to leverage lawful intercept from ATIS and 3GPP SA3 LI (similar to the U.S.). One unique feature for Canadian lawful intercept relates to intercepting and reporting only the location for the person under surveillance. This feature is standardized in Canadian LAES Location Reporting (ATIS-0700009).

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CANADIAN WIRELESS PUBLIC ALERTING SERVICE (WPAS)

Canada is currently developing a nationwide public alerting feature for LTE. The work is being done across many groups, including CRTC, ATIS and 3GPP.

The Network Working Group (NTWG) of the CTRC Interconnection Steering Committee (CISC) took the technical lead on Canada’s WPAS work. (See http://www.crtc.gc.ca/cisc/eng/cisf3d0.htm). It defined the requirements for WPAS and has been corresponding on the standards work with ATIS. It is also developing the C-Interface specification between the National Alert Aggregation & Dissemination (NAAD) system and mobile operator networks.

A few of the Canada’s WPAS requirements include:

- Support only for LTE (no required support for GSM or CDMA).
- All messages will be mandatory (the user cannot opt out).
- Adherence to Canada’s official bilingual laws, where messages will be in both French and English.
- A special WPAS alerting tone will be used along with a special WPAS Alert banner.

More information is available at http://www.crtc.gc.ca/public/cisc/nt/NTRE055.docx

ATIS standardized the device behavior in ATIS-0700021 (Canadian Wireless Public Alerting Service (WPAS) LTE Mobile Device Behavior). The Canadian WPAS service will use CBS (Cell Broadcast short message service) message identifier ‘4370’ from 3GPP TS 23.041.

LOCAL NUMBER PORTABILITY

Canada also supports local number portability as a regulatory feature. Visit the CRTC website for information regarding local number portability.

VOLTE AND RCS COEXISTENCE

IMS - FUTURE PROOF NETWORK EVOLUTION PATH

OVERVIEW

When introducing VoLTE and RCS Messaging into networks and devices, it is important to understand the positioning of and interaction between the two. This section explains how RCS Messaging and VoLTE can coexist in the same device, served by the same IMS network and be used by the same subscription using the same phone number. It also outlines the evolution toward rich communication of the future. This includes providing support for decisions that have to be made by operators regarding when to introduce particular capabilities and functionalities, and how to evolve the network, taking the device evolution into account. The network needs to be ready for the devices of the future in order to take advantage of new device capabilities as soon as they debut.

VoLTE is viewed as voice and video service over LTE, without necessarily confining it to the minimum profile as defined in IR.92/IR.94. VoLTE promises HD voice and video and new services for communication handling without compromising the service performance expected from of a first-line telephony service.

RCS Messaging services are complement to the telephony service, where the telephony service may be classic CS or IP based (such as VoLTE or VoWiFi). RCS services are access technology independent and are available today with cellular (2G, 3G and 4G) and Wi-Fi.
The network and device architectures need to support the combined VoLTE and RCS Messaging service experience. Subscribers expect a predictable, high grade of service regardless of whether they are at home, roaming or in different environments (e.g., indoor, outdoor, office, hotel). User expectations for quality and cost also need to be considered when describing the use cases and experience. Those needs may even depend on the current environment and differ when roaming and when at home. For example, a video call over Wi-Fi might be preferred when in a hotel and roaming, while using LTE might be the preference when in the home operator network.

The network needs to support different implementations of RCS Messaging and VoLTE in the same device and the use of the device on a range of access technologies, possibly determined by user choice. Access technologies can be LTE, WCDMA, GSM, CDMA and Wi-Fi in different combinations, depending on the market. Regardless of which access technology is being used, all operator-controlled services, such as VoLTE and RCS Messaging, are handled by either the same IMS core network or different IMS core networks. The IMS network will need to support devices performing authentication and registration via cellular access and via Wi-Fi. Depending on the device implementation, these IMS registrations for the same user and device can be for VoLTE only, for RCS Messaging only or can be a combined registration for VoLTE and RCS Messaging. Depending on the configuration and implementation, RCS supports either single or dual IMS registration. The North America RCS deployment supports single IMS registration, while the European RCS deployment supports dual IMS registration.

VoLTE will always use the IMS APN to secure correct QoS settings and roaming behavior. Depending on the device implementation, the RCS Messaging service will be provided either on the IMS APN or on the Internet APN when using the cellular network. Specifications also allow the usage of Wi-Fi as defined in GSMA PRD IR.51. Figure 13 illustrates these connections.

The IMS network will have to distribute incoming traffic (forking/hunting) over the correct IMS registration, depending on service type and by that mitigate certain conditions and rejections in the device. After the forking process, the IMS network informs the packet core about the specific service in use. This enables the correct QoS rules to be applied in the packet core and in the RAN, depending on service type and operator policies.
Multi-device scenarios, such as a single user with a mix of devices with different capabilities, is a probable scenario that the network will have to accommodate. This task includes consolidation of capability discovery, forking or call hunting for the correct voice client in VoLTE and in RCS Messaging and use cases beyond those already described in RCS specifications.

IMS as a technology has been developed and standardized to be able to serve any device over any access using any service. IMS was originally built around having one IMS client per device and each IMS client provisioned with separate user information and credentials even if they share the same public phone number or SIP address.

The implementation of the IMS core and application layer logic ensures that the VoLTE and RCS Messaging services can be delivered in a homogeneous manner. This occurs independent of what the device implementation is and how many different devices a user may have. The IMS network has the capabilities to handle multiple registrations from separate SIP stacks/clients, multiple authentication methods used by the different SIP stacks in the device and the ability to sort out and distribute traffic in a controlled manner.

ROLE-PLAY BETWEEN APPLICATIONS AND SERVICES

OVERVIEW

This section describes the role-play between applications and underlying IMS. There are two key contributing enablers. One is the network transparency. Although networks remain in control of IMS services, they should be transparent for the applications using these services for transportation of application traffic.

To give an ancient comparison example, the networks were transparent to the introduction of fax, which used the underlying cross-network service telephony, for fax-to-fax traffic. It was rapidly deployed thanks to the network transparency. The other enabler is the possibility for application developers to access the IMS communication services inside the devices. This can be achieved by the 3GPP two-layer architecture, where devices provide IMS communication services APIs to applications. The new GSMA device API version will incorporate RCS 5.2 functionality. The RCS capability discovery mechanism can be used for showing the end-user the applications which mutually support with another end-user.

For OMA RESTful Network API for RCS, see [RCS – the platform for innovation](https://example.com).

THE ROLE-PLAY (HIGH LEVEL)

3GPP, OMA and GSMA have profiled and standardized multiple IMS Communication Services (IMS Commserv). These can be used for domestic and international cross-operator services connectivity. Applications can utilize these services as transportation channels to achieve application-to-application communication with both domestic and global reach.

An application provider should select one or more (one at the time) of these IMS communication services as underlying service for the transportation of the application traffic. By doing so, the application inherits the characteristics of the underlying service such as its QoS, session/media rules, NNI SLAs and SIP-AS functions. The application will, however, replace the IMS communication service default application user interface (UI). To summarize, an application provider selects an IMS communication service with a characteristic that suits the wanted application logic and performance. Figure 14 illustrates this hierarchy.
An IMS communication service is identified by a 3GPP IMS Communication Service Identifier (ICSI). An application is identified by a 3GPP IMS Application Reference Identifier (IARI).

**DEVICE VIEW**

Figure 15 shows the architecture of a device having implemented services with a reasonable high abstraction level API towards the applications, as a key enabler for a successful role-play between them. The IMS communication service ID (ICSI) acts like an IMS stack port, and the application ID (IARI) acts like an IMS communication service port.

An API call initiated from application 3 to service 1 or 2 should result in the device sending an initial SIP request including the IARI in the A-C (Accept-Contact) header, and the ICSI of the underlying service in both the PPS (P-Preferred-Service) and the A-C headers.
The device can inform the network about its supported applications and services during the IMS registration. This triggers the network to do an optimal forking of incoming application SIP requests. By using either of the two RCS capability discovery mechanisms (OPTIONS or Presence), the device can show the applications supported by other users in the address book.

An incoming application SIP request is dispatched from the IMS stack to the IMS communication service onto the targeted application via an API, according to Figure 15. The APIs shown above should be standardized by Service Delivery Organizations (SDO) such as OMA/W3C and utilized by the GSMA.

Application ID reservation can be done by email to the TSG CT WG1 Secretary. For more information, visit http://www.3gpp.org/Uniform-Resource-Name-URN-list.

**E2E VIEW**

Figure 16 shows that how operators can settle their interconnection agreements on the level of IMS communication services rather than on the applications using these services. As a result, networks can become transparent to application innovation which is a key enabler to boost new applications with zero network impact.

**Figure 16. Cross-Operator Application Traffic Flow Through an IMS Communication Service**

3GPP has always intended to allow applications to communicate using IMS services. This section is an effort to present this intention in a condensed way to shed some light on how devices and networks should behave to facilitate good role-play between applications and services within devices and between networks.
OVERVIEW

This section describes RCS in the communication evolution. It also discusses how operators can embrace this evolution and at the same time preserve their existing service business.

When the RCS specification is updated and evolved with new use cases and features, operators can incorporate them into their evolution of existing messaging, telephony and contact management services. RCS brings multi-device support to keep up with today’s users’ lifestyles. For example, it minimizes the difference between telecom services available on various user devices such as smartphones, tablets and laptops. Similarly, device vendors can utilize and build on their existing UIs to seamlessly support new use cases and features.

Marketing requirements differ in different regions, so there are two RCS implementation guidelines developed, one for North America (PRD RCC.59) and one for Europe to help OEM RCS implementations.

In order to secure communication interoperability and efficient network and device deployments, there should be one globally unified and consolidated profile of RCS-based features going forward. With the exception of the capability-discovery feature, each operator can then choose to implement features from the unified and consolidated RCS profile without needing to agree with other operators about implementing the exact same features.

MESSAGING EVOLUTION

RCS Messaging services can be used to eventually replace legacy SMS and MMS, thus eliminating the need for operators to maintain the associated legacy messaging infrastructure.

While RCS endorses and includes VoLTE and SMS over LTE services, it brings an enriched user experience. That’s accomplished via an integrated service experience in the device and through addition of new messaging features based on latest industry standards, such as OMA CPM. OMA Converged IP Messaging provides the standard evolution support for interoperability of the RCS Messaging features, multi-device handling, converged message store and interworking procedures of CPM Messaging with SMS and MMS.

RCS brings in the converged messaging view where legacy (SMS and MMS) and IMS-based messages, media and files are being exchanged between users via the appropriate technologies selected based on users’ access and service capabilities, as well as the existing services agreements at inter-operator NNI. The targeted experience is smooth messaging communication between a VoLTE-only enabled device, a RCS client and a legacy device, such as using SMS on 2G.

CPM Message Store defines a Common Message Store. This enables storage of IMS-based messages, legacy messages (such as SMS and MMS messages), multimedia and files exchanged. This design creates the possibility to accommodate other storage objects (e.g., VVM objects) in the future.

In deployments where a mix of VoLTE-only devices, RCS Messaging clients and legacy devices coexist, RCS provides adapted delivery models for each client/device type. The RCS Messaging Server may determine the type of client(s) that a user has by making use of IMS Core features such as third-party registration to obtain the IMS Communication Identifiers (ICSIs) supported by each client. Alternatively, client capabilities of the user can be obtained via RCS capability discovery mechanisms using OMA.
Presence or SIP OPTIONS. The RCS Messaging Server can then know the exact mix of RCS and SMSoIP capable (VoLTE) devices, and hence can determine the appropriate delivery means for each.

Different delivery mechanisms can be triggered in the RCS Messaging Server. One example is making an interworking decision to SMS or MMS due to an RCS recipient user not being registered in IMS. Another is a deferring decision to deliver to the RCS clients at a later time when the RCS client registers again in IMS. The Store & Forward feature for IMS messaging not only includes the deferred delivery of messages via IMS, but also the possibility to store them in the CPM Message Store. With this feature, users will never miss a message from a conversation even if they not actively engaged in that conversation.

Finally, group communication is made seamless because the RCS Messaging Controlling Function can establish group messaging, chat or file transfer between RCS and legacy users.

**USER EXPERIENCE**

Device vendors can implement new RCS features under the existing user interface. This enables operator control of the established communication. Figure 17 shows a messaging example.

Figure 17 shows a user who selects the messaging service (*left red circle*), selects Emma, types a message ('Hey Friend') and taps the 'send' icon (*right red circle*). At this point, the device may signal the network using a CPM messaging service (e.g., the RCS Chat feature technology) rather than using SMS as before. The IMS network will receive the signal to deliver the text message either by CPM messaging or SMS technology, depending on the receiver's capability. Until this point, the user would have the same service interface. However, the user will see an improvement in the messaging experience when "is composing" and "is displayed" indications pop up in the message window, such as when Emma has read the message and starts to type a reply.

The RCS specification also utilizes and endorses new features within the telephony service area, such as video calling, and adding video to a voice call. For voice calls, RCS has endorsed the technology of VoLTE (GSMA PRD IR.92). The voice call user interface could remain the same, as illustrated by the green phone button in Figure 17. A voice call is set up using the domain selection in force at a particular time, such as using VoLTE when domain selection is VoLTE, and using CS voice when domain selection is CS.

The call UI can be enhanced to include an add-video feature by presenting a video-camera icon, as shown in Figure 18. Tapping the video-camera icon results in the device sending a SIP request to the operator’s
IMS network to add video to the ongoing voice call and establishing a video communication. This is an example how an RCS feature can be discretely integrated into the existing UI.

**Figure 18. Extended Call-UI with Video**

Another feature endorsed by RCS is the ability to select a user from the address book and make a video call directly from the contact card. The UI could draw the user’s attention to this feature by highlighting a video-camera icon in the same UI-row as other operator-controlled features, such as voice or messaging.

The video camera icon in the address book contact card is shown when a capability discovery resulted in the knowledge that the remote end also supports video. However, a video service entry point may be selectable when a contact’s capability or availability for that service is unknown at a given point in time. Tapping the video camera icon allows the device to send a video call request signal to the operator’s IMS network. Figure 19 illustrates this scenario.

**Figure 19. Extended Call-UI Allowing Video Calls from the Address Book**

**EVOLUTION MODULARITY**

RCS enables flexibility, as well as incremental and independent service evolution tracks. Operators may choose to implement different messaging and/or telephony RCS features from the RCS specifications. As long as the features have been detailed in the RCS specification, and included in the global industry consolidated profile, the interoperability and user experience are secured using the capability discovery mechanism, thus avoiding trial and error.
A user can have multiple devices, such as a smartphone and a data-only device. These two devices may incorporate different features. If a data-only device supports video calling but the smartphone does not, the capability-discovery query from operator B’s customer will show that operator A’s customer has the video capability. The IMS network will reach and connect a video call request to the data-only device.

Over time, each operator introduces more and more of the features from the global industry consolidated profile of RCS features. The sum of interoperable communication continues to increase among all users.

ROAD TOWARD A GLOBAL RCS UNIVERSAL PROFILE

Because the majority of intelligent mobile devices today use Android, GSMA has been talking to Google about the possibility of Androidn adding native RCS support. The negotiation came to a fruition when Google acquired Jibe Mobile, an RCS solution provider, in September 2015 and subsequently welcomed by GSMA into the RCS ecosystem in 2016 MWC. GSMA, its mobile operator members and Google have developed a plan to support RCS in future Android operating system as soon as possible. On behalf of its mobile operator members, GSMA and Google have agreed in principle to develop a universal profile. This will be based on the GSMA developed RCS specifications and supported by future Android versions as the common RCS implementation for all Android-based devices. A two-step process has been developed:

1. Near term to develop pre-universal RCS profile for each RCS deployed regions (North America, Europe and China) to support existing RCS developments and prepare them to evolve into universal profile support when ready.

2. A common universal RCS profile will be supported by all Android-based devices when it is available.

A common universal RCS profile will speed up global RCS deployment and will provide a consistent end user experience with rich communications. The common universal profile provides a base for all interoperable RCS implementations for devices that use any operating system.

BEST PRACTICES FOR RCS SERVICE OFFERING

INTEGRATED USER EXPERIENCE VS. APP BASED

Early RCS Messaging implementations used downloadable clients, an approach similar to OTT messaging services. This implementation cannot integrate with the device-native VoLTE implementation. This approach didn’t fare well compared to OTT applications. As a result, the North America mobile ecosystem decided to implement native RCS and VoLTE clients to offer a seamless IP service experience.

Also, the integrated RCS clients will be able to use SIM card credentials, just like VoLTE clients, to access RCS services. There will be no sign-up screens or one-time passwords to use the service. Customers will enjoy the security and reliability of a native, carrier-grade service built by device OEMs with a reputation for high quality and reliability.

Integrated RCS Messaging clients will also share the same IMS Stack with other IMS services (e.g. VoLTE, ViLTE, etc.), as Figure 20 shows. This creates efficiencies on the client by having multiple IMS-based services running, providing a core set of functionality like authentication and security as compared to a downloadable app-based solution, where all of the services may have their own IMS stack (Figure 21). The network also gains efficiencies in maintaining one registration and one main identity for multiple services rather than multiple registrations for each service with possible different user identities that have to be maintained within the network nodes.
THIRD-PARTY HOSTED SOLUTIONS VS. CARRIER DEPLOYMENTS

In order to not invest heavily in network infrastructure to support RCS services, the third-party hosted RCS solution could be an alternative. This offers an operator the opportunity to launch RCS services quickly with minimum capital investment risks. As appealing as this may sound, there are some challenges faced with this deployment strategy:

- First and foremost, a good strategy for RCS is to offer an integrated experience on a common IMS stack for the multiple reasons mentioned in the previous sections. If choosing a hosted solution, due to the competitive mobile landscape, the operator will eventually have to have its own deployment in order to offer the same level of service and experience as other carriers.

- The operator will give up the ability to offer carrier-grade reliability for a core service such as RCS that can be realized on a carrier network.

- App-based clients provided by third-party hosted solutions for their network are offered as best effort. They are unable to provide QoS on LTE like an integrated client can on a carrier network.

Third-party hosted RCS solutions do provide benefits, but operators should be aware that the future is becoming clear for RCS services and the landscape is quickly changing and evolving to provide RCS as an integrated experience. RCS services offered will be considered core communication, displacing SMS/MMS and enhancing the messaging experience. Third parties may have a short-term advantage by providing a turnkey solution in helping a carrier deploy RCS. But that advantage might eventually become a disadvantage as other operators begin rolling out integrated RCS clients that third-party solutions might not be able to match in some aspects.

RCS SERVICE GRADE – RELIABILITY AND AVAILABILITY

Reliability and availability will be two of the key features that differentiate integrated RCS offerings from OTT communication applications. Integrated clients will be able to take advantage of QoS offered on LTE by means of assigning QoS Class Identifier (QCI) values to each bearer (IP flow) between the user’s mobile device and the EPC. By assigning QCI values to signaling and media bearers, RCS messaging traffic can be prioritized on the network over general Internet traffic. This provides users with higher availability and reliability for accessing the EPC in saturated cell sites. It also improves performance because signaling and media bearers will not be competing with Internet traffic. Also, the signaling and media flows might be traversing the same nodes as VoLTE traffic, which already have carrier-grade hardware and redundancy.
built in. Thus RCS Messaging traffic will take advantage of the current infrastructure in place for VoLTE designed for high availability and reliability.

**RECOMMENDATIONS FOR RCS INTEROPERABILITY BETWEEN OPERATORS**

Recommendations for interoperability between operators for the purposes of exchanging RCS traffic are addressed in "GSMA PRD RCC.54 – RCS Interconnection Guidelines." The document focuses on information originally gathered from North American deployments covering RCS5.1 services. The document will evolve with the future RCS versions to incorporate additional feedback from RCS deployment and interconnection activities across different regions and profiles, as well as the evolution of the RCS services. Consequently, the document covers a profile of “GSMA PRD IR.90, the RCS Interworking Guidelines.”

However, GSMA PRD RCC.54 is not limited to guidelines for RCS Messaging service interoperability between operators, but includes recommendations for interoperability for VoLTE and conversational video services. These recommendations are applied in the determination of mutual agreement for interconnection between operators. In order to define mutual agreements for interoperability, an Interworking Form for IMS Based Services tailored to RCS services is provided in Annex B of GSMA PRD IR.90, including the NNI details and parameters associated with the RCS services they plan to interconnect.

In support for interconnection efforts in market deployments, the GSMA has also provided the PRD RCC.51NNI Test Cases Specification as part of RCS5.1 v4.0 release, as well as an overall Interworking Template Agreement in GSMA PRD AA.69. All of these help define the service-level interworking between interoperability partners.

**RCS VS. OTT COMMUNICATION SERVICES**

RCS services include presence and user-capability exchange, IM, group chat, file transfer, image share, video share, video calling, network-based contact list and SMS/MMS interworking. These features are becoming essentially table stakes for IP communications solutions. Successful OTT communications apps may start with one or two services, such as VoIP, video calling or messaging. Over time, however, most of them ultimately evolve to offer a full suite of communications services. A recent example of this trend is WhatsApp’s announcement that it has launched VoIP services to complement its messaging service.

So why do we have this trend toward full service? The barrier to entry into the IP communications market continues to lower, bringing in more and more players/competitors. In order to attract and retain customers and increase opportunity for revenue, a rich user experience is required.

Mobile operators currently provide a globally interoperable communications ecosystem supporting universal reach for voice and messaging (xMS). There are now almost 7 billion mobile subscriptions in the world. This global communications ecosystem is not only an expected social contract for most consumers, but also a basic underpinning for the world’s economy. Businesses must be able to reliably call other businesses and their customers. Customers must be able to reliably call businesses. Therefore, the transition from today’s PSTN/PLMN to tomorrow’s next generation all-IP communications ecosystem must ensure a smooth transition. The sunset of the PSTN/PLMN must be done with care while acknowledging and addressing the fact that there are a growing number of OTT providers that are disrupting the mobile broadband service and application ecosystem.

The original vision of IMS – to provide a globally interoperable communications ecosystem supporting basic and advanced services – should be the end goal for the transition to an all-IP world. The question is, should this end goal be achieved through mobile operators deploying IMS and RCS or through a yet-to-be defined “road” via a collection of OTT providers? For the answer to be mobile operators and IMS, it is critical that
we identify the new business models and transition points that mobile operators need in order be successful and prosper in the business of communications providers.

RCS is the vital part of this strategy. It provides the features needed for IMS to be a full-service communications system that will enable mobile operators to compete. RCS also provides the communications features that younger generations may prefer to use. There is recognition in the industry that a trend toward texting over talking continues to grow, especially with millennials. Mobile operators must be ready to address the needs and preferences of all of their customers. SMS does not sufficiently meet the needs of this growing market, and hence those people are using OTT apps. However, if mobile operators offered these advanced communications services natively and with global interoperability, the need to use OTT services will decline.

By some estimates, two-thirds of people worldwide have access to SMS. This translates to mobile operators having billions of messaging users today sending trillions of messages a year. Given this, mobile operators should not simply give up on messaging to the OTT providers as they transition to an all-IP network.

Instead, mobile operators should use RCS to provide xMS service. Users don’t care what technology is behind their built-in messaging service as long as they can use the service seamlessly. So if the mobile operator keeps the same messaging pricing plans and possibly the same messaging user interface as xMS for RCS, subscribers will use RCS because it works just like xMS with rich content: built into the device with universal reach. Subscribers really don’t care if a different technology is behind the service. Once basic RCS is in place, mobile operators can introduce the advanced RCS capabilities, which will enable new revenues and provide the stickiness necessary to keep subscribers using their communications service rather than an OTT service.

There are also other possible approaches for RCS deployment that may be equally or more successful.

**IS RCS A COMPETITOR OR COMPLEMENTARY TO WEBRTC?**

WebRTC is an emerging HTML5 technology that enables native integration of IP communications services into the browser. WebRTC is flexible and does not mandate or specify any particular signaling protocol for communications or call control. Instead, it allows developers to use the signaling protocol they prefer. This is true for real-time voice and video sessions and also for messaging communications services such as RCS.

RCS provides table stakes features that successful OTT providers offer. As communications services evolve, mobile operators will have to support the ability for subscribers to access their services from many different devices and from the Web via WebRTC. OTT providers are or will be providing Web communications experiences, as well. In addition, this means that there is a need for network-based versions of capabilities for functions such as address book, presence and ringtones because devices such as TV sets, laptops and tablets will not have these capabilities locally.

In the early standardization days of WebRTC, there was some confusion about whether WebRTC standards alone would be sufficient to support voice and video communications sessions. Today, it is well understood that an additional layer of call control signaling, such as SIP, is needed. WebRTC standards chose to leave this open to the implementer. However, at times there is still similar confusion as to whether WebRTC’s data channel is sufficient to support messaging communications services or if a higher layer protocol is needed here, as well.

To clarify, at IETF 86, it was agreed that:
- SDP would only be used (for now) to negotiate the creation of an SCTP association and stack (SCTP/DTLS/UDP) to support data channels (i.e., an m-line would be used to create the SCTP association).

- No SDP mechanism to define individual data channels will be specified at this time.

- A new data channel is created in an in-band declarative fashion by sending an open message on a new stream with related data channel attribute information:
  - Label (non-interpreted token for use by the application).
  - Protocol (IANA standard field to specify the protocol to be transported in the stream).
  - Reliability characteristics (e.g., ordered/not, reliable/not, max re-xmit).
  - Priority (to be used by the browser to prioritize streams).

- Peer uses same stream ID and characteristics in other direction if it accepts the data channel.

- Peer can only reject a new data channel by closing the stream.

- It is left to new protocol-specific RFCs to define potential SDP extensions to negotiate within SDP the use of a specific protocol on a data channel (only raw data to be defined now).

From time to time, there are public announcements about an OTT provider using the raw data channel to provide a messaging or file transfer service. These announcements make it appear as if the OTT application did not need to define or use an additional higher layer signaling protocol, such as MSRP. However, upon closer inspection, it should be understood that to create a data channel, it is necessary to indicate the following in the open message:
  - Label (non-interpreted token for use by the application).
  - Protocol (IANA standard field to specify the protocol to be transported in the stream).
  - Reliability characteristics (e.g., ordered/not, reliable/not, max re-xmit).
  - Priority (to be used by the browser to prioritize streams).

Therefore, even if very simple, there needs to be some designed protocol – which can be proprietary – that runs on top of the data channel for the messaging service. In other words, and for example, peering entities of an OTT WebRTC data channel app must understand that “Label = A” means that this data channel is used for some service, such as transferring a file between peers. This understanding is basically part of a protocol procedure. RCS is simply an example of a collection of standardized protocols that can be run on top of WebRTC via WebSocket or data channel.

At this point, it should be clear that RCS is not a competitor to WebRTC but instead is very complementary to it. The rest of this section will describe the options for how the various RCS services can be offered within a WebRTC-enabled browser.

RCS is a collection of communications services. Some of these services are signaling only. Others require a bearer. Figure 22 illustrates an IMS-RCS architecture for WebRTC.
1. **RCS signaling-only features** – There are a number of RCS services that only require the signaling channel:
   a. XDMS uses XCAP protocol.
   b. Presence uses SIP (SIMPLE) protocol.
   c. Instant Messaging uses SIP (SIMPLE) protocol for pager mode IM.

2. **RCS signaling and bearer features** – There are a number of RCS services that use both a signaling channel and a bearer channel:
   a. IM uses MSRP protocol over bearer channel and SIP for signaling, such as for large message mode and session mode.
   b. File transfer uses MSRP over data bearer channel and SIP for signaling. In addition, RCS also provides an HTTP bearer for file upload/download as alternative.

Figure 23 illustrates IMS-RCS signaling for WebRTC:
For RCS signaling-only features, the implementation options are:

1. RCS XDMS features including contact list support, has a defined XCAP interface.
   - WebRTC IMS client can use XCAP signaling to the XDMS server.

2. RCS presence can be supported by the WebRTC app in one of two ways:
   - Use SIP-over-Web socket. Signaling is handled through the WebRTC GW.
   - Use OMA presence RESTful API.

3. RCS IM in pager mode uses a single IM carried in signaling only. There are two implementation options:
   - Use SIP-over-Web socket. Signaling is handled through the WebRTC gateway.
   - Use OMA Chat RESTful API.

Figure 24 illustrates an IMS-RCS architecture for signaling and media bearer.
For RCS features that require signaling and bearer, the implementation options are:

1. RCS IM in large message mode and instant mode, as well as file transfer uses MSRP over data bearer channel in addition to SIP signaling:
   - Use SIP over Web socket for SIP signaling handled through the WebRTC gateway.
   - MSRP bearer protocol. Bearer is setup over the WebRTC data channel for transport of instant messages and file transfers.

Other IMS services not defined by RCS can use the WebRTC data channel in the same manner that’s being proposed for RCS MSRP:

2. T.140 is defined as a SIP-based real-time text service. This can be used for the hearing impaired and other use cases such as emergency service:
   - Use SIP-over-Web socket. Signaling is handled thru the WebRTC gateway.
   - T.140 over SCTP bearer is set up over the WebRTC data channel for transport of IMs.

3. BFCP is used for Web conferencing services, such as webinars, that support multiple communications features including texting:
   - Use SIP-over-Web socket. Signaling is handled thru the WebRTC gateway.
   - BFCP over SCTP bearer is setup over the WebRTC data channel for transport of IMs.

Note that items (2) and (3) are not RCS services. They are IMS services that could use the data channel in the same manner that is being proposed for RCS MSRP. Therefore, they have been included in this list in order to provide a more holistic explanation.

Finally, we conclude this section with architecture and protocol stack diagrams – Figures 25 and 26 – for the support of VoLTE and RCS services in a browser enabled by WebRTC and as is being defined in 3GPP.
Figure 25. WebRTC and IMS/RCS Service Architecture

Figure 21. WebRTC and IMS/RCS Protocol Stacks
The demand for a connected lifestyle is stronger than ever. The enabling networks are in place. Devices are smarter and more capable, and there’s an ever-expanding set of applications that provide communications capabilities.

RCS can reinvent communications to inspire new and more engaging conversations. RCS is a critical component to offer this new conversation experience and make our life even more connected, providing some key attributes such as:

- **Social presence** – A key principle of RCS is to make interactions more frequent, easy and vivid through a very rich social presence service. While social presence is similar to social websites, it includes many security, authentication and privacy mechanisms that make it very safe.

- **Multi-device** – RCS has been designed from the start for multi-device support, allowing subscribers to communicate through the connected-device that suits them best. This means that it can be used by subscribers with different devices. Each subscriber also can have multiple devices, such as a mobile phone and a tablet, connected at the same time with the same account. RCS defines mechanisms to aggregate presence information from multiple devices into a single view and enables users to choose on which devices to reply to an IP-message, for example. Ultimately RCS enables people to connect, share and organize their connected life.

Refreshing the key aspects of our daily communications — including voice, messaging, video and digital media — entails adding nuanced personalization, contextual awareness, enhanced privacy, greater usability, immersiveness, unified access to contacts and information about our conversations. The result is a more engaging and fulfilling conversation experience:

- **Connect** – RCS invigorates communications by enabling conversations to be initiated simply and intuitively from any screen on any network. RCS increases the effectiveness of communications, allowing people to see when others are available and offering more connection options. It can enrich social networking interactions, allowing people to securely talk, chat and connect face-to-face with individuals and groups.

- **Share** – RCS can enhance conversations with reliable sharing services on any screen. RCS can enable people to share what they do, what they see and what they have in common with their contacts with a single click. RCS can also enable users to record and file what they share and receive in a secure cloud diary so it is always accessible.

- **Organize** – RCS can help people take control of their communications by providing a unified inbox and archive for all their conversations, making it easy to prioritize important messages.

RCS can also unify contacts across social networks and various communication services to enable new level of organization and convenience. As people will be better organized, they’ll be able to reconnect more often and spend more time using their services.
MASHUP SERVICES

Several types of communication innovation can deliver the new conversation experience:

- **Communications as a service.** This concept is familiar. Operators can already extend voice, video and messaging services from traditional devices, such as smartphones, to new devices, such as tablets, laptops and TV sets, and deliver a new RCS experience on terminals.

- **Communications as a feature.** Another experience is the deployment of RCS communications as a feature, such as embedding communications inside websites. Instead of building their own communications system, website owners in sectors such as banking and retail can lease communications from the operator. This opens up opportunities for analytics and an improved consumer experience, such as the agent and shopper sharing a screen while completing a form or interacting with product images.

- **Communications inside apps.** Innovation will focus on the creation of new types of apps with communications embedded inside. In this case, the “conversation” is a smaller piece of a much larger pie. For example, a social sports application can be enriched with real-time communications of voice, video, presence, IM and content sharing.

- **Extending communications into 3D, holographic solutions and adding context.** New technologies will continue to emerge that can be leveraged to enhance the communications experience and enable new use cases with new business models. 3D and holographic displays are an example of upcoming technologies that will be available on smartphones. What are new use cases that can be enabled with these technologies? Here is a proposed phasing of scenarios:
  - 3D/hologram messaging
  - 3D/hologram real-time, interactive sharing
  - 3D/holographic telephony

The following illustrates possible examples of use cases for the proposed early phases:

1. Alice “finds” or buys a “congratulations” hologram from the Internet or a mobile operator and uses her RCS hologram messaging service to send to Bob and Carol. (In the future Alice might be able to create her own holograms.) Sending the hologram may mean using one of multiple approaches:
   - Send the hologram with procedures that include store-and-forward capabilities to both Bob and Carol.
   - Or while Alice is in an IM/group chat session with Bob and Carol, she sends a hologram similar to a next-gen sticker or emoticon.

2. Alice (operator A) is in a VoLTE conference call with Bob (operator B) and Carol (operator C). Alice shares the hologram with Bob and Carol. Bob moves or modifies the hologram. Both Alice and Carol see the changes in real-time. This and the previous example can also be useful in enterprise scenarios as well.

These use cases may be realized through extensions to existing RCS features.
Contextual information about participants in a session—such as their latest tweet, their mood, most recent e-mails exchanged between the call parties, other people who have been on common sessions and so on—could potentially be provided as part of the family of presence services.

RCS provides a foundation for extending communications into the future leveraging new technologies either through further capabilities specifications or via exposure of APIs.

**API EXPOSURE TO DEVELOPERS**

APIs are programmatic interfaces that expose the mobile operators' network capabilities by hiding the complexity of the underlying network. These interfaces may be abstract, defined independent from underlying technologies such as operating system, network access or binding protocols. Therefore, APIs act as an abstraction layer between the network and software development platforms.

Operators must adapt their strategy to include APIs for both Web-based and traditional communications. By unifying mobile devices, browsers and applications through APIs, operators can differentiate from Web-only and telecom-only competitors and instead engage subscribers who prefer to use both the Web and telecom services.

Expanding with open APIs will also increase operator brand's reach and generate new revenue by opening communications to innovation. It makes innovation quicker and more affordable while helping operators differentiate by branding more robust services onto the Web.

By exposing APIs to developers, operators will benefit in the following ways:

- **Accelerated innovation** – Exposing APIs enables delivering new applications faster to the market by reducing time to market from years to months.

- **Differentiated services** – Attracting Web developers helps operators to create unique and differentiating services for consumer, business and vertical industry markets, ultimately generating new revenues.

- **New Markets** – It enables exploration of new wholesale business models as part of an end-to-end API strategy for exposing the value of the network to developers and generating new revenues from API usage.

Exposing certain network capabilities (RCS APIs) to developer communities enables software developers to create new applications, enhance the existing operators' services and also to open up new business opportunities (e.g., automotive, health care) across consumer and enterprise markets. This approach encourages innovation in the application and service space and allows users to be presented with a large and diverse range of third-party applications to choose from.

The OMA has produced RESTful network APIs based on requirements received from many industry forums, such as 3GPP, GSMA, Small Cell Forum and BEREC. OMA RESTful Network APIs are modular APIs that target specific services (e.g., messaging, presence, voice and video calls, location) and network capabilities (e.g., QoS) that are exposed to third-party developers but can also be used over the UNI by a lightweight Web-based client.

The OMA and the GSMA have collaborated and complement each other to provide industry with a set of standardized network APIs. A standardized API approach allows more developers to build apps and services for operator networks, so users could be provided with a wide variety of new applications and services every day.
At the GSMA’s request, based on market requirements, the OMA has defined several RCS API profiles of its RESTful Network API specifications to be used for the GSMA’s RCS projects. An API profile is a subset of supported operations and/or parameters, selected from an existing OMA network API or API protocol binding. The RCS API specifications contain tables with information on what operations are mandated in the profile that must be implemented, in order to claim conformance with the specific profile as required by the GSMA (e.g., OneAPI, RCS API).

Profile v1.0 functionality of the RCS Profile of RESTful Network APIs defines subsets of the following APIs:

- Notification Channel
- Chat
- File Transfer
- Third-Party Call
- Call Notification
- Video Share
- Image Share
- Anonymous Customer Reference
- Capability Discovery.

Profile v1.0 has been subsequently evolved up to Profile v3.0. Figure 27 illustrates this evolution.

The OMA also created and published API governance specifications, including RESTful APIs Guidelines, Templates and Best Practices.

As part of the support functions for the network API framework, the OMA has published a common delegated authorization framework. The framework is based on IETF OAuth 2.0, which enables a third-party application to obtain limited access to an HTTP service. This can be either on behalf of a resource owner by orchestrating an approval interaction between the resource owner and the HTTP service, or by
allowing the third-party application to obtain access on its own behalf. The OMA enabler is called Autho4API (Authorization Framework for Network APIs) and it profiles and extends the OAuth 2.0 framework.

Autho4API provides delegated authorization functionality that can be used with the OMA RESTful network based APIs. It enables a resource owner, owning network resources exposed by a network API, to authorize third-party applications (desktop, mobile and Web applications) to access these resources via API on the resource owner’s behalf.

The OMA has just started work on additional API framework functionality besides authorization and authentication, covering infrastructural policy, business policy, assurance/O&M, accounting and interconnect access.

CONCLUSION

VoLTE has gained traction in North America with commercial launches by multiple Tier 1 operators. As other Tier 1 and regional operators launch VoLTE, service performance and user experience continue to be refined so that the VoLTE service performance is either on par or better than legacy CS on 2G and 3G networks.

IMS infrastructure in operator networks is being enhanced to provide real-time IP services such as voice and video. OTT players have messaging platforms and RCS provides a competitive offering. RCS Messaging natively integrated into OEM devices and chipsets offers a seamless out-of-box experience to the end-user that cannot be easily matched by others. After refining VoLTE, ViLTE and RCS Messaging in their own networks and for their own users, the logical next step for operators is to provide the same level of experience to other users via interconnections for a truly global reach of these services.

There are some outstanding regulatory requirements around emergency calls and lawful interception that have to be addressed and refined for this to become reality. However, the wheels are already in motion, with the 3GPP standardization body working on proposed solutions and incorporating finalized solutions into normative specifications.

The position of VoLTE and RCS Messaging is to provide mobile operators with new features within the framework of the existing three major service tracks: telephony, messaging and contact management. Moreover, a strategy for RCS Messaging is to strive for defining a universal profile avoid fragmentation among terminals and networks.

The vision is to have a single consolidated or universal specification where the evolution of RCS should be determined by the operators themselves in the pace that suits their profile and market. This is possible thanks to the capability discovery mechanism that avoids the end-user trial and error experience in case two operators in the same domestic market would choose to bring in slightly different sets of features from the latest RCS specification into their service portfolio.

A key benefit is the seamless introduction of new features using existing UIs. This stimulates the usage of new operator controlled features and maintains the operator’s relevance as service provider for the subscriber. This will enable mass adoption and delivery of RCS Messaging services similar to voice and SMS today. Operators can introduce individual features from a unified profile at a pace that addresses market demands.
## ABBREVIATIONS

<table>
<thead>
<tr>
<th>Abbreviation</th>
<th>Description</th>
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<tbody>
<tr>
<td>AMR-WB</td>
<td>Adaptive Multi Rate – Wideband</td>
</tr>
<tr>
<td>ATCF</td>
<td>Access Transfer Control Function</td>
</tr>
<tr>
<td>ATGW</td>
<td>Access Transfer Gateway</td>
</tr>
<tr>
<td>API</td>
<td>Application Program Interface</td>
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<td>CS</td>
<td>Circuit Switch</td>
</tr>
<tr>
<td>CPM</td>
<td>Converged IP Messaging</td>
</tr>
<tr>
<td>EPC</td>
<td>Evolved Packet Core</td>
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<tr>
<td>EVS</td>
<td>Enhanced Voice Service</td>
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<tr>
<td>GSMA</td>
<td>GSM Association</td>
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<tr>
<td>HPLMN</td>
<td>Home PLMN</td>
</tr>
<tr>
<td>IBCF</td>
<td>Interconnection Border Control Function</td>
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<tr>
<td>LBO</td>
<td>Local Break Out</td>
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<tr>
<td>MMS</td>
<td>Multimedia Messaging Service</td>
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<tr>
<td>NAT</td>
<td>Network Address Translation</td>
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<tr>
<td>NENA</td>
<td>National Emergency Number Association</td>
</tr>
<tr>
<td>NNI</td>
<td>Network-Network Interface</td>
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<tr>
<td>OMA</td>
<td>Open Mobile Alliance</td>
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<tr>
<td>OEM</td>
<td>Original Equipment Manufacturer</td>
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<tr>
<td>OTT</td>
<td>Over-The-Top</td>
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<tr>
<td>PLMN</td>
<td>Public Land Mobile Network</td>
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<tr>
<td>PRD</td>
<td>Permanent Reference Document</td>
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<tr>
<td>PSAP</td>
<td>Public Safety Answering Point</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switch Telephony Network</td>
</tr>
<tr>
<td>QoS</td>
<td>Quality of Service</td>
</tr>
<tr>
<td>RCS</td>
<td>Rich Suite Communications</td>
</tr>
<tr>
<td>SBC</td>
<td>Session Border Control</td>
</tr>
<tr>
<td>SDP</td>
<td>Session Description Protocol</td>
</tr>
<tr>
<td>SLA</td>
<td>Service Level Agreement</td>
</tr>
<tr>
<td>SMS</td>
<td>Short Message Service</td>
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<tr>
<td>SRVCC</td>
<td>Single Radio Voice Call Continuity</td>
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<tr>
<td>ViLTE</td>
<td>Video over LTE</td>
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<tr>
<td>VoLTE</td>
<td>Voice over LTE</td>
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<tr>
<td>VPLMN</td>
<td>Visited PLMN</td>
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<tr>
<td>WebRTC</td>
<td>Web Real Time Communication</td>
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<tr>
<td>XCAP</td>
<td>XML Configuration Access Protocol</td>
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The mission of 5G Americas is to advocate for and foster the advancement and full capabilities of LTE wireless technologies, including LTE-Advanced and beyond to 5G, throughout the ecosystem's networks, services, applications and wirelessly connected devices in the Americas.

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