Coexistence of GSM, HSPA and LTE

May 2011

CONTENTS

EXECUTIVE SUMMARY

This whitepaper covers the coexistence of GSM, HSPA and LTE and migratory aspects from Rel-7 to Rel-8 and beyond based on the 3GPP specification. The topics covered include:

- The Radio Access Network Aspects of the Coexistence
- The Core Network Considerations
- The Quality of Service (QoS) Requirements
- The Future Of Voice/Messaging Services on LTE including CSFB, SRVCC and VoLTE and IMS Services
- Regulatory Aspects
- Multimode and Multiband Devices and Machine Type Communications for Coexisting Networks

The chapter on Radio Access Networks discusses the spectrum considerations and the concept of Multi-Standard Radios (MSR) that will be key to coexistence in the future. Impact on antennas is briefly touched upon as well. Then the chapter discusses UICC/USIM roaming. It closes with coverage triggered session continuity from LTE to WCDMA/GERAN Networks.

The chapter on core network considerations provides a core network overview i.e. the Evolved Packet Core (EPC) followed by migratory aspects related to pre-Rel-8 mobility as well as Rel-8 based mobility. Subscriber data aspects are touched upon related to HLR evolution to HSS. Finally, some roaming considerations are presented based on Home Tunneling as compared to Home Tunneling with the possibility of local breakout.

In this whitepaper, QoS in 3GPP is explained and the concepts key to delay sensitive packet data e.g. GBR (guaranteed bit rate) and non-GBR are presented. The difference between pre-Rel-8 and Rel-8 functionality e.g. QoS classification for LTE vs. for WCDMA Networks is also presented.

The LTE Services addressed include Circuit Switched Fallback (CSFB) - the mechanism to provide voice services on the GSM/WCDMA network before IMS enabled LTE voice service is available. Voice over LTE (VoLTE) based on IMS is then presented and supplementary IMS Services are introduced. Then moving out from IMS Voice capable LTE coverage to WCDMA/GSM only coverage, Single Radio Voice Call Continuity SRVCC is explained based on both Rel-9 and Rel-10 architecture. Roaming considerations also are briefly touched upon and the concept of IMS Centralized Services ICS is introduced. The chapter finishes with a description of messaging services over LTE based on SMS over SGS (specified in CSFB) and IMS Messaging.

The chapter on Regulatory Aspects covers Lawful Intercept, Emergency Services, Non-Voice Emergency Services, Priority Services and Commercial Mobile Alert System.

Then in the chapter on devices for seamless migration the spectrum and technology requirements of devices for coexistence are presented. Multimode and Multiband devices with MIMO capability are discussed. The requirements for supporting Voice over LTE (IMS Voice) are also presented.

The final chapter presents an update on Machine Type Communications (MTC) as reflected by standardization activities.

1 INTRODUCTION

The whitepaper covers the end-to-end considerations for the successful coexistence of GSM, HSPA and LTE.

1.1 COEXISTENCE OF GSM, HSPA/WCDMA AND LTE

As shown in Figure 1, devices with the three technologies: GSM, UMTS and LTE will co-exist for the foreseeable future. The objective of this whitepaper is to identify all considerations from the RAN, Core and Service layer to ensure a seamless migration for the end-user.

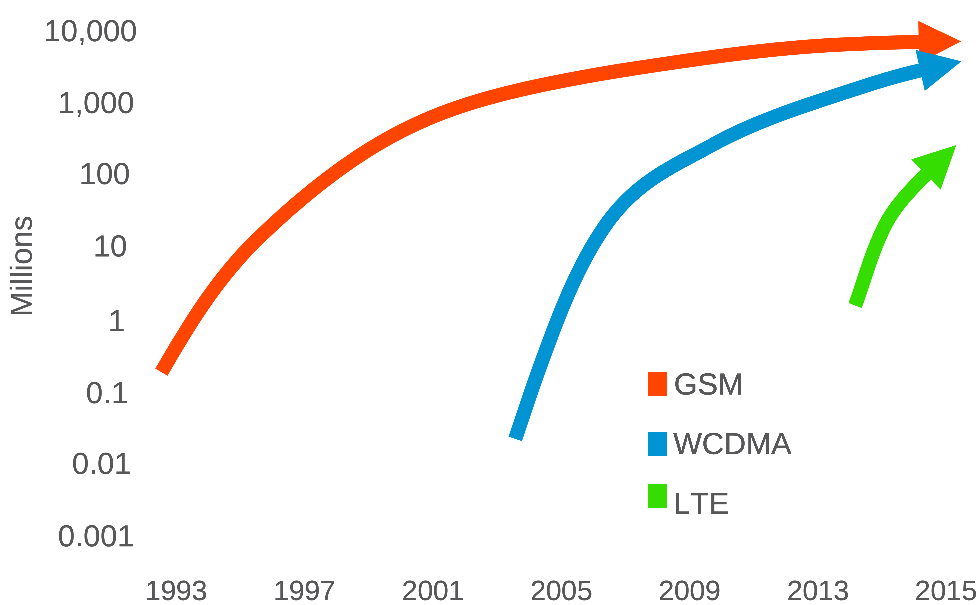

Figure 1: Timeline for GSM, HSPA/WCDMA, LTE Subscriber/Device Evolution

1.2 THE SYSTEM ARCHITECTURE EVOLUTION ASPECTS

The broad objectives of the SAE (System Architecture Evolution now referred to as EPC – Evolved Packet Core) were to evolve the 3G access technologies and their supporting GPRS core network by creating a simplified All-IP architecture to provide support for multiple radio accesses, including mobility between various access networks, both 3GPP and Non-3GPP standardized technologies.

The goal is to provide a system evolution aiming at improving the performance of the existing system, high performance handover between 3GPP accesses, easy migration to EPS, and manageable impacts on roaming infrastructure upgrades, while reusing some of the very strong 3GPP architecture principles like clear separation of control and user plane operation, and All-IP.

Requirements for reducing the latencies and delays in the radio access have imposed that the RAN adaptation loops be radically optimized latency-wise by moving some of the supporting functions from the old centralized node RNC to the new eNB, in the end the RNC being completely eliminated from the RAN architecture via the transfer of its functions to the new Core Network (CN): EPC. That is equivalent to changing the borderline between the E-UTRAN and the EPC as compared to previous 3G systems.

However, this revolutionary change has resulted in the benefits of a flat architecture of both E-UTRAN and EPC, with plenty of opportunities to optimize the backhaul, transport, as well as the overall network management for the operators.

One major characteristic of the new system architecture conceived for EPS is the fact that it adopted a “network based mobility model” (with two flavors: GTP or PMIP), a choice that is essential in ensuring that the UE does not act differently depending on the protocol used by the network entities.

This is one of the most powerful features of the EPC: the major interface in the Core Network, S5/S8, between the only two types of nodes in the flat EPC, S-GW and PDN GW, is specified in two different variants, one utilizes the GTP protocol which is used in the legacy GSM/GPRS and WCDMA/HSPA networks and the other uses the IETF PMIPv6 protocol.

1.3 SCOPE

The document focuses on the various important segments concerning the successful integration of GSM, HSPA and LTE as listed below.

VOICE SOLUTIONS

3GPP Voice over LTE (VoLTE) Solutions are discussed, which include CSFB (Circuit Switch Fall Back), IMS based VoLTE and SRVCC (Single Radio Voice Call Continuity).

VOLGA is not considered as it is not a 3GPP Specification. VOLGA is based on the existing 3GPP Generic Access Network (GAN) standard, with the purpose of extending mobile services over a generic IP access network. The main concept in VOLGA is to connect the already existing Mobile Switching Centers to the LTE network via a gateway and make use of existing GAN concepts.

MOBILITY

The document presents EPS mobility solutions. Mobility with non-3GPP is capability of LTE that is not considered in this document.

ROAMING ASPECTS

From a coexistence perspective for roaming services of devices between pre-Rel-8 networks and Rel-8 and beyond networks, the document presents the following issues:

- The interfaces supported by the two networks to communicate and seamlessly transfer service flows across the two networks desirably in both directions
- The ability of devices to operate in diverse multiple bands around the world to help subscribers roam
- The ability of devices and subscribers to authenticate themselves with the roaming network

RADIO NETWORK DESIGN AND PERFORMANCE

The radio link throughput and latency performance for HSPA, HSPA+ and LTE networks is discussed at length in the 3G Americas whitepaper by Rysavy Research[1] and therefore is not addressed in the current whitepaper on coexistence. Another aspect related to coexistence and network design is antenna migration for the radio network to support multiple frequency bands and technologies with MIMO capability. That discussion is also covered in another 3G Americas whitepaper.[2]

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[1] *Transition to 4G: 3GPP Broadband Evolution to IMT-Advanced,* Rysavy Research/3G Americas, Sept 2010
[2] *MIMO and Smart Antennas for 3G and 4G Wireless Systems Practical Aspects and their Deployment*, 3G Americas, May 2010

2 RADIO ACCESS NETWORK CONSIDERATIONS

The Radio Network Considerations for coexistence of LTE, HSPA and GSM networks include the spectrum requirements, roaming and coverage triggered handovers to WCDMA/GSM. In the section on spectrum considerations the concept of Multi-Standard Radios (MSR) is presented.

2.1 TECHNOLOGY COEXISTENCE

It takes years to complete a nationwide build-out of a new technology. A potential scenario of LTE / HSPA and GSM/EDGE Build out is illustrated in Figure 2.

Figure 2: Illustrative GSM/EDGE, WCDMA/HSPA and LTE Buildout

It makes sense to continue the evolution of WCDMA/HSPA (to HSPA+) so that the user experience does not degrade when the user is out of LTE coverage.

2.2 SPECTRUM CONSIDERATIONS

The FDD frequency bands relevant for GSM, HSPA and LTE coexistence in the Americas are highlighted in the Appendix in Table 7.

MULTI-STANDARD RADIO

Due to this lack of spectrum, operators must optimize utilization of existing spectrum. Multiradio solutions enable operators to share some components like the baseband for several carriers of the same technology, while leveraging frequency specific radios. The 3GPP definition of MSR in 3GPP, TS 37.104 is:

"*Base Station characterized by the ability of its receiver and transmitter to process two or more carriers in common active RF components simultaneously in a declared RF bandwidth, where at least one carrier is of a different RAT than the other carrier(s). MSR implies different technologies on the same frequency band*."

A scenario for MSR radio is depicted in Figure 3. GSM Technology is utilized in the allocated frequency band. After some time there is a mixed configuration of both GSM and WCDMA within the same frequency band. Further along the time line the entire frequency band is allocated to WCDMA with no change in hardware.

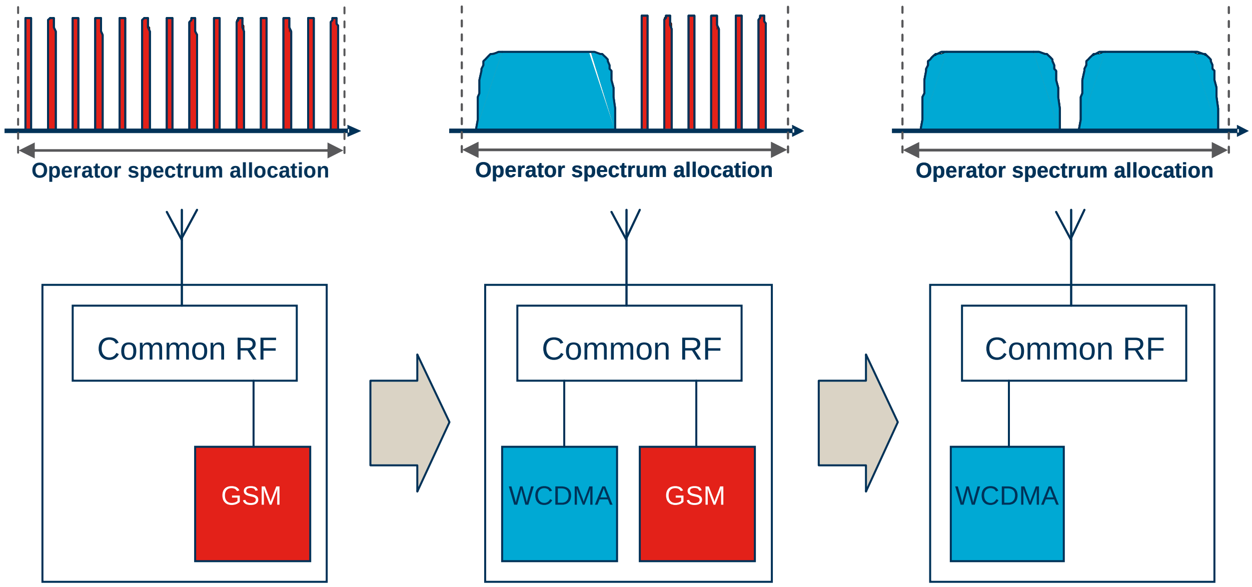

Figure 3: Example of MSR the same radio hardware is used for GSM, GSM/WCDMA and finally WCDMA only

The practical advantage of MSR is that for a given band the same investment can potentially ease the migration from GSM to WCDMA/LTE. An MSR capable radio for band category 2 can enable this migration.

The scenario is that GSM is reaching maturity for an operator and investing further in "GSM only" capable radios is not cost-effective. On the other hand, WCDMA is in growth mode with a greater lifespan. Also LTE will potentially be introduced at some point. An MSR radio can be used for GSM to start with and then be converted to WCDMA/LTE later on as shown in Figure 4.

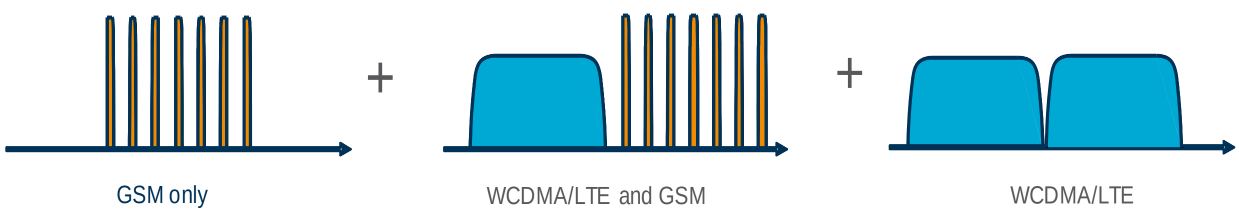

Figure 4: Migration from Spectrum used for GSM Only to WCDMA/LTE and GSM to finally WCDMA/LTE

MSR requirements are applicable for band definitions and band numbering as defined in the specifications TS 45.005 [2], TS 25.104 [3], TS 25.105 [4] and TS 36.104 [5]. For the purpose of defining the MSR BS requirements, the operating bands are divided into three band categories as follows:

- Band Category 1 (BC1): Bands for E-UTRA FDD and UTRA FDD operation
- Band Category 2 (BC2): Bands for E-UTRA FDD, UTRA FDD and GSM/EDGE operation
- Band Category 3 (BC3): Bands for E-UTRA TDD and UTRA TDD operation

Finally the specification defines MSR as:

- "Single RAT" mode where a radio transmits only one radio access technology (Rel-8)

- “Multiple radio access technologies” are transmitted on same radio amplifier (Rel-9)

In summary MSR enables operators to meet current demand with existing spectrum based on one technology with the option to reuse the same hardware for another technology in the future.

2.3 THE UICC/USIM AND ROAMING

Industry pundits hope that by the time LTE will be widely deployed worldwide, common radio bands will be agreed upon so that new multi-bands handsets will enable seamless roaming in a large number of countries. However, during a transition period, not all the handsets will support all the possible radio bands. Therefore, there will be a need to change handsets when roaming in some countries. The USIM will allow users to switch to a more appropriate handset when necessary, in order to get his home operator services.

The ability to define LTE roaming preferences has been introduced from USIM specification 3GPP 31.102 Rel-8. In order to optimize roaming charges, operators need to upgrade their UICC to a Rel-8 or beyond profile, and to properly configure the USIM. To offload network traffic efficiently, LTE network shall be set as higher priority in the operator controlled PLMN selector with Access Technology.

If a legacy UICC/USIM is inserted into the device, this will not prevent the user from accessing LTE network since backward compatibility is guaranteed. However, as the legacy UICC/USIM is not configured with LTE, by default access to the LTE network will occur with lowest priority.

2.4 COVERAGE TRIGGERED WCDMA/GERAN SESSION CONTINUITY

The Coverage Triggered WCDMA/GERAN Session Continuity feature uses the Event A2 (serving cell becomes worse than threshold) measurement process[3].

The UE measurements are reported to the serving RAN to make the final determination on redirection to the WCDMA network. Types of measurements used in the handover evaluation process include the following:

- Reference Signal Received Power (RSRP), representing the mean measured power per reference signal
- Reference Signal Received Quality (RSRQ), providing an indication of the reference signal quality

The UE indicates to the serving RAN when it has poor LTE coverage. The poor coverage release can be set to trigger on the RSRP value, RSRQ value, or both. The measurement reports sent by the UE to the serving RAN contain either one or both of these values.

The RAN determines whether to release the UE with a redirection to a WCDMA network, depending on the UE capabilities, and other parameters. If the UE is released with a redirection to a WCDMA network, the release message contains the UMTS Absolute Radio Frequency Channel Number (UARFCN), to help the UE find a suitable WCDMA cell.

[3] Further details on Coverage Triggered WCDMA and GERAN (GSM) Session Continuity can be found in: 3GPP TS 36.300 Overall description; Stage 2; 3GPP TS 36.304 User Equipment (UE) procedures in idle mode; 3GPP TS 36.331 Radio Resource Control (RRC); Protocol Specification

3 CORE NETWORK CONSIDERATIONS

This section deals with the phased migration from Rel-7 and earlier versions for legacy packet core interfaces for IRAT mobility with LTE, HSPA and GSM.

3.1 CORE NETWORK OVERVIEW

The EPC architecture as defined in Rel-8 has enhanced the core network by addressing session management issues along with a comprehensive policy framework. At the same time the definition has enabled it to be used as a unified core network for legacy network. This section will provide an overview of the EPC architecture, both for 3GPP operators that may evolve a Rel-7 and an earlier version of deployed network to LTE as well as for operators to interwork with other LTE networks in a roaming arrangement.

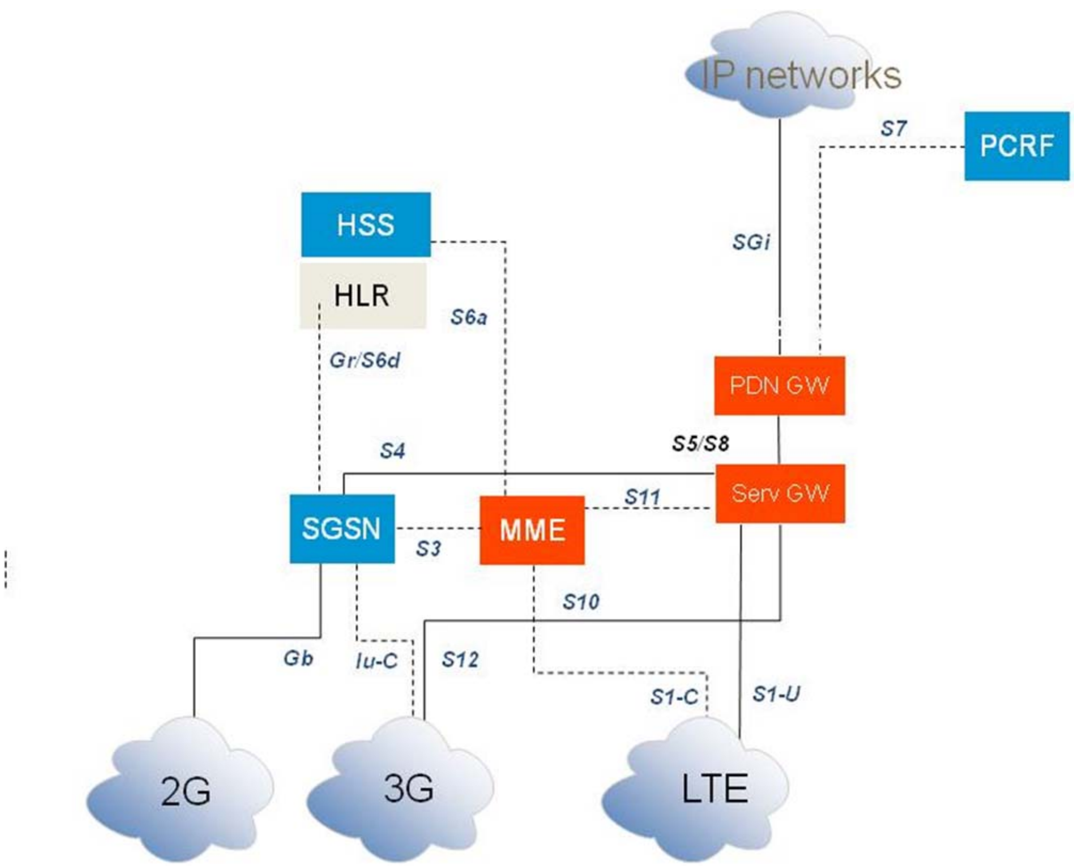

Figure 5: Evolved Packet Core

The main principles of the LTE-EPC architecture include:

- IP-based protocols on all interfaces between network elements
- Separation of the control and user plane. This entails an optimized architecture for the user plane including a reduction in the number of network elements that are traversed by data packets

- A common anchor point known as PDN-GW for various versions of 3GPP defined access technologies

- Ability to assign independent IP addresses and other attributes to multiple flows for a given user that can be assigned to different APN

- A comprehensive policy framework that interworks with the user and control plane network elements to provide appropriate QoS for flows

EPC architecture has defined the following network elements:

- The PDN gateway (P-GW) serves as an anchor point for LTE. In a network with multiple access technologies, it can also serve as a single common anchor point for all those technologies. Two different mobility protocols have been standardized, based on GTP and Proxy Mobile IP. The expectation is that GTP based protocol would be used where a legacy 3GPP network is being evolved to LTE and Proxy MIP would be used where a non-3GPP network has to interwork with LTE

- The Serving Gateway (S-GW) is the anchor point that connects the eNodeB to the core network

- The Mobility Management Entity (MME) handles control signaling for a given session. The MME functionality is kept separate from the gateways to facilitate network deployment, independent technology evolution, and fully flexible scaling of capacity

- Policy Charging and Rules Function (PCRF) completes the policy framework for LTE network. Even though the genesis of PCRF can be traced to IMS, its current usage has broadened to encompass policy based QoS for both IMS and non-IMS based applications running over the network

  S6a is the interface that is used between the MME and the HSS.

3.2 MIGRATORY ASPECTS

We can envision two different scenarios of LTE deployment. The first is where an operator already has a combination of a GSM, UMTS, and HSPA network deployed and would like to evolve it to an LTE network for seamless interworking across all access types. The second scenario is where LTE is deployed as a green field network. This section discusses the migration from Rel-7 architecture with Gn/Gp to S4/S12 (3GPP Rel-8) interfaces. In either case, the operator also has to consider enabling its users with roaming capabilities across other 3GPP operators' networks.

As with any technology evolution, the question is how to address the impact of new network elements, defined for LTE, on the legacy network. On one hand, the option of upgrading all RNC and SGSNs can be a costly proposition and on the other hand, seamless interworking between the two networks is an imperative.

In case the 2G/3G RNC and SGSN are not being upgraded, one could use GTP based Gn/Gp interfaces to communicate with the EPC elements. Gn interface is used between the 2G/3G SGSN and LTE defined MME and P-GW. Note that the use of Gn and Gp interface mandates that the GTP based mobility protocol is supported by the P-GW. Similarly, for roaming scenarios Gp interface is used when an SGSN is located in a vPLMN that communicates with a P-GW located in the hPLMN. This is addressed in the subsection, "Pre-Rel-8 Gn/Gp Mobility" below.

In the case that the 2G/3G SGSN has been upgraded to support LTE defined S3, S4, and S12 interfaces, one could use either of GTP or PMIP mobility protocols and use the P-GW as the common anchor for all the 3GPP access technologies. This is addressed in the subsection, "Rel-8 S3/S4 based EPS Mobility" below.

The following subsections address EPS mobility. They compare the Gn/Gp Architecture with the S3/S4 Architecture (Rel-7 and Rel-8 respectively).

PRE-REL-8 GN/GP MOBILITY

This mobility type supports mobility between LTE and already installed (legacy) 3GPP WCDMA/GSM GPRS networks using the existing interfaces and mobility mechanisms. In fact, the existing GPRS nodes will see the LTE/EPC nodes as other GPRS nodes. Hence, all adaptations for interworking are made in the LTE/EPC side alone, including transfer and mapping mechanisms.

The MME and the PDN-GW both support the Gn interface towards the pre-Rel-8 SGSN for IRAT mobility. Sessions from an LTE capable UE are always anchored on the PDN-GW. Inter-access session mobility is possible when the UE moves between GERAN/UTRAN and LTE coverage with the help of the Gn interface between the SGSN and the MME and the PDN GW. Various solutions exist for providing HLR and HSS functionality in the network with consistent subscription data across all access technologies.

The benefit is that no upgrades are required in the existing networks (besides the user management parts), and it is viable until the operator decides to upgrade to S3/S4 networks.

Figure 6 shows the LTE-WCDMA mobility. Also, the 3G Direct-Tunnel (between the RNC and the PDN-GW) may be used to offload the SGSN from payload, as illustrated by the Iu-U interface.

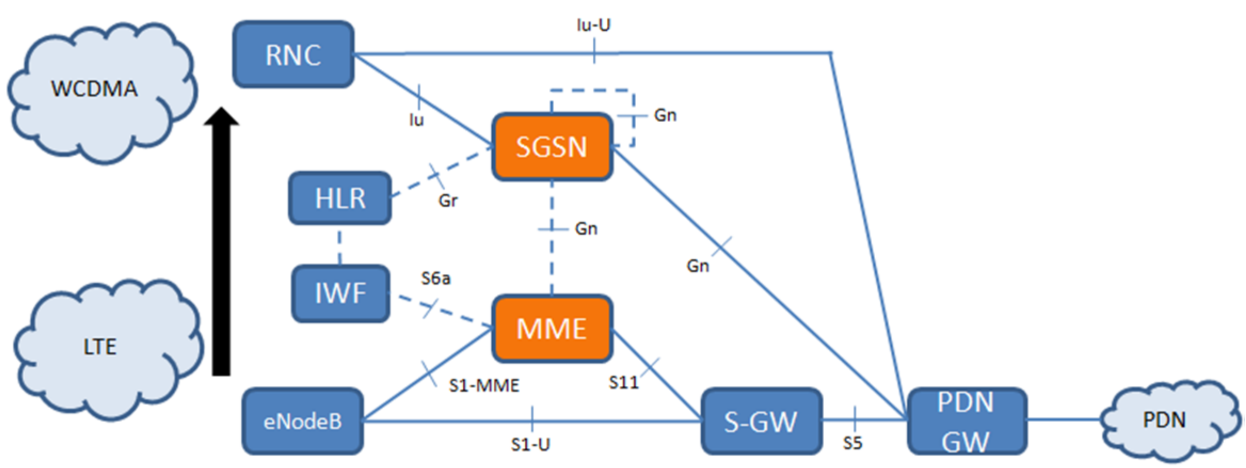

Figure 6: IWF for reuse of HLR by MME

An operator that has both EPC and pre-Rel-8 packet core elements in the network have the following restrictions imposed on session mobility and bearer setup:

- The Idle Mode Signaling Reduction (ISR) mechanism (which reduces the frequency of LTE tracking area and 2G/3G Routing Area Updates cannot be used between the MME and pre-Rel-8 SGSNs since the Gn/Gp SGSNs do not support ISR procedures

- 3GPP Rel-8 introduces the concept of dual-stack bearers, whereas pre-Rel-8 SGSNs do not understand the concept of dual stack PDP contexts. Thus, separate EPC bearers for IPv4 and IPv6 must be created such that both IP addresses can be preserved when the UE moves from LTE to 2G/3G coverage

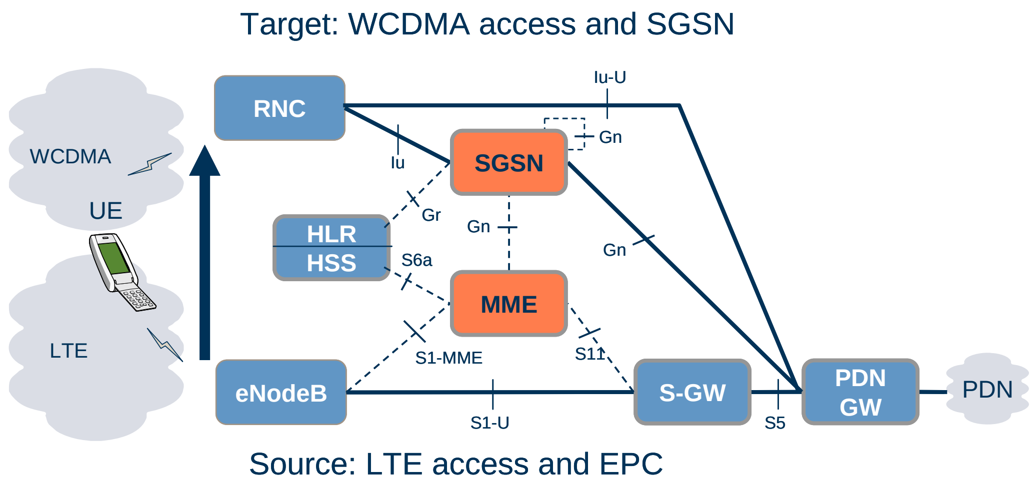

Figure 7: Pre-Rel-8 Gn/Gp Based Mobility between LTE and WCDMA/GSM

REL-8 S3/S4 BASED EPS MOBILITY

This mobility type supports mobility between LTE and upgraded WCDMA/GSM access networks, using the 3GPP EPC "S"-interfaces and related functions for interworking and mobility. Inter-access mobility is enabled by the S3 (SGSN-MME) and S4 (SGSN-SGW) interfaces.

The benefits of this solution (in comparison to the Gn type mobility) are, in brief:

- SGW is the common session anchor for roaming and non-roaming traffic for all 3GPP radio technologies which leads to a simplified network architecture

- Allows for usage of ISR mechanisms which target increased terminal battery life through decreased signaling while in idle mode

- Allows for usage of 3G Direct Tunnel also for roaming users, further offloading the SGSN payload plane

- Allows for mobility between 3GPP and non-3GPP accesses while retaining a common PDN GW and hence maintaining an active session without changing terminal IP address

Figure 8 shows the LTE-WCDMA mobility. Also the 3G Direct-Tunnel may be used to offload the SGSN payload, in the picture illustrated by the S12 interface.

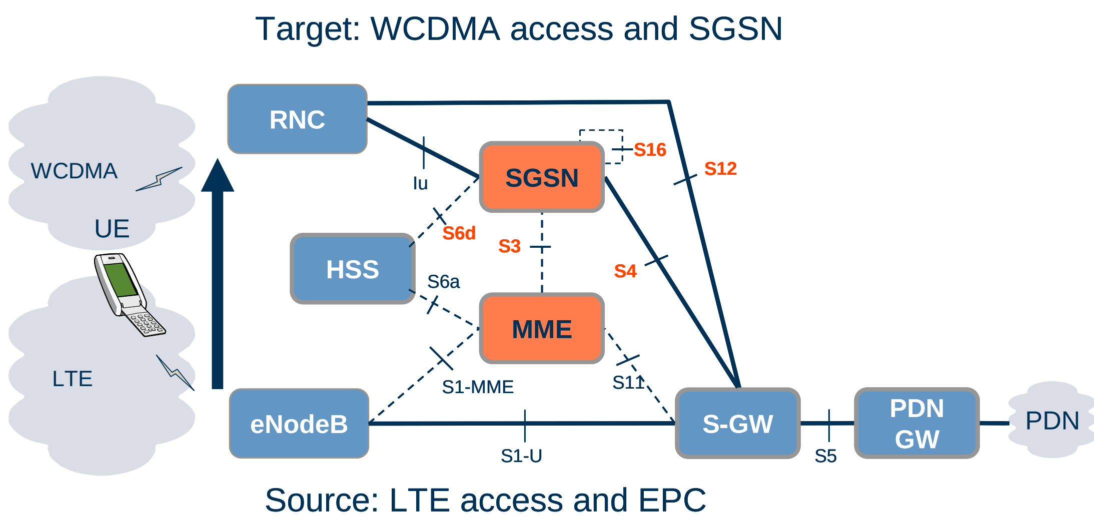

Figure 8: Rel-8 S3/S4 Based Mobility between LTE and WCDMA/GSM

3.3 SUBSCRIBER DATA ASPECTS: USIM/ISIM

Coexistence of LTE with WCDMA/HSPA and GSM Network requires a smooth transition of home location register HLR towards HSS based on 3GPP specifications. This also helps in providing next generation Data Layered Architecture that addresses operator's needs regarding centralization of subscriber data used for authentication across multiple layers of the same session.

The SIM/USIM is the security token in 2G/3G networks for authenticating a subscriber in the operator's HLR. The USIM will continue to provide user authentication function at the access level to the LTE network.

However, it is necessary to define a mechanism for service level authentication as well, for services such as the IMS. The ISIM (an application on the UICC) is defined to hold both the user's access level credentials and the IMS Private User Identity that is stored in the HSS. The ISIM enables the user to authenticate to the LTE operator's IMS network and access its services.

3.4 ROAMING SCENARIOS

There are two roaming scenarios based on home tunneling compared to home tunneling with the possibility of local breakout as shown in Figure 9. The 2G/3G networks refer to legacy CS or PC based accesses like GSM/WCDMA in the network.

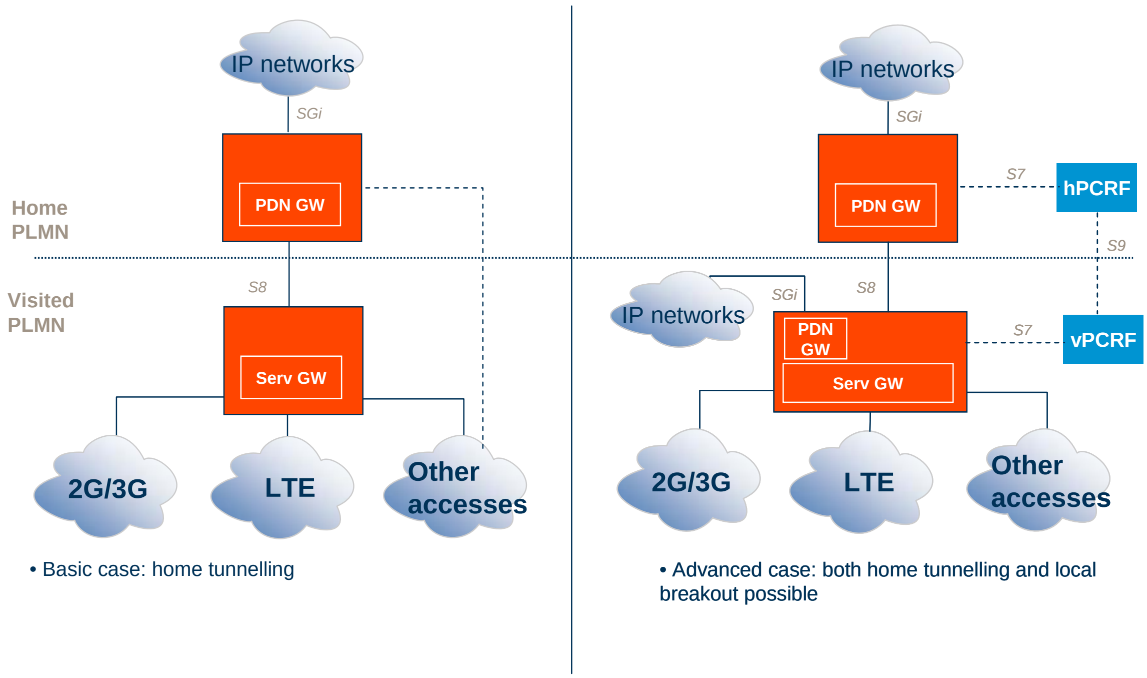

Figure 9: Home Tunneling (Routed) vs. Home Tunneling with possibility of local breakout

HOME TUNNELING OR HOME ROUTED SCENARIO

In this scenario, a subscriber that roams into a partner network, referred to here as Visited PLMN is still served by its Home PLMN. That is, the subscriber's IP address is assigned by the P-GW located in the Home PLMN. This enables all traffic to be routed to/from the Home PLMN. Similarly, the policies for setting QoS priority as well as other charging are directly set by the hPCRF.

HOME TUNNELING ALONG WITH POSSIBILITY OF LOCAL BREAKOUT

In this scenario, the idea is that some services such as VoIP that are delay sensitive need not be home routed. Similarly, besides delay sensitive flows, other flows that need not be seen by the home network could also be routed directly from the P-GW located in the Visited PLMN through the mechanism called the "local breakout." The idea is that depending on whether a flow needs to be home routed, it will be assigned an IP address from different P-GWs, one located in the Visited PLMN and another located in the Home PLMN.

Corresponding to the hPLMN and vPLMN the standard has also defined hPCRF and vPCRF respectively. This section will explain the S7 and S9 interfaces and details around vPCRF and hPCRF. S7 interface is used between the P-GW and PCRF in general. However, hPCRF communicates its policies for a given subscriber that will be assigned an IP address from the vPLMN for local breakout by using S9 interface between hPCRF and vPCRF.

4 QOS IN 3GPP

QoS is an important area for the successful growth of wireless networks. Applications have various requirements from the transport network in terms of delay, bandwidth and error rate that they desire for optimal performance or user experience. This poses a challenge for deployment of wireless networks.

However, it poses an even greater challenge when different access networks with varying capabilities have to coexist and a user tries to move seamlessly across these access networks with an expectation that there is no impact to the service or application one is using. LTE Rel-8 has made significant strides in defining QoS[4] capabilities and a policy framework to support those capabilities. It also allows multiple IP-CAN sessions similar to multiple PDP contexts in pre-Rel-8 systems. In both cases, IP addresses are bound to IP-CAN sessions or PDP contexts.

Devices in an LTE system are assigned a data bearer and attach to the system. This is unlike previous mobile broadband technologies, where data bearers were assigned at request from the device. Therefore, LTE is considered an “always-on” technology.

Each IP-CAN session can support multiple IP-CAN bearers which in turn can support multiple service data flows. Each IP-CAN bearer is considered to be an independent bearer with separate QoS Class Identifier (QCI) and other defining attributes. Broadly speaking there are two types of bearers that have been defined in the LTE Rel-8 network: Guaranteed Bit Rate and Non-Guaranteed Bit Rate.

Guaranteed Bit Rate (GBR) is where network resources equivalent to a certain bit rate are reserved at the time of the creation of the bearer. This type of bearer requires active management of its attributes through the lifetime of the bearer and hence a need to support dynamic management of policy rules that govern such bearers. Also, this type of bearer is typically used for conversational or streaming applications where certain bit rate has to be maintained for the duration of the session. It is important to note that even though GBR bearers guarantee network resources by reserving them well in advance, including bandwidth for successful delivery of packets it does not guarantee against deterioration of radio conditions due to the geographical environment. Treatment of a bearer in such conditions is determined by the policies in a specific deployment.

The second type of bearer is Non-Guaranteed Bit Rate (Non-GBR) bearer and is typically used for interactive applications such as IMS signaling, progressive video streaming, web browsing, chat, etc., or background applications such as FTP and email. For this type of a bearer there is no reservation for pre-defined bandwidth or bit rate. As a result for such bearers, congestion could lead to dropped packets or delay in delivery of packets which would be considered as an expected behavior.

One significant difference between Rel-8 networks and pre-Rel-8 networks is in the way bearer flows are established. In Rel-8, at the time of session initiation a default bearer is established per IP address assigned to a session which is always defined as a Non-GBR bearer. However, subsequently if an application or flow requires specific attributes for a flow, an additional dedicated bearer can be established that is either a GBR or Non-GBR bearer. A similar capability exists for pre-Rel-8 networks where a dedicated bearer can be seized after establishing a secondary PDP context.

One of the primary characteristics of a default bearer is that unless a bearer other than a default bearer can be identified for a given packet it will flow over the default bearer. This also implies that in case different QoS treatment is to be given to packets from two different applications, one or more dedicated bearers need to be established that can distinguish the two flows separately from these two applications. However, if for some reason the dedicated bearer is removed, all the packets identified with that bearer will from then onwards flow over the default bearer.

From an interoperability and coexistence perspective, this underscores the need for the legacy networks to support secondary PDP contexts otherwise during handovers from an LTE network to UMTS or GPRS

[4] QoS is defined in 3GPP specification TS 23.203 and TS 23.107

networks the dedicated bearers in an LTE network would be carried over the primary PDP context. These primary PDP contexts will not be able to distinguish separate classification that individual dedicated bearer packets would require. This may especially be an important issue to address since many legacy networks may not have enabled secondary PDP context in their networks (even though the standard may support it) which would be essential for the transfer of dedicated bearers to such networks.

4.1 QOS CLASSIFICATION AND DIFFERENCES IN 3GPP NETWORKS

In Rel-8 LTE network, QCI information is explicitly signaled across the network. In comparison, in the GPRS Rel-8 network, QCI is signaled as a vector of the pre-Rel-8 QoS parameters. As a result, this has some impact on coexistence of networks; especially for roaming users where except for a few QoS parameters such as Allocation Retention Priority (ARP), GBR and MBR most other parameters are proprietary and hence will be dependent on the network operator.

Similarly, operator policies decide (for lack of any standardized mapping) the mapping between the QCI defined by 3GPP and Diff Serve Code Point (DSCP) markings used in the IP transport network. Since in Rel-8 existing bearers can be modified, the Gateways should be appropriately able to modify corresponding DSCP markings in the downlink and RAN elements such as eNodeB should be able to modify corresponding uplink packets markings.

QOS CLASSIFICATION IN LTE REL-8

For each of the GBR and Non-GBR type of bearers, several QCI are defined to further characterize these bearers with attributes such as priority, packet delay budget, and packet error loss rate. In all, nine QoS classes have been defined in the Rel-8 network as shown in Table 1.

Table 1: Standardized QCI characteristics for EPC Rel-8 Network[5]

QCI	Resource Type	Priority	Packet Delay Budget (NOTE 1)	Packet Error Loss Rate (NOTE 2)	Example Services
1 (NOTE 3)	GBR	2	100 ms	10^{-2}	Conversational Voice
2 (NOTE 3)		4	150 ms	10^{-3}	Conversational Video (Live Streaming)
3 (NOTE 3)		3	50 ms	10^{-3}	Real Time Gaming
4 (NOTE 3)		5	300 ms	10^{-6}	Non-Conversational Video (Buffered Streaming)
5 (NOTE 3)	Non-GBR	1	100 ms	10^{-6}	IMS Signalling
6 (NOTE 4)		6	300 ms	10^{-6}	Video (Buffered Streaming) TCP-based (e.g., www, e-mail, chat, ftp, p2p file sharing, progressive video, etc.)
7 (NOTE 3)		7	100 ms	10^{-3}	Voice, Video (Live Streaming) Interactive Gaming
8 (NOTE 5)		8	300 ms	10^{-6}	Video (Buffered Streaming) TCP-based (e.g., www, e-mail, chat, ftp, p2p file sharing, progressive video, etc.)
9 (NOTE 6)		9			

NOTE 1: A delay of 20 ms for the delay between a PCEF and a radio base station should be subtracted from a given PDB to derive the packet delay budget that applies to the radio interface. This delay is the average between the case where the PCEF is located "close" to the radio base station (roughly 10 ms) and the case where the PCEF is located "far" from the radio base station, e.g. in case of roaming with home routed traffic (the one-way packet delay between Europe and the US west coast is roughly 50 ms). The average takes into account that roaming is a less typical scenario. It is expected that subtracting this average delay of 20 ms from a given PDB will lead to desired end-to-end performance in most typical cases. Also, note that the PDB defines an upper bound. Actual packet delays - in particular for GBR traffic - should typically be lower than the PDB specified for a QCI as long as the UE has sufficient radio channel quality.

NOTE 2: The rate of non congestion related packet losses that may occur between a radio base station and a PCEF should be regarded to be negligible. A PELR value specified for a standardized QCI therefore applies completely to the radio interface between a UE and radio base station.

NOTE 3: This QCI is typically associated with an operator controlled service, i.e., a service where the SDF aggregate's uplink / downlink packet filters are known at the point in time when the SDF aggregate is authorized. In case of E-UTRAN this is the point in time when a corresponding dedicated EPS bearer is established / modified.

NOTE 4: This QCI could be used for prioritization of specific services according to operator configuration.

NOTE 5: This QCI could be used for a dedicated "premium bearer" (e.g. associated with premium content) for any subscriber / subscriber group. Also in this case, the SDF aggregate's uplink / downlink packet filters are known at the point in time when the SDF aggregate is authorized. Alternatively, this QCI could be used for the default bearer of a UE/PDN for "premium subscribers".

NOTE 6: This QCI is typically used for the default bearer of a UE/PDN for non privileged subscribers. Note that AMBR can be used as a "tool" to provide subscriber differentiation between subscriber groups connected to the same PDN with the same QCI on the default bearer.

To provide traffic separation in the uplink, an operator-configurable mapping is offered between QCIs and Logical Channel Groups (LCGs), in addition to multiple bearers. LCGs are also referred to as radio bearer groups.

Service prioritization is enabled by mapping QCIs to logical channel priorities used by the User Equipment (UE) for uplink rate control. The following table maps the QoS class identifier (QCI) to different bearers.

From a coexistence standpoint, LTE Rel-8 QoS definitions are not the same as the corresponding pre-Rel-8 descriptions. For that reason some level of mapping between the corresponding QoS classes is required to be standardized for bearers that move between the two types of access networks. UMTS

[5] 3GPP 23.203, *Policy and charging control architecture*

defines four QoS classes: conversational, streaming, interactive best effort, and background best effort. They have been mapped to the nine QCI classes defined under the Rel-8 PCC architecture for GPRS.

QOS CLASSIFICATION IN UMTS AND GPRS NETWORKS

In UMTS, there are four basic traffic classes that had been defined[6]. They are: conversational, streaming, interactive, and background. To enable seamless mobility of bearers across UMTS and GPRS access networks, a one-to-one mapping was defined as shown in Table 2.

Table 2: Recommended mapping for GPRS QoS Class Identifier to/from UMTS QoS parameters

GPRS QoS Class Identifier value	UMTS QoS parameters			
	Traffic Class	**Traffic Handling Priority**	**Signalling Indication**	**Source Statistics Descriptor**
1	Conversational	n/a	n/a	Speech (NOTE)
2	Conversational	n/a	n/a	Unknown
3	Streaming	n/a	n/a	Speech (NOTE)
4	Streaming	n/a	n/a	Unknown
5	Interactive	1	Yes	n/a
6	Interactive	1	No	n/a
7	Interactive	2	No	n/a
8	Interactive	3	No	n/a
9	Background	n/a	n/a	n/a
NOTE: The operator's configuration should reserve QCI values that map to "speech" for service data flows consisting of speech (and the associated RTCP) only.				

4.2 NETWORK-INITIATED VS. TERMINAL-INITIATED QOS CONTROL PARADIGMS

There are two distinct paradigms to request established dedicated bearers in wireless networks. They are terminal-initiated and network-initiated QoS control paradigms. The earlier 3GPP networks only supported terminal-initiated QoS control. With the definition of Rel-7 GPRS and then later in Rel-8 EPC, network-initiated QoS control was introduced.

In a terminal-initiated QoS control paradigm, a UE is responsible for requesting for specific bearer attributes. This, in turn, implies that applications running on a UE be able to access radio channel capabilities as well as be able to request specific bearer attributes at the time of flow establishment. These in turn would have to be communicated to the radio access network (RAN). Typically, such QoS requests are performed using terminal vendor-specific application programming interface (API) calls and could vary for each terminal type.

[6] Quality of Service (QoS) concept and architecture in UMTS is described in 3GPP TS 23.107

On the other hand, a network-initiated QoS control paradigm relies on the QoS request being generated from the network side. This request could be generated by the PCEF located at the gateway or even from the policy infrastructure such as PCRF. As more and more smartphones are being introduced by the operators, this paradigm has become even more attractive. Traditionally, web browsing has mostly been operated over best effort bearers. However, with the advent of specialized applications running on smartphones, the network operator can initiate and provide specialized QoS to an individual bearer that meets the requirements of the application.

Network-initiated QoS control lessens the role of terminal for QoS and policy control. This is applicable both for operator controlled applications such as IMS based voice, streaming TV, etc., and other third party application and content providers (ACP). Some of the advantages of network-initiated QoS control are:

- An application provider can request specific QoS for a given bearer from the network which could be access agnostic and independent of the terminal type. Such requests would typically be from specialized applications that communicate with their corresponding network-based servers rather than for simple web browsing. This is turn will foster a richer set of applications due to ease in development over multiple platforms

- The above also helps where an application may run on a UE that is separate from the radio modem such as laptop etc.

- It enables more consistent exception-handling procedures that can be deployed by the network operator. For example, in case there is not enough bandwidth or other resources available at the time of request, the network can subsequently fulfill the request based on network policies without the application generating another request. The same can be accomplished for any modifications to the QoS of dedicated bearers based on network conditions as result of other bearers in the APN

As can be seen, both QoS control paradigms can be useful. However, as the set of applications running on today's wireless networks become more complex and have richer QoS requirements, network-initiated paradigm has distinct advantages over the other.

5 LTE SERVICES

This chapter gives an overview of the different standards and industry initiatives to provide telephony over LTE access. The recommendations are to use 3GPP standards when introducing VoLTE access. The reason for this is to avoid fragmentation on the terminal market as well as to optimize the integration efforts by introducing IMS from start in core instead of taking intermediate steps before IMS Voice is introduced.

5.1 SERVICES OVER LTE

Multiple services are envisioned for continuity of end-user experience as LTE is introduced to an existing 3GPP network. 3GPP has evaluated multiple solutions for voice over mobile broadband, including voice over HSPA+. The result has been a standard defining the minimum requirements for VoLTE, as defined in IR.92. Table 3 below describes 3GPP defined standards for voice and SMS over EPS as well as the transition solution in areas where LTE coverage is not available.

Table 3: 3GPP defined standards for voice and SMS over LTE

Service Capability	Prior to LTE introduction	Potential intermediary step at LTE introduction	On LTE Introduction
Basic Voice Service	Provided by CS Domain	Provided by CS domain after CS Fallback	Provided by IMS
Supplementary Services	Provided by CS Domain	Provided by CS domain after CS Fallback	IMS MMTEL; possibility for IMS MMTEL for 2G/3G access using IMS Centralized Services
SMS	Provided by CS Domain	SMS over SGs	SMS over IP or SMS over SGs
Emergency Calls	Provided by CS Domain	Provided by CS Domain	IMS Emergency Calls, or E911 provided by CS Domain using CS Fallback or by UE autonomously RAT switching
Service Continuity	Provided by CS Domain	Provided by CS Domain after CS Fallback or VoIP over PS access	PS Handover for mobility within LTE; SRVCC for mobility to 2G/3G access.

CSFB: This is a network and device based mechanism by which devices active on LTE are required to re-tune to 2G/3G. Therefore they perform CS (network) Fallback (CSFB) from LTE to access legacy CS based services like voice and SMS. LTE/EPC provides a mechanism to page the subscriber over the LTE access.

SRVCC: This is a network based mechanism by which a single radio voice call continuity (SRVCC) feature ensures voice call continuity between IMS over PS access and CS access for calls that are anchored in IMS when the UE is capable of transmitting/receiving on only one of those access networks at a given time.

To implement SRVCC in a 3GPP network[7], from E-UTRAN to UTRAN/GERAN, MME first receives the handover request from E-UTRAN with the indication that this is for SRVCC handling, and then triggers the SRVCC procedure with the MSC Server enhanced with SRVCC via the Sv reference point. There is no interworking function between the MME and the MSC.

Deployment options for Voice over LTE will depend on each operator's specific scenario. Operators have different starting points which will influence the deployment strategy, i.e., operators may start with:

- Voice support in LTE using CSFB[8] or
- IMS Voice support in LTE including support for SRVCC; CSFB support for inbound roamers

Also the frequencies that will be used for the LTE build out will heavily influence the time it takes to achieve mainly continuous or even full LTE coverage.

Figure 10 illustrates a scenario for overlaying of LTE over an existing 2G/3G network:

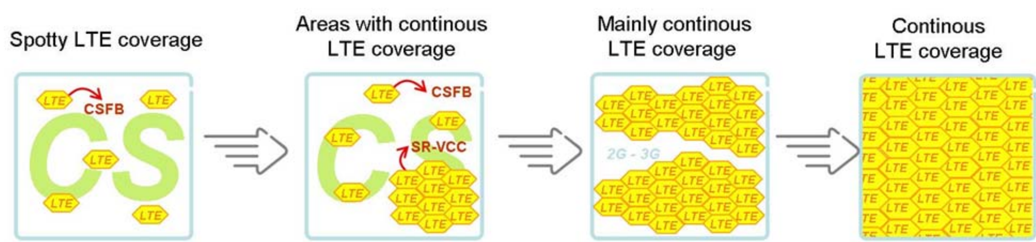

Figure 10: Illustrative Scenario of Voice over LTE migration over time

Initially LTE coverage is non-contiguous. For continuity of 2G/3G wireless services, it is required that all CS based services use CS Fallback. This requires that devices re-tune away from LTE towards 2G/3G and use legacy wireless and core networks for services.

In continuous LTE coverage, IMS based VoIP and Multimedia servers are used to replace legacy CS voice with VoIP and Multimedia. As LTE coverage increases and LTE voice is supported by IMS, and to provide contiguous service in some areas when UE moves out of LTE coverage, SRVCC feature in the network ensure continuity in voice between PS based LTE and CS based 2G/3G networks in the LTE contiguous areas. Outside the contiguous LTE coverage, CSFB is used to provide legacy services.

In case of mainly continuous LTE coverage, PS based handover (PS HO) and/or SRVCC is used to maintain VoIP and legacy services between LTE areas.

Some operators consider upgrading 3G networks as well supporting IMS Voice by deploying IMS Centralized Services (ICS). This provides communication services such that all services, and service control, are based on IMS mechanism and enablers, complementing LTE coverage, either when

[7] SRVCC: http://www.3gpp.org/ftp/Specs/archive/23_series/23.216/
[8] CSFB: http://www.3gpp.org/ftp/Specs/archive/23_series/23.272/

introducing IMS Voice on LTE or thereafter. In this case PS HO must be supported between LTE and 3G for seamless service experience.

5.2 CIRCUIT SWITCHED FALLBACK

CS Fallback is a concept for offering CS domain services together with LTE / E-UTRAN radio.

A CS Fallback capable UE, which is attached to E-UTRAN, may use GERAN or UTRAN to establish CS services. The function has been standardized as part of 3GPP Rel-8.

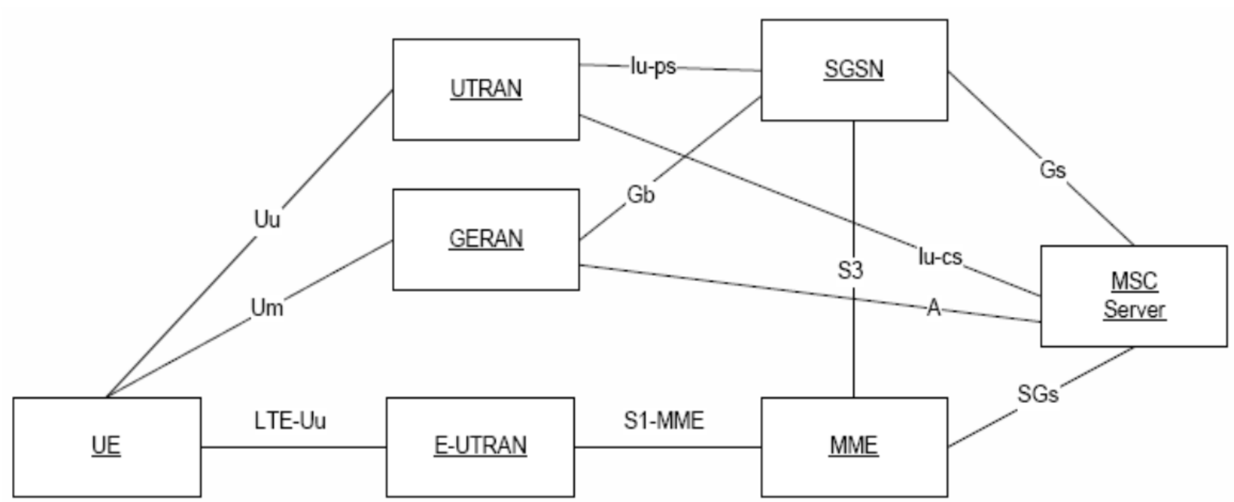

Figure 11: EPS architecture for CS fallback and SMS over SGs

Figure 11 shows the interfaces related to support for legacy voice and SMS over different accesses.

The need for CSFB arises in multiple scenarios which include:

- To provide voice services to LTE capable terminals before an operator has launched IMS Voice over LTE
- In initial stages, as a complement to IMS MMTel in order to support emergency calls and SMS when they are not yet supported in EPC/LTE network
- As an intermediate solution to retain current roaming relationship before IMS roaming agreements are settled, since that will require some time
- To steer (PS based) voice traffic from LTE to an overlay GSM or WCDMA network (with CS based architecture), when LTE spectrum is not adequate to support both VoIP and high data rates advertised in the market

Some operators also want to decouple the LTE rollout from IMS/MMTel and then CS Fallback is the only standardized alternative.

For this purpose an interface is introduced between the MSC and the MME, called SGs-interface and is based on the SCTP protocol. It provides the support for Mobility Management, Paging and SMS, as shown in Figure 12: Circuit Switched Fall Back (CSFB).

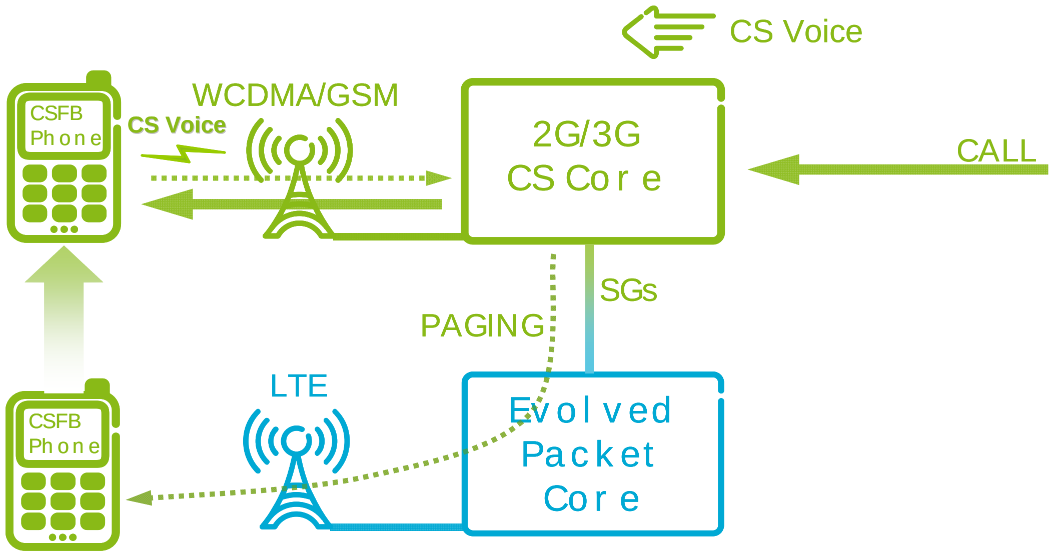

Figure 12: Circuit Switched Fall Back (CSFB)

A prerequisite for CS Fallback is that the UE is registered in the MSC while being attached to E-UTRAN and registered in the MME. This is achieved by using combined Mobility Management procedures for EPS and CS.

When originating a voice call and when receiving a page for CS voice the UE is moved to 2G/3G and the voice is sent over one of these access types. The page response is then sent from the new RAT supporting the CS. It can be done via IRAT PSHO or RRC Connection Release with Redirect from LTE to WCDMA (UTRAN) or GSM (GERAN) and via CCO (with or without NACC) to GSM (GERAN). The UE will return to LTE after call completion if LTE is preferred and coverage exists.

Additionally, 3GPP has specified a special SMS handling (SMSoSGs). SMS can be sent via the SGs-interface without doing a fallback to CS. UEs that are interested in SMS but not in other CS services (e.g., Laptop cards) have the possibility to attach with an SMS only option to E-UTRAN. Such a UE is able to use the special SMS handling without the need to support the fallback to CS.

There are three main CSFB procedures:

1. Release with Redirect (without or with automatic SI exchange)
2. Packet Switched HandOver (PSHO)
3. Cell Change Order (CCO)

Depending on the CSFB procedure different network elements are affected. The CSFB function is only possible to realize in areas where LTE coverage is overlapped with GSM, WCDMA or coverage. CSFB allows retaining the current CS roaming relationships between operators, since CS voice is still used.

RRC RELEASE WITH REDIRECT

The basic CSFB option is RRC (Radio Resource Connection) Release with Redirect defined in Rel-8. The impacted nodes are indicated by the CSFB in the circle in Figure 13.

With RRC Connection Release with Redirect, it will be possible to introduce CSFB without any major updates to current GSM (GERAN) and WCDMA (UTRAN) system. The UE will return to LTE (E-UTRAN) after call completion.

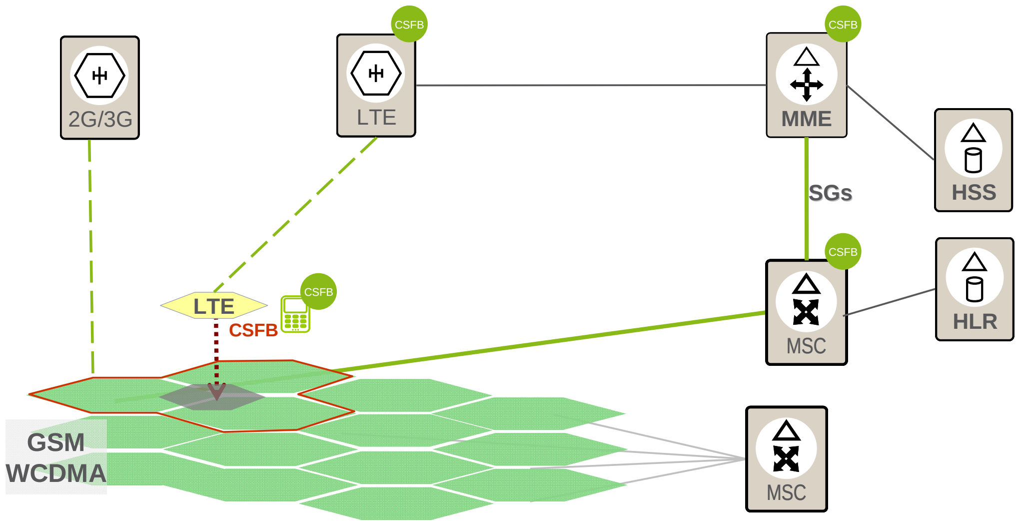

Figure 13: Release with redirect

Impacted nodes are UE, eNB, MME and MSC. No impact on GSM or WCDMA. The main characteristics of this procedure are:

- Slower call set-up time because broadcast SI (System Information) needs to be read by the UE in the target RAN
- Long PS outage time for WCDMA or DTM (Dual Transfer Mode) GSM

ENHANCED RELEASE WITH REDIRECT, AUTOMATIC SYSTEM INFO EXCHANGE WITH RIM

The enhanced option in Rel-9 may use the standardized RIM (RAN Information Management) procedures[9] to transfer System Information (SI) from UTRAN/GSM and WCDMA to LTE[10]. This requires additional impact on the total network. 3GPP impacts the nodes indicated by RIM in the circle in Figure 14. It is recommended that RIM is used for GSM, and deferred measurement[11] is used in WCDMA.

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[9] Currently, there are procedures defined on the Gb and Gn interfaces to enable signaling of GERAN SI/PSI ([Packet] System Information) between BSSs. This *RAN Information Management (RIM)* mechanism was defined initially for the use of NACC, although in a manner that could be extended for applications other than NACC. It consists of the following messages:
- RAN INFORMATION REQUEST - from Source BSS to Target BSS – requests GERAN SI/PSI.
- RAN INFORMATION – from target BSS to source BSS – analogous to the Information Exchange over Iur and includes GERAN SI/PSI for one or more GERAN cells.
- RAN INFORMATION ACKNOWLEDGE – from Source BSS to Target BSS.
- RAN INFORMATION ERROR - to inform about e.g. message syntax errors.

[10] http://www.3gpp.org/ftp/Specs/archive/36_series/36.410/

[11] http://www.3gpp.org/ftp/specs/archive/25_series/25.331/ . When active, the UE can transmit RRC messages on RACH and receive RRC messages commanding it to enter CELL_DCH with reading a subset of overhead messages

RIM needs to be rolled-out in the source and target RAN as well as for SGSN and MME. SGSN and MME will be needed to route the RIM container to the correct target node.

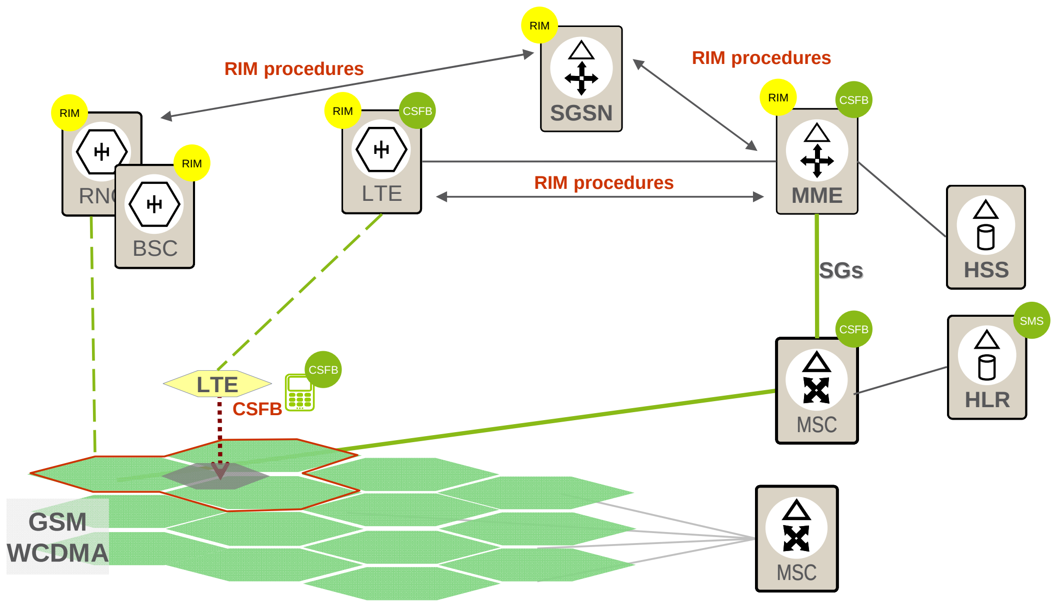

Figure 14: Enhanced Release with redirect using RIM procedures

Additional Nodes impacted include RIM support in RNC, BSC, eNB, SGSN and MME. In addition a network Impact is that RIM has to be rolled out in RAN and PCN. The benefit is faster call set-up times compared to the non-enhanced solution.

PS HANDOVER BETWEEN LTE AND GERAN FOR CSFB

Another alternative option to the RRC release is CSFB[12] based on PSHO[13]. The impacted nodes are indicated by the PS-HO in the circle in Figure 15. When the device originates a call or receives a page for a terminating call, it checks if the underlying network is PSHO capable.

If the underlying network is PSHO capable, the UE requests a PS Handover (with indication of CSFB) to the underlying UTRAN or GERAN network. On completion of the PSHO, the UE requests the network to suspend the newly acquired data bearer. The UE then sets up a CS bearer to handle an originating voice call, or respond to the MSC with a CS Paging response. The MSC continues with CS call setup over a CS bearer. If the underlying network is not PSHO capable, procedures using Cell Change Order are executed.

When PSHO is supported, the PS bearer is prepared first in the target cell before CSFB takes place. The benefit is a shorter service interruption time.

[12] Sec. 6 and Sec. 7 of http://www.3gpp.org/ftp/Specs/archive/23_series/23.272/
[13] Description of PSHO: http://www.3gpp.org/ftp/Specs/archive/23_series/23.401/

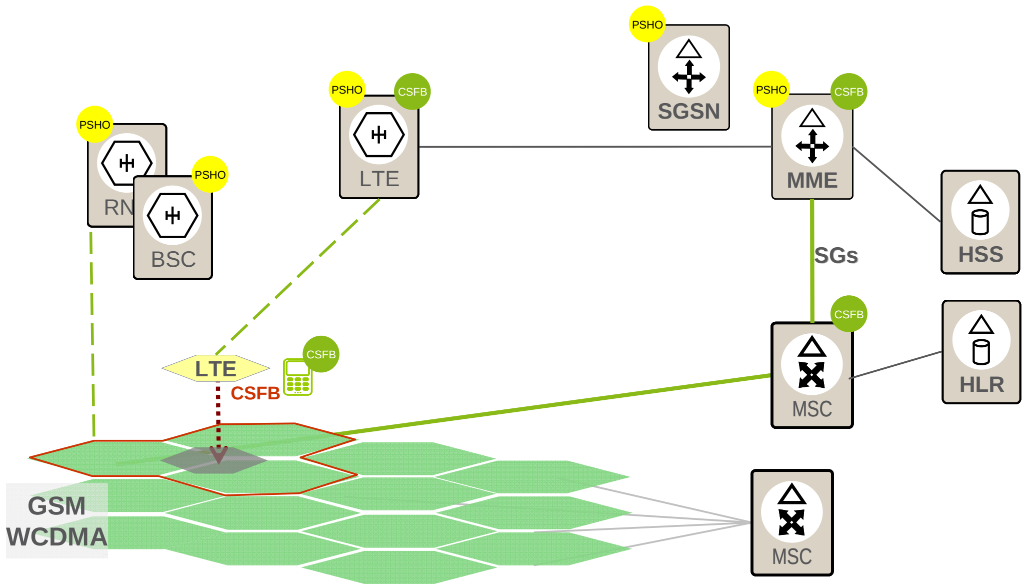

Figure 15: PS Handover between LTE and GERAN for CSFB

Additional node impact is the support of PSHO with CSFB from LTE to BSC. The benefit includes shorter PS outage time during CSFB IRAT handover.

CELL CHANGE ORDER

At reception of the request from the MME to page the UE, if the UE and network support inter-RAT cell change order to GERAN and the target cell is GERAN, and PSHO is not supported to GERAN:

- The eNodeB can trigger an inter RAT cell change order (optionally with NACC) to a GERAN neighbor cell by sending an RRC message to the UE.
- The inter-RAT cell change order may contain a CS Fallback Indicator which indicates to UE that the cell change order is triggered due to a CS fallback request.
- The UE moves to the new cell in GERAN. The UE uses the NACC information and/or receives the broadcast System Information and when it has the necessary information to access the GERAN cell, establishes a radio signaling connection.

5.3 IMS VOICE

This section discusses the IMS Multimedia Telephony specified by 3GPP in order to provide a flexible and enriched service. It includes traditional supplementary services that are enhanced as well as services beyond voice like messaging, video, file sharing, etc. The supplementary services are specified in 3GPP TS 22.173.

INTRODUCTION TO IMS

The IP Multimedia Subsystem (IMS) is an architectural framework for delivering Internet Protocol (IP) multimedia services. To ease the integration with the Internet, IMS uses IETF protocols wherever possible, e.g. Session Initiation Protocol (SIP). SIP is a text based protocol that is extensible with parameters that describe the type of multimedia session being established.

According to 3GPP, IMS is not intended to standardize applications but rather to aid the access of multimedia and voice applications from wireless and wireline terminals, i.e. create a form of fixed-mobile convergence (FMC). This is done by having a horizontal control layer that isolates the access network from the service layer. From a logical architecture perspective, services need not have their own control functions, as the control layer is a common horizontal layer.

The CSCF (Call Session Control Function) handles SIP messaging originated at the device in the (wireless) access and maintains session state for VoIP and Multimedia sessions.

The HSS (Home Subscriber Server shared with the EPC network in most implementations) handles subscriptions and authentication of IMS (and EPC) subscribers

The SIP-AS (Session Initiated Protocol-Application Server) component represents any of the SIP or IP based application servers that are triggered by the CSCF based on the service profile downloaded to the CSCF.

The MRF function addresses setup of the media flows between end points, based on any specific codecs or transcoding specified by the CSCF.

The PCRF is used for enforcing Policy/QoS/Charging when VoIP is implemented using the One Voice specification. The PCRF acts as a bridge between the IMS control layer (CSCF) and the policy enforcement layer in the wireless PS access. Figure 16 illustrates an IMS solution in a simplified manner. The lines in the figure represent the flow of SIP messaging between the components of the IMS network.

One Voice specifies the use of an IMS core with a voice application server to provide VoLTE. The PCRF and HSS entities in the EPC architecture are reused within the IMS architecture for providing policy control and subscriber management respectively. This makes the IMS core very relevant to the transition of GSM, HSPA voice services to LTE.

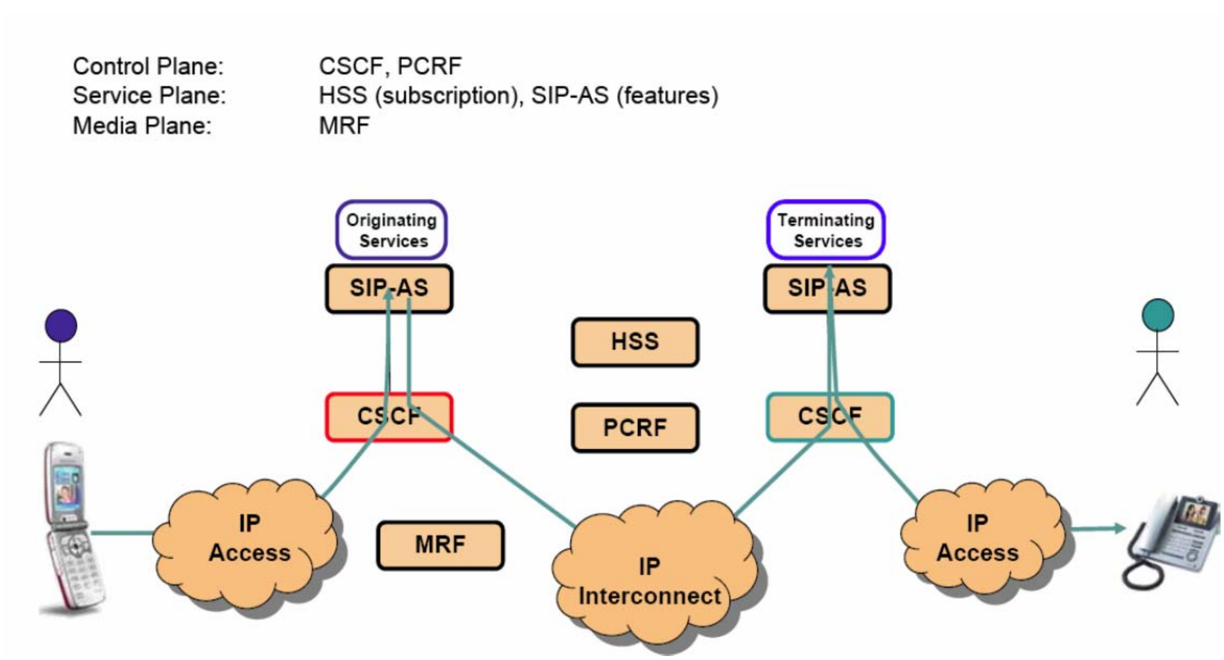

Figure 16: Simplified illustration of IMS

INTRODUCTION TO MMTEL

3GPP Multimedia Telephony service (MMTel) is specified to support a converged telephony offering that allows the operators to offer the multimedia telephony service over many different access types (Note only 3GPP mobile access shown related to the scope of this document).

IMS MMTel users can communicate with other MMTel users across operator boundaries to the full extent of the multimedia telephony experience, since MMTel service interoperability is fully supported via standardized NNI and UNI.

Interconnection with other proprietary VoIP systems (SIP or H.323) and PSTN/PLMN, also allows an MMTel user to communicate with end-users on legacy telephony systems.

IR.92[14] specification by GSMA defines the IMS profile for Voice and SMS over LTE, which identifies a minimum mandatory set of features, which are defined in 3GPP specifications to allow a good interoperability between a wireless device (the User Equipment [UE]) and network.

MMTEL BASIC SERVICE

MMTel offers the following set of end-user services:

- Real-time End-User Services
  - Voice and Video call
  - Video share: Video share is an end-user service which realizes a voice and video communication method between two peers. The video communication is simplex and is usually not time synchronized with the voice stream
- Non real-time End-User Services:
  - Image/Video clip/Audio clip share: Image share, Video clip share and Audio clip share are end-user services that are special cases of File transfer
  - File transfer: File transfer is an end-user service which gives the possibility to transfer one file from one end-user to another. The file can be of any sort
  - Chat: Chat is an end-user service which realizes a communication method in which text or multimedia messages are sent within a communication session between two or more peers

MMTel provides the end-user with an enriched real time communication experience based on several media components. An end-user can combine different media such as audio, video and real time text. By this an end-user are able to change media type during a call such as toggle between a video and a voice call. MMTel also supports sharing of pictures, movie clips and audio clips between two users.

Two or more users can thus communicate in real time using different media components including:

- Real-time voice transfer (full duplex)

[14] *IR.92.3.0 - IMS Profile for Voice and SMS*, http://gsmworld.com/documents/IR9230.pdf

- Voice-synchronized real-time video transfer (simplex and full duplex)
- Text chat (MSRP[15] based)
- File transfer (MSRP based)
- Video-clip, picture, and audio-clip sharing (MSRP based)

MMTEL SUPPLEMENTARY SERVICES

MMTel supplementary services apply to the entire communication session, regardless of active media components such as voice, video or text. One addition to note is the conferencing service, which enables users to add and remove call participants to create ad hoc multiparty calls. MMTel follows and evolves according to the 3GPP/TISPAN specification of supplementary services.

One other important part of MMTel is the standardized NNI, which enables operators to interconnect with one another. This way, a user who belongs to one operator can communicate with a user who belongs to a different operator without any preparations; the network will make sure that agreed and allowed media sessions are established. The standard is backwards compatible, which means MMTel services can interwork with current fixed and mobile telephony standards.

MMTel supports a large set of supplementary services compliant to 3GPP and TISPAN, the basic services are:

- Basic Voice Communication
- Video Communication
- Text Chat
- File Sharing
- Add/Drop Media
- Service for Unregistered Users

The sharing service involves the creation of an MSRP (Message Session Relay Protocol) session between two UEs. An MSRP session is negotiated in the same way as RTP based sessions using SIP and SDP signaling. The main difference is that MSRP requires the establishment of a TCP connection between the UEs.

Once the MSRP session has been established it can be used to send (share) different types of media e.g. pictures, movie clips, audio clips, etc. It can also be used to send text messages on a line-by-line basis.

For sharing, the MSRP session will typically be set up in parallel with an audio session. Supported media types are image, voice, video and text.

[15] An MSRP (Message Session Relay Protocol) session is negotiated in the same way as RTP based sessions using SIP and SDP signaling. It is used for real time transfer of text between end points on the user plane.

MTAS supports adding and removing of additional media flow into an existing session between two MMTel end-users, for example the addition of real-time text to an existing audio session. Another example is when a two-party audio/video call is modified to remove the video part.

5.4 LTE TO CIRCUIT SWITCHED VOICE CALL CONTINUITY

SRVCC is defined in 3GPP TS 23.216. SRVCC allows IMS session continuity (specified in 23.237) when the UE has a single radio, thus only one RAT can be active at one time.

When moving out from IMS Voice capable LTE coverage, SR VCC allows voice continuity via handover to 2G/3G CS. It is considered an important business advantage for operators since it allows a superior VoIP service that cannot be matched by third party voice application providers until LTE coverage is perfected.

SRVCC architecture includes a centralized IMS HPLMN based model in 3GPP Rel-8/Rel-9 and an IMS HPLMN/VPLMN based distributed model in 3GPP Rel-10.

A Rel-10 IMS HPLMN/VPLMN model can minimize the voice interruption during the handover to CS if the Home IMS is located far away from the serving LTE network.

3GPP REL-9 ARCHITECTURE ENHANCEMENTS

The following is the 3GPP Rel-9 SRVCC Architecture. The solid and dashed lines in Figure 17 below represent user data and control signaling respectively.

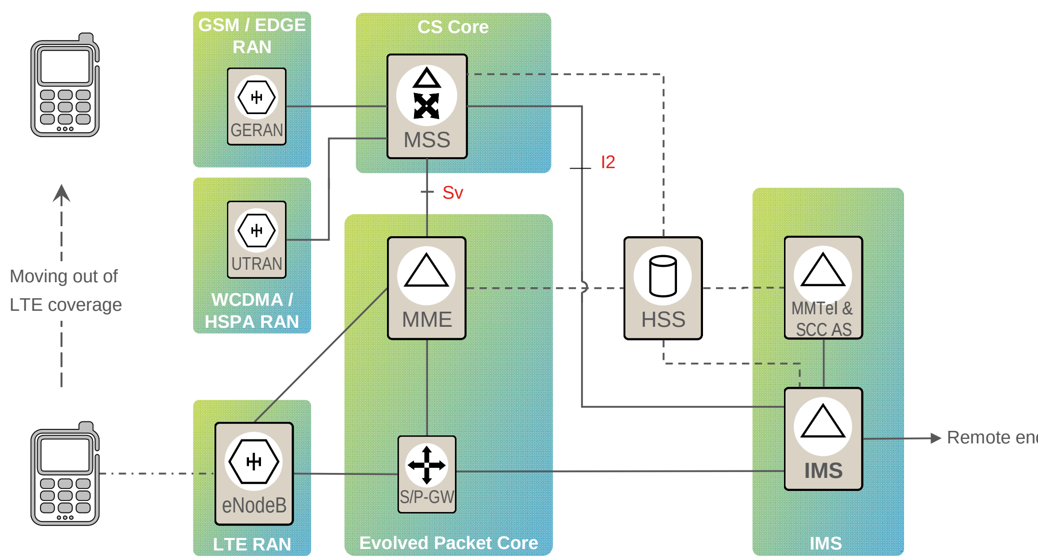

Figure 17: 3GPP Rel-9 SRVCC Architecture

The Rel-9 SRVCC solution impacts the eNodeB, the MME, HSS and the MSC Server and requires Service Continuity support in the IMS (SCC AS).

Important Interfaces:

- A new interface, Sv, has been specified between the MME and the MSC to execute the actual handover

- I2 interface is a new reference point between MSC enhanced for ICS/SRVCC and IMS. I2 in SRVCC context is used for access transfer signaling as well as additional call state transfers e.g. held calls, conf and alerting call state

- Note that a solution based on SRVCC with ISUP (Mg) instead of I2 is also possible, but this will not support transfer of mid call state. This is particularly troublesome for calls in alerting state (in Rel-10), since alerting state can be quite significant portion of the total call.

SCC AS is an IMS Application Server acting as a B2BUA on ISC (and Ma), in the home network.

Messaging is not within the scope of this discussion, as stated in Appendix A2 of the IR.92 (One Voice) specification.

3GPP REL-10 ARCHITECTURE ENHANCEMENTS

The following is the 3GPP Rel-10 SRVCC Architecture:

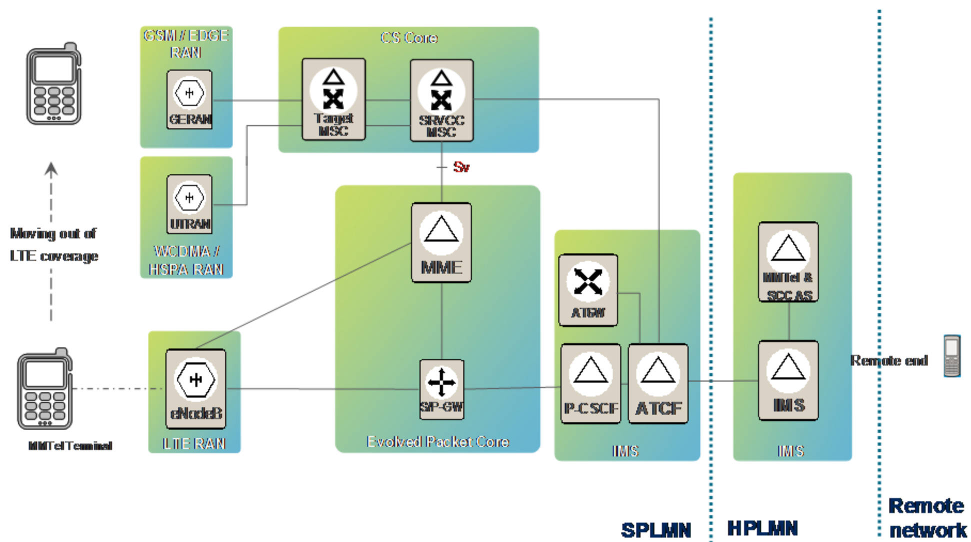

Figure 18: 3GPP R10 SRVCC Architecture to reduce latency at edge

The main intention of the enhancements of SRVCC in Rel-10 has been to lower the voice interruption delay during the access transfer.

The reason for the possible voice interruption is related to two parallel procedures: The radio access handover (HO) and the remote update call leg with new SDP. When the HO procedure is executing, the remote update of both call legs uses an SDP offer/answer exchange towards the remote party. This ensures that the remote party voice RTP stream is changed toward the MGW on the originating leg. As a result, the transferring party HO procedure from LTE to CS is completed.

Considering that one or both parties involved in a call subject for access transfer might also be roaming, this might result in a voice interruption due to long round-trip time of the remote party update.

The enhancement in Rel-10 includes improvements to the procedure to address this issue by moving the access transfer function out of the home IMS into the serving network (visited if not home).

The ATCF (Access Transfer Control Function) acts as SIP signaling anchor and sits on the SIP session path. The ATCF also controls a media plane function (ATGW – Access Transfer Gateway).

Messaging is not within the scope of this discussion, as stated in Appendix A2 of the IR.92 (One Voice) specification.

5.5 ROAMING CONSIDERATIONS

This section discusses the voice support mechanisms for roamers. An LTE network operator that has adequate LTE coverage to provide its home subscriber to fully utilize IMS voice with mobility may still need to continue the capability to provide CSFB for their inbound roamer. This is because the HPLMN of the inbound roamer may not yet support IMS voice and requires CSFB to provide voice services. 3GPP and GSMA work on IMS services roaming architecture and charging etc. is in progress.

Similarly, the LTE UE using IMS Voice at HPLMN may still be required to have CSFB capabilities when roaming out to other VPLMN. This is because the VPLMN may not have adequate LTE coverage and does not support SRVCC.

5.6 IMS CENTRALIZED SERVICES (ICS)

In the migration towards EUTRAN and EPC, ICS enables that services are centralized in the IMS, not only when the UE is on a PS access but also when the UE is on a CS access. This will ensure a consistent user experience (independent of access type). Service centralization ensures that operators can do new service development on IMS and the possibility to do more advanced services than possible in today's CS mobile telephony. All services defined within GSMA IR.92 are supported.

Two approaches exist: a Network-based approach and a terminal based approach. Both approaches exist in 3GPP in two variants:

1. ICS enhanced MSC server. Here a new interface, the I2 interface, between MSC and IMS is introduced. The MSC-S will act as an IMS UA towards IMS on behalf of the UE. i.e., The MSC-S will register the user. The MSC-S will act as an ISC MSC-S if ISC indication is sent from HLR. This is the only solution that supports service centralization before, during and after SRVCC on all required accesses (LTE, 3G, 2G) is the (I2) network-based solution.

2. No MSC ICS enhancement, but routing of both originating and terminating calls will be through IMS. For originating calls this is done by CAMEL triggers.

The UE-based approach's two variants:

1. Using Gm reference point (normal IMS signaling) for call control, but media being on CS. This will require simultaneous use of PS and CS domain (UTRAN or GSM DTM).

2. Using a new interface called I1 for call control, but media being on CS. This is an interface between UE and IMS, but the transport is not IP, it is assumed to be USSD.

Note: I2 solution still enables the use of Gm for other services e.g., for RCS while camping on 3G and 2G. If concurrent use of these services with Telephony is wanted UMTS or GSM DTM is required.

ARCHITECTURE FOR ICS

The following is the Architecture overview of ICS implemented with an MSC Server enhanced for ICS:

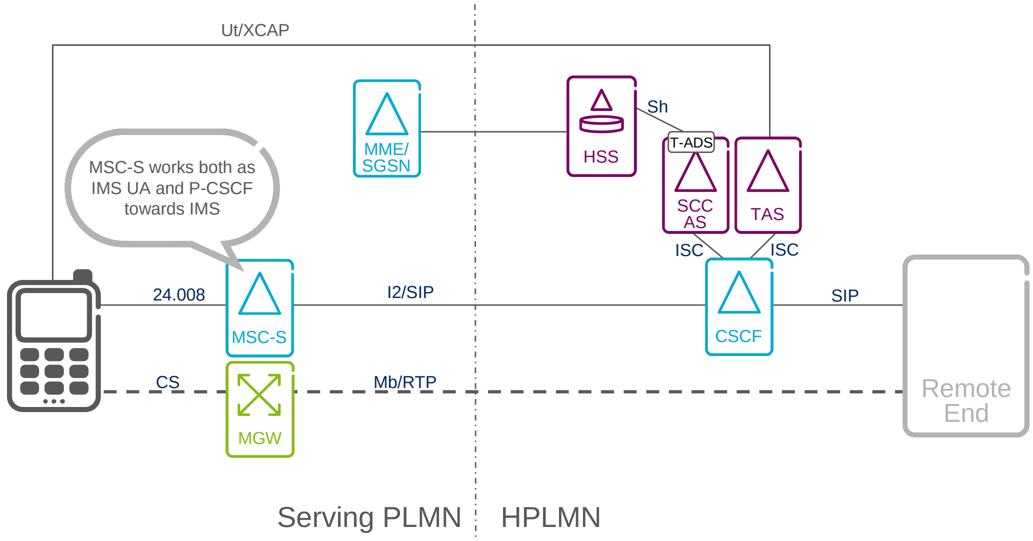

Figure 19: IMS Centralized Services (ICS) Architecture

The UE is not required to have any specific ICS support; hence a legacy UE may be used. As stated above, the MSC Server acts as gateway and acts as a SIP UA on behalf of the UE. A new interface between the MSC server and IMS called I2 is introduced for this purpose.

5.7 MESSAGING OVER LTE

SMS OVER SGS

This section presents SMS over SGs by using the SMS solution specified in CSFB.

SMS over SGs is specified in 3GPP TS 23.272[16], Circuit Switched Fallback in Evolved Packet System.

[16] *Circuit Switched (CS) fallback in Evolved Packet System (EPS); Stage 2*, 3GPP, (http://www.3gpp.org/ftp/Specs/html-info/23272.htm)

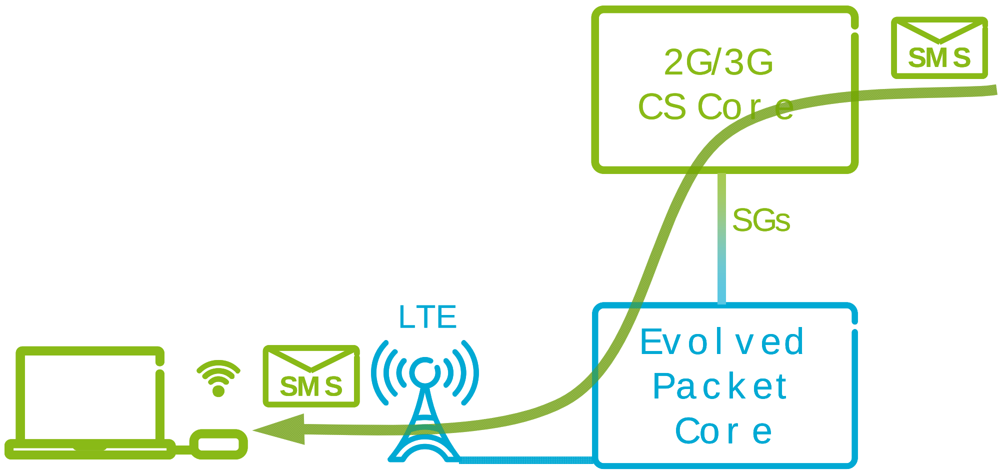

Figure 20: SMS over SGs

The high level procedure for a mobile terminating SMS is summarized as follows:

- The UE registers with combined EPS/IMSI attach for “SMS-only,” and updates MSC and MME using combined TA/LA update procedure
- When an incoming SMS arrives to the MSC, the MSC will send a paging via SGs interface to the MME, and the MME will tunnel the paging message to the UE using NAS transport
- The UE will answer the page and the MSC will send the SMS via SGs interface towards the MME, which will tunnel the short message to the UE using NAS transport

The SMS is transferred via SGs to MME and carried in NAS signaling to the device while the terminal is in LTE, avoiding the need to execute fallback to WCDMA/GSM. Existing CS roaming agreements are reused.

IMS MESSAGING

IMS Messaging or SMS over IP as defined in 3GPP TS 24.341[17]. When IMS Voice over LTE is introduced this will be the standard for SMS over LTE.

In IMS Messaging the transport to the phone is using SIP via IMS and an IP SM GW. It is applicable for LTE devices and networks after introducing VoLTE (IMS & MMTel) and for broadband access. It requires IMS and IMS roaming.

[17] *Support of SMS over IP networks; Stage 3*, 3GPP, (http://www.3gpp.org/ftp/Specs/html-info/24341.htm)

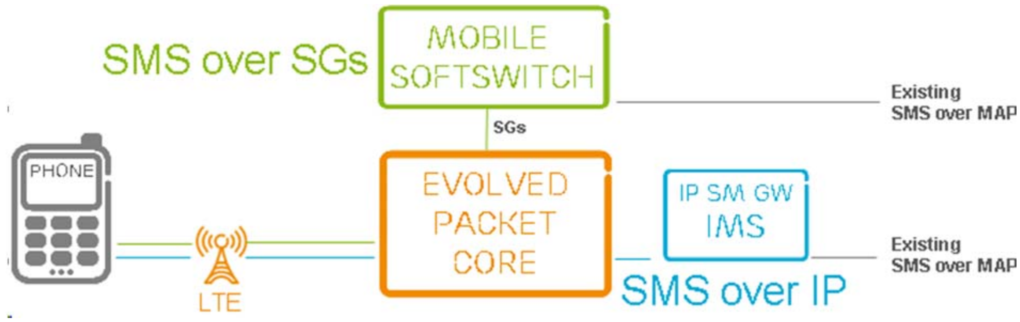

Figure 21: IMS Messaging or SMS over IP

When a device registers as an IMS user, an incoming SMS will be directed to IMS and delivered via IP. When the user has registered on IMS from multiple devices, the SMS is sent to all of them.

Messaging can be store-and-forward like SMS, MMS in legacy 2G/3G networks in page mode, or session based using MSRP sessions. In session based messaging, the MSRP protocol is used over a TCP connection to carry characters between endpoints in near real-time messaging.

In cases where the SMS cannot be delivered over IP (due to CSFB or SRVCC into a CS only access), the SMS may fail to reach the UE via IMS. To solve this, Terminating Access Domain Selection (T-ADS) capability described in TS 23.204[18] can be used. This capability allows to retry in case the termination via PS has failed and to select CS access instead. This T-ADS capability can also be used to ensure SMS termination via CS access, in case no IMS registration has been performed. This can also be used to give CS access priority over PS access.

If it is a configuration SMS that is received in all user terminals, then the user must know in which terminals it is applicable. For these configuration cases, OMA-DM (Open Mobile Alliance (OMA) Device Management (DM)) is preferred over the use of a configuration SMS, since in OMA-DM, the session identifies the terminal type, and the terminals not affected can disregard the information.

In the case of configuration SMS, if the device is attached to a PC, then it is most probable that the IP stack resides in the PC. When a configuration SMS arrives at the PC, the PC client would need to forward the identified configuration request to the dongle/module to be handled as if it was received via CSFB.

IMS also enables deployment of centralized, smart messaging servers over PS accesses like LTE. This includes consolidation of messaging, network based address books common to all applications, capability for presence and device capability governing message delivery etc.

[18] *Support of Short Message Service (SMS) over generic 3GPP Internet Protocol (IP) access; Stage 2*, 3GPP, (http://www.3gpp.org/ftp/Specs/html-info/23204.htm)

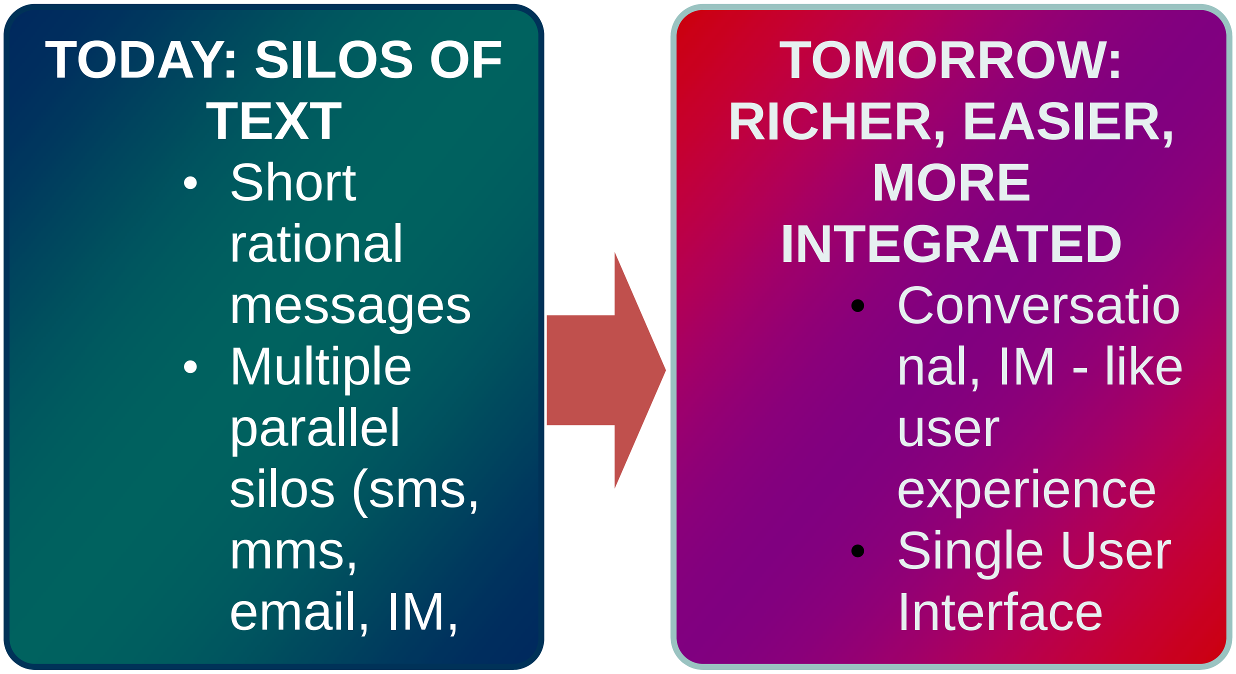

Figure 22: Evolution of Messaging

UICC/USIM AND IMS MESSAGING

It is recognized that SMS based messaging is a very popular service in GSM and HSPA/WCDMA. 3GPP decided that users will be offered the same type of services on LTE. Thus, in addition to SMS-C MSISDN address, the USIM (or IMS enabled "ISIM") will hold Public Service Identity (PSI) address of the SC for the user to continue to enjoy the SMS over IP services on LTE networks.

The PSI stored on the ISIM can be used by the IMS network to invoke the appropriate messaging system (SMS, MMS, voicemail, email, social networking, etc.), based on added context like device capabilities and access capabilities.

6 REGULATORY ISSUES

This section of the white paper will discuss the following regulatory issues:

- Lawful Intercept
- Emergency Services
- Telecommunications Device for the Deaf (TDD)
- Non-Voice Emergency Services (NOVES)
- Priority Services
- Commercial Mobile Alert Service (CMAS)

6.1 LAWFUL INTERCEPT

The purpose of Lawful Interception (LI) is to obtain communications network data pursuant to lawful authority for the purpose of analysis or evidence.

The Communications Assistance for Law Enforcement Act (CALEA)[19] is a United States wiretapping law passed in 1994. The purpose of CALEA is to help law enforcement and the FBI more effectively carry out wiretap operations, especially in view of the emerging digital voice and wireless networks at the time. CALEA provides the Federal statutory framework for network operator assistance to LEAs in providing evidence and tactical information. In 2005, CALEA was applied to public broadband networks Internet access and Voice over IP services that are interconnected to the Public Switched Telephone Network (PSTN).

The Telecommunications Industry Association (TIA) and the Alliance for Telecommunications Industry Association (ATIS) jointly developed the joint standard J-STD-025-B for lawfully authorized electronic surveillance[20]. Electronic surveillance refers to the interception and monitoring of communications (i.e., call content), call-identifying information, or both, for a particular telecommunications subscriber as lawfully authorized. CALEA is applicable to the GSM, HSPA and LTE environments.

J-STD-025-B defines the interfaces between a telecommunication service provider (TSP) and a Law Enforcement Agency (LEA) to assist the LEA in conducting lawfully authorized electronic surveillance. A TSP, manufacturer, or support service provider that is compliant with J-STD-025-B will have a "safe harbor" under 107 of CALEA. J-STD-025-B defines the lawful intercept standards for both circuit switched and packet data telecommunications.

ATIS has also developed several specifications related to lawfully authorize electronic surveillance (LAES). The ATIS specification ATIS-0700005[21] defines LAES for 3GPP IMS based VoIP and other

[19] Pub. L. No. 103-414, 108 Stat. 4279, codified at 47 USC 1001-1010.
[20] ANSI/J-STD-025-B, *Joint Standard Lawfully Authorized Electronic Surveillance*, July 17, 2006.
[21] ATIS-0700005, *Lawfully Authorized Electronic Surveillance (LAES) for 3GPP IMS-Based VoIP and Other Multimedia Services*, May 2007.

multimedia services. The ATIS specification ATIS-1000013[22] defines LAES for Internet broadband access.

The 3GPP Technical Specification 33.106[23] defines the requirements for lawful intercept and 3GPP Technical Specification 33.107[24] defines the architecture and functions for lawful intercept.

6.2 EMERGENCY SERVICES

Emergency Services is the ability for a subscriber to place a voice call to a Public Safety Answering Point (PSAP) to request assistance. In North America, the three digit number, 9-1-1, is the most common way for a subscriber to request emergency services. In October 1999, the Wireless Communications and Public Safety Act of 1999 (9-1-1 Act) took effect with the purpose of improving public safety by encouraging and facilitating the prompt deployment of a nationwide, seamless communications infrastructure for emergency services in the United States.

The Wireless Communications and Public Safety Act of 1999 (9-1-1 Act) took effect on October 26, 1999. The purpose of the 9-1-1 Act is to improve public safety by encouraging and facilitating the prompt deployment of a nationwide, seamless communications infrastructure for emergency services. One provision of the 9-1-1 Act directed the FCC to make 9-1-1 the universal emergency number for all telephone services in the United States.

The FCC's wireless Enhanced 9-1-1 (E9-1-1) rules seek to improve the effectiveness and reliability of wireless 9-1-1 services by providing 9-1-1 dispatchers with additional information on wireless 9-1-1 calls. The FCC's wireless E9-1-1 rules apply to all wireless licensees, broadband Personal Communications Service (PCS) licensees, and certain Specialized Mobile Radio (SMR) licensees. E9-1-1 is divided into two parts:

1. Under E9-1-1 Phase I, the FCC requires the wireless carriers to provide the PSAP with the telephone number of the originator of a wireless 9-1-1 call and the location of the cell site or base station transmitting the call.

2. Under E9-1-1 Phase II, the FCC requires wireless carriers to provide information that is more precise to PSAPs, specifically, the latitude and longitude of the caller. This information must meet FCC accuracy standards, generally to within 50 to 300 meters, depending on the type of technology used.

3GPP has conducted several studies regarding the support of emergency calls. The 3GPP technical specification TS 22.101[25] defines the requirements for voice based emergency services. The 3GPP technical specification TS 23.167[26] defines the support of emergency calls via the IP Multimedia Subsystem (IMS) and, therefore, defines the support of emergency calls in the LTE environment.

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[22] ATIS-1000013, *Lawfully Authorized Electronic Surveillance (LAES) For Internet Access and Services*, March 2007
[23] 3GPP TS 33.106, *3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; 3G security; Lawful Interception requirements*
[24] 3GPP TS 33.107, *3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; 3G security; Lawful Interception architecture and functions*
[25] 3GPP TS 22.101, *3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Service aspects; Service principles*
[26] 3GPP TS 23.167, *3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; IP Multimedia Subsystem (IMS) emergency sessions*

The support of emergency services is mandatory for mobile devices that support wireless operator provided voice communications on GSM, HSPA or LTE networks.

6.3 TELECOMMUNICATIONS DEVICE FOR THE DEAF (TDD)

A telecommunications device for the deaf (TDD) is an electronic device for text communication via a telephone line, used when one or more of the parties has hearing or speech difficulties. This device and capability is also called TTY for teletypewriter since it is based upon the TTY protocols.

The typical TTY is a device about the size of a typewriter or laptop computer with a QWERTY keyboard and small screen that uses LEDs or an LCD screen to display typed text electronically. In addition, TTYs commonly have a small spool of paper on which text is also printed — old versions of the device had only a printer and no screen. The text is transmitted live, via a telephone line, to a compatible device, i.e. one that uses a similar communication protocol.

Also, there are systems in place so that a deaf person can communicate with a hearing person on an ordinary voice phone using a human relay operator. There are also carry-over services, enabling people who can hear but cannot speak (hearing carry-over a.k.a. HCO), or people who cannot hear but are able to speak (voice carry-over a.k.a. VCO) to use the telephone.

Support of this service is mandatory for wireless operators in the United States. Mobile phones sold in the United States must support this service via tethering to a TTY device. Support of this service is mandatory for wireless operators in the United States.

This service is based upon protocols which were designed over 60 years ago and is not well suited for the evolving wireless networks. The Non-Voice Emergency Services (NOVES) described in the next section is the evolution of this service for the next-generation networks.

3GPP Technical Specification TS 22.26[27] provides the requirements for the support of this type of service in 3GPP networks including IMS based networks.

6.4 NON-VOICE EMERGENCY SERVICE (NOVES)

For consumers, non-voice communication such as text messages and instant messaging via wireless devices has been very successful for non-emergency services. Unfortunately, many consumers assume that they can utilize these types of non-voice communications as mechanisms to communicate with emergency services whenever emergency assistance is required. Such mechanisms currently do not exist. The Emergency Services community has a desire to have multimedia emergency services supported with the same general characteristics as emergency voice calls.

Currently, 3GPP TS 22.101[28] service requirements for emergency calls (with or without the IP Multimedia Core Network) are limited to voice media. NOVES is intended to be an end-to-end citizen to authority communications. NOVES could support the following examples of non-voice communications to an emergency services network:

- Session based text messages (which does not include SMS) from citizen to emergency services

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[27] 3GPP TS 22.226, *3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Global Text Telephony; Stage 1*

[28] 3GPP TS 22.101, *3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Service aspects; Service principles*

- Session based and session-less instant messaging type sessions with emergency services
- Multimedia (e.g., pictures, video clips) transfer to emergency services either during or after other communications with emergency services
- Real-time video session with emergency services

In addition to support the general public, this capability would facilitate emergency communications to emergency services by individuals with disabilities (e.g., hearing impaired citizens).

A NOVES device is a next-generation end-user device (e.g., wireless LTE device) that utilizes trusted applications to provide secure transport of messaging and media content, and location information of the reporting device to the emergency authorities, in addition to two-way voice communications between citizens and emergency authorities (e.g., PSAPs).

3GPP technical report TR 22.871[29] is a recent study by 3GPP on non-voice emergency services. 3GPP is targeting the standardization of NOVES to be completed for 3GPP Rel-11.

The National Emergency Number Association (NENA) has also recently completed a technical informational document on non-voice-centric use cases and suggested requirements[30]. This document was one of the inputs into the 3GPP study mentioned above.

6.5 PRIORITY SERVICES

Priority services encompass both voice and multimedia communications.

The Federal Communications Commission (FCC) issued a Report and Order on July 13, 2000 allowing cellular providers to offer wireless priority services for circuit switched voice communications to personnel at the Federal, State, and local levels to help meet the National Security/Emergency Preparedness (NS/EP) communication needs of the United States. This ruling established the regulatory, administrative and operational framework that enables cellular service providers to provide Wireless Priority Service to NS/EP personnel.

Wireless Priority Service (WPS) is a system in the United States that allows high-priority emergency telephone calls to avoid congestion on wireless telephone networks. This complements the Government Emergency Telecommunications Service (GETS), which allows such calls to avoid congestion on landline networks. The service is overseen by the Federal Communications Commission (FCC) and administered by the National Communications System (NCS).

WPS only provides a higher probability of call completion. WPS will not preempt calls in progress, so the user will be queued and will have to wait for bandwidth to become available. The queuing priority of the WPS user is defined by the NCS when the WPS account is established for the user. WPS calls so not automatically get priority on landline networks. The user must use WPS in conjunction with GETS to have priority services on an end-to-end basis.

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[29] 3GPP TR 22.871, *3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Study on Non-Voice Emergency Services*
[30] NENA 73-501, *NENA Use Cases & Suggested Requirements for Non-Voice-Centric Emergency Services*, Version 1.0, January 11, 2011

The NCS in conjunction with the wireless industry developed the industry requirements for Wireless Priority Service (WPS) for GSM Systems[31]. The 3GPP Technical Specification 22.067[32] defines the WPS service requirements and 3GPP Technical Specification 23.067[33] defines the WPS architecture.

The FCC has not issued any rules or regulations regarding the support of priority services for IP based multimedia services. However, the NCS in conjunction with the wireless industry has developed an industry requirements document for the support of multimedia priority services in an UMTS environment [34]and a separate industry requirements document for the support of multimedia priority services in an LTE environment[35]. The 3GPP Technical Specification 22.153[36] defines the requirements for the support of multimedia priority services in 3GPP networks.

6.6 COMMERCIAL MOBILE ALERT SYSTEM (CMAS)

The Commercial Mobile Alert System (CMAS) is a mechanism for emergency officials to send alert messages to mobile devices within a specified area. There are three types of alert messages supported by CMAS:

1. Presidential alerts
2. Alerts pertaining to imminent threats to life or property
3. Child abduction/AMBER alerts

CMAS was developed as a result of the Warning Alert and Response Network (WARN)[37] Act which was part of the Security and Accountability Port Act of 2006 (SAFE Port Act) passed by the US Congress in September 2006 and signed into law by President Bush on October 13, 2006. In compliance with the WARN Act, the FCC established the Commercial Mobile Service Alerts Advisory Committee (CMSAAC) to development recommendations to the FCC.

The recommendations from the CMSAAC are contained in the FCC CMAS Notice of Proposed Rule Making (NPRM)[38]. Subsequent to the CMAS NPRM, the FCC has issued three separate CMAS reports and orders[39,40,41] and one CMAS reconsideration and erratum document[42]. The regulatory rules for CMAS are contained in the Code of Federal Regulations 47 CFR Part 10.

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[31] National Communications System (NCS), *Wireless Priority Service (WPS) Industry Requirements for the Full Operating Capability (FOC) for GSM-Based Systems*, Industry Requirement (IR) Document, Issue 2.0, January 2004

[32] 3GPP TS 22.067, *3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; enhanced Multi Level Precedence and Pre emption service(eMLPP); Stage 1*

[33] 3GPP TS 23.067, *3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; enhanced Multi Level Precedence and Pre emption service(eMLPP); Stage 2*

[34] National Communications System (NCS), *National Security/Emergency Preparedness (NS/EP) – Long Term Evolution (LTE) Access Network Industry Requirements (IR) for Next Generation Network (NGN) Government Emergency Telecommunications Service (GETS,* Industry Requirement (IR) Document, Issue 1.0, April 2010

[35] National Communications System (NCS), *National Security/Emergency Preparedness (NS/EP) – Universal Mobile Telecommunications System (UMTS™) Access Network Industry Requirements (IR) for Next Generation Network (NGN) Government Emergency Telecommunications Service (GETS,* Industry Requirement (IR) Document, Issue 1.0, April 2010

[36] 3GPP TS 22.153, *3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Multimedia priority service*

[37] Security and Accountability For Every Port Act of 2006 (SAFE Port Act), Pub.L. 109-347, Title VI-Commercial Mobile Service Alerts (WARN Act).

[38] FCC 07-214; *Federal Communications Commission Notice of Proposed Rulemaking in the Matter of the Commercial Mobile Alert System*; December 14th, 2007

[39] FCC 08-99, *Federal Communications Commission First Report and Order In the Matter of The Commercial Mobile Alert System*; April 9, 2008.

CMAS is applicable to all CMRS technologies including GSM, UMTS, and LTE. CMAS standards have been developed by ATIS and by joint ATIS/TIA work efforts. The ATIS CMAS standards are based upon the Cell Broadcast Service (CBS) capabilities as defined in 3GPP TS 23.041[43] and upon the Public Warning System (PWS) as defined in 3GPP TS 22.268[44].

Figure 23 shows the Cell Broadcast architecture for the support of CMAS in GSM networks[45]:

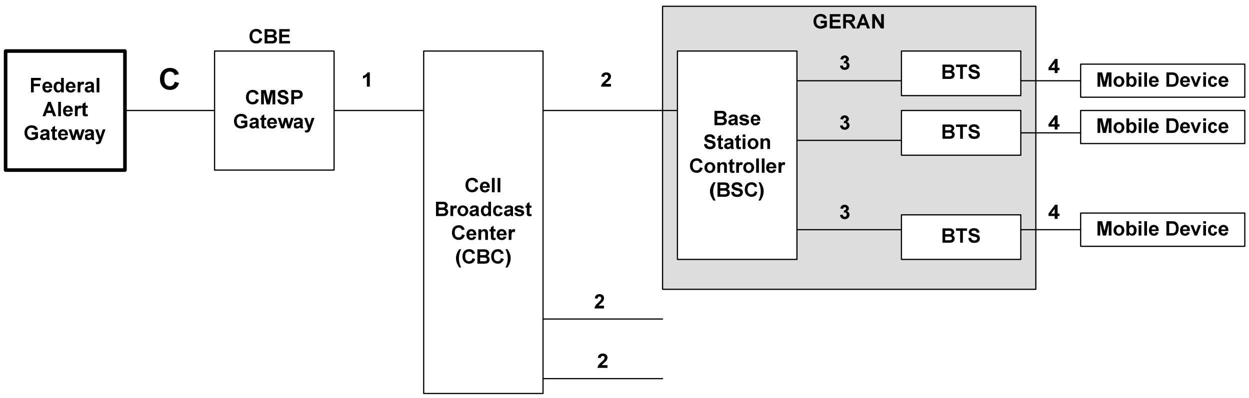

Figure 23: GSM Cell Broadcast Network Architecture for CMAS

Figure 24 shows the Cell Broadcast architecture for the support of CMAS in UMTS networks.[46]

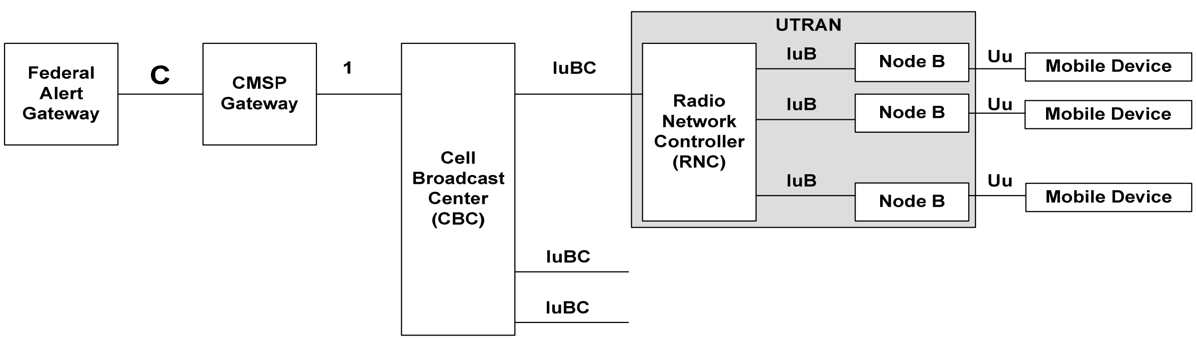

Figure 24: UMTS Cell Broadcast Network Architecture for CMAS

Figure 25 shows the warning system architecture for the support of CMAS in LTE networks[47].

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[40] FCC 08-164, *Federal Communications Commission Second Report and Order and Further Notice of Proposed Rulemaking In the Matter of The Commercial Mobile Alert System*; July 8, 2008.
[41] FCC 08-184, *Federal Communications Commission Third Report and Order and Further Notice of Proposed Rulemaking In the Matter of The Commercial Mobile Alert System*; August 7th, 2008.
[42] FCC 08-166, *Federal Communications Commission Order on Reconsideration and Erratum In the Matter of The Commercial Mobile Alert System*; July 15, 2008.
[43] 3GPP TS 23.041, *3rd Generation Partnership Project; Technical Specification Group Core Network and Terminals; Technical realization of Cell Broadcast Service (CBS).*
[44] 3GPP TS 22.268, *3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Public Warning System (PWS) Requirements*.
[45] ATIS-0700006, *CMAS via GSM/UMTS Cell Broadcast Specification*; March 2010
[46] ATIS-0700006, *CMAS via GSM/UMTS Cell Broadcast Specification*; March 2010

Standards have been developed as a joint effort between ATIS and TIA applicable to GSM, UMTS, and LTE networks[48]

ATIS also has its own standards for CMAS applicable to GSM, UMTS, and LTE networks[49], and LTE.[50]

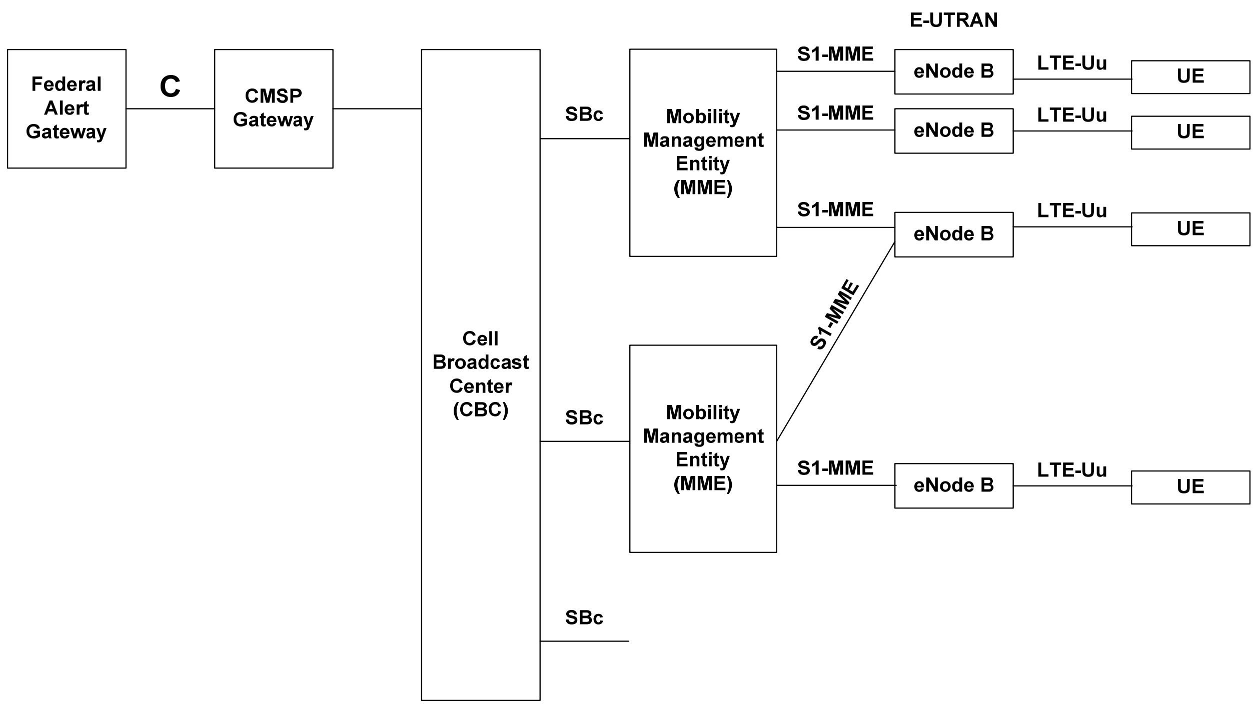

Figure 25: Warning System Architecture for CMAS

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[47] ATIS-0700006, *CMAS via EPS Public Warning System Specification*; August 2010

[48] ATIS/TIA joint standards for GSM, UMTS, and LTE networks: J-STD-100, Joint ATIS/TIA CMAS Mobile Device Behavior Specification, Jan 30, 2009; J-STD-101, Joint ATIS/TIA CMAS Federal Alert Gateway to CMSP Gateway Interface Specification, Oct 2009; J-STD-102, Joint ATIS/TIA CMAS Federal Alert Gateway to CMSP Gateway Interface Test Specification, Feb 2011.

[49] ATIS standards: ATIS-0700008, Cell Broadcast Entity (CBE) to Cell Broadcast Center (CBC) Interface Specification; March 2010; ATIS has developed the following standards for the support of GSM via Cell Broadcast Service in GSM and UMTS networks; ATIS-0700006, CMAS via GSM/UMTS Cell Broadcast Service Specification; March 2010; ATIS-0700007, Implementation Guidelines and Best Practices for GSM/UMTS Cell Broadcast Service Specification; October 2009.

[50] ATIS-0700010, CMAS via EPS Public Warning System Specification; August 2010

7 DEVICES FOR SEAMLESS MIGRATION

While device selection will be strongly driven by subscriber preferences when selecting which devices to offer to their customer base, LTE service providers will need to take into account the following factors:

- GSM, HSPA, LTE Interoperability
- Multi-mode devices, multi-band devices
- Dual stack IPv4/IPv6 capabilities
- Mobility features e.g. SRVCC HO and/or CS-Fallback
- Roaming

7.1 MULTIMODE AND MULTIBAND DEVICES

The term "multimode" is often used loosely in the wireless industry in the context of describing a mobile device. Within the 3GPP family of technologies, this multimode designation implies support for the legacy technologies in addition to the current technology. In the case of an LTE device, this means support for WCDMA/HSPA and GSM/EDGE in the same device, in addition to LTE.

As with any new wireless technology, building coverage takes time and usually starts in the high-density metropolitan areas. The legacy technologies such as GSM/EDGE or HSPA provide the backdrop for service continuity in areas which have not been covered by LTE.

The coverage growth model as outlined above takes advantage of multimode devices. These devices provide the ability to grow the coverage in the needed increments while allowing the operator to provide seamless interworking with the legacy technologies in areas where LTE is not present. As well, a multimode, multiband device allows an LTE subscriber to roam to other networks which may not yet support the LTE technology, perhaps using the HSPA capability in the device. This eliminates any technology islands and regions of limited usability.

Expanding the multimode capability in the mobile device from two 3GPP technologies to three requires the following:

- Good scale support from the chipset vendor community
- Expansion of the testing scope to include LTE (device testing and interoperability testing with networks)
- Interworking testing between LTE and the legacy technologies
- Sufficient RF band support capability in the device to ensure operation over the operator's own bands as well as the needed roaming bands, across all three technologies

As the technology progression from GSM to HSPA to LTE is taking place, the number of operating frequency bands is increasing dramatically. In the early days of GSM, mobile devices supported only one or two operating bands at most. The advent of the WCDMA/HSPA technologies brought more operating bands into consideration, requiring device and chipset manufacturers to provide solutions which supported additional bands in a mobile device, making tri-band and even quad-band support available over the 3G technologies, with quad-band GSM/EDGE support widely available in these same devices.

Today, some device manufacturers are even bringing penta-band WCDMA/HSPA support in mobile devices while still supporting all four of the underlying GSM/EDGE bands.

LTE brings additional bands for mobile device implementation consideration. At present there are more than 30 LTE bands defined in the 3GPP standards. As the LTE technology matures and network coverage increases, device manufacturers will be expected to support the local and roaming bands associated with LTE. This, in practice, could mean supporting two to three additional frequency bands, on top of what is required for the legacy technologies. LTE's ability to provide higher data rates than the legacy technologies is partly predicated on the use of multiple antennas to transmit and receive data. This advanced technology, known as MIMO (Multiple-Input Multiple-Output) requires multiple radio paths to be supported on the LTE bands. This effectively means multiple antennas must be implemented in the mobile device for the LTE bands meaning additional antennas and radio components such as amplifiers and filters.

Figure 26: Device eco-system will evolve with MIMO capability built in laptops, dongles, tablets, smartphones

The UE category of multimode HSPA terminal categories based on downlink capability[51] are listed in Table 4. In the table the term Dual Cell[52] also referred to as dual carrier allows for better resource utilization and spectrum efficiency by means of joint resource allocation and load balancing across the downlink carriers.

[51] 3GPP TS 25.306, *Universal Mobile Telecommunications System (UMTS) UE Radio Access capabilities*
[52] 3GPP TR 25.825, *Dual Cell HSDPA Operation*

Table 4: List of HSPA UE Categories (Q: QPSK, 16: 16QAM, 64: 64QAM) based on downlink performance

Category	Release	No MIMO No Dual Cell	MIMO No Dual Cell	Dual Cell No MIMO	MIMO Dual Cell	Max Speed
13	Rel-7	Q, 16, 64	-	-	-	17.6 Mbps
14	Rel-7	Q, 16, 64	-	-	-	21.1 Mbps
15	Rel-7	Q, 16	Q, 16	-	-	23.4 Mbps
16	Rel-7	Q, 16	Q, 16	-	-	28.0 Mbps
17	Rel-7	Q, 16, 64 -	- Q, 16	-	-	17.6 Mbps 23.4 Mbps
18	Rel-7	Q, 16, 64 -	- Q, 16	-	-	21.1 Mbps 28.0 Mbps
19	Rel-8	Q, 16, 64	Q, 16, 64	-	-	35.3 Mbps
20	Rel-8	Q, 16, 64	Q, 16, 64	-	-	42.2 Mbps
21	Rel-8	-	-	Q, 16	-	23.4 Mbps
22	Rel-8	-	-	Q, 16	-	28.0 Mbps
23	Rel-8	-	-	Q, 16, 64	-	35.3 Mbps
24	Rel-8	-	-	Q, 16, 64	-	42.2 Mbps
25	Rel-9	-	-	-	Q, 16	46.7 Mbps
26	Rel-9	-	-	-	Q, 16	55.9 Mbps
27	Rel-9	-	-	-	Q, 16, 64	70.6 Mbps
28	Rel-9	-	-	-	Q, 16, 64	84.4 Mbps

For example, a category 14 device specified in Rel-7 with 64 QAM can support a maximum speed of 21.1 Mbps. Similarly a category 20 device specified in Rel-7 with 64 QAM and dual cell (dual carrier) can support a maximum speed of 42.2 Mbps. A category 28 device specified in Rel-9 with 64 QAM, dual cell and MIMO can support a maximum speed of 84.4 Mbps.

Table 5 shows the LTE UE Categories. As shown, a category 1 UE has a maximum downlink speed of 10 Mbps without MIMO (1 layer for spatial multiplexing). A category 2 UE has a maximum downlink speed of 50 Mbps with 2X2 MIMO.

Table 5: LTE UE Categories

Category	1	2	3	4	5
DL peak rate	10	50	100	150	300
UL peak rate	5	25	50	50	75
Max DL mod	64QAM				
Max UL mod	16QAM				64QAM
Layers for spatial mux.	1	2			

Some multimode multiband devices are illustrated in Figure 27.

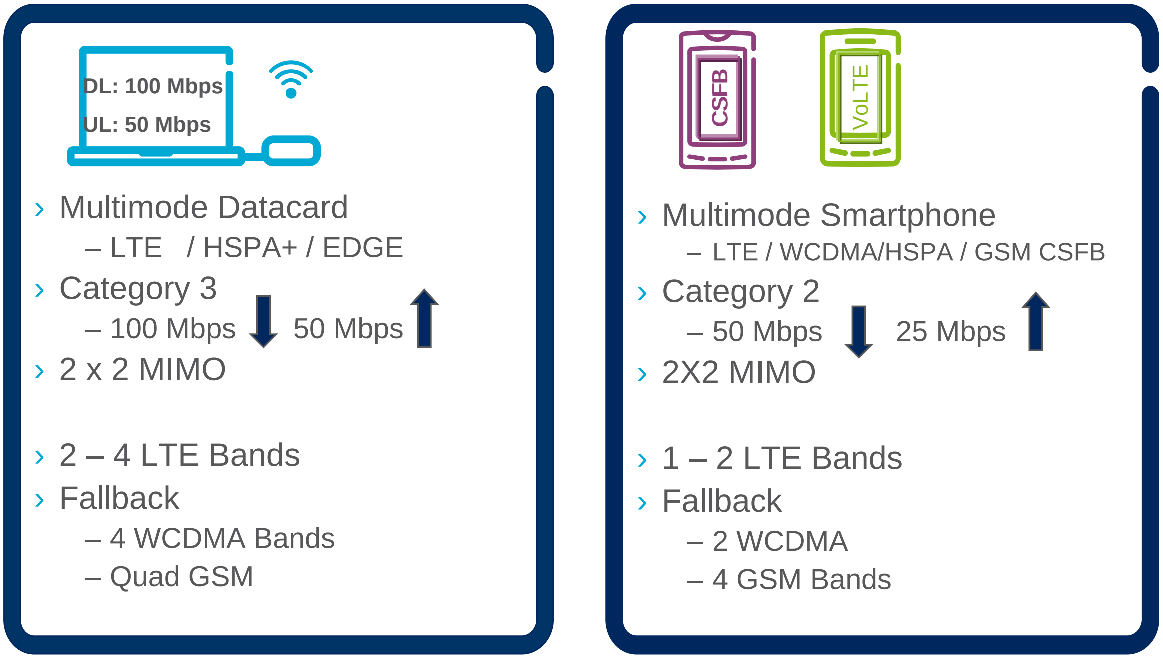

Figure 27: Examples of Multimode Multiband Devices supporting LTE, HSPA and GSM

When considering where the industry is moving, specifically towards multiple antenna technologies, the antenna design in the mobile device must consider the following:

- Frequency band of operation
- Signal to noise targets
- Interference mitigation targets (from other users both from inside the cell and in adjacent cells)
- Form factor of the device
- Other radios in the device which may include WLAN, GPS, Bluetooth, and FM Radio

When these factors are considered and then extended to include multiple frequency bands of operation, the design equation becomes more complex as the factors listed above all have a dependency on the frequency bands employed in the device. The other radios present in a device may operate near one of the required cellular frequency bands or may require some additional isolation inside of the device from one of these bands.

When considering frequency bands and combinations thereof, device manufacturers often categorize the bands in terms of the high band or low band designation. High bands are frequency bands which are greater than 1 GHz (examples: 1900, AWS, and 2100 MHz) the low bands are the frequency bands which are less than 1 GHz (examples 700, 850, and 900 MHz). The higher bands are generally easier to implement in a device in terms of size as the higher frequency bands require less antenna length. The lower bands, by nature, require a larger antenna for optimal performance.

ANTENNA SEPARATION AND FREQUENCY BAND OF OPERATION

There is no hard upper limit on the number of frequency bands that can be supported in a mobile device. On the practical side, the limits are generally set by cost, mobile device size and performance considerations.

In the increasingly complex handsets that have to support multiple radios and multiple bands, the antenna remains the critical analog component, whose performance is directly constrained by the laws of physics. The antenna size determines the upper bound of performance irrespective of the downstream electronics to process the captured signal. Antenna design factors include:

- Mutual coupling and interference between antennas operating in other frequency bands. This raises electromagnetic compatibility (EMC) and interference (EMI) issues. The coupled energy also negatively affects the radiation efficiency of the antenna

- Specific Absorption Rate (SAR) and Hearing Aid Compatibility (HAC) regulatory constraints. The efficiency of the antenna determines the strength of its unwanted near-field, which is directly related to the handset SAR reading. This near-field also determines the amount of energy that is coupled into the user's hearing aid device, and hence affects the HAC rating of the handset[53]

- Power limits and processing requirements are restricted by SAR constraints for transmitting and on battery life performance expectation for both transmit and receive. In many scenarios the handset needs to constantly measure and synchronize with network infrastructure on multiple technologies – in case it was asked to get subsequent service from another technology

Key factors that play into the above listed challenges are how well the antenna is designed and where it is placed in the handset. There is a known balance between the antenna size and its performance – the smaller the antenna is below the minimum required size, the lower its expected performance[54],[55]. The antenna size is that defined by the volume of the sphere enclosing the electrically small antenna radiating element, illustrated in Figure 28. Depending on the frequency of operation, the ground plane contributes to a portion of the antenna radiation in the far-field. Hence, the ground plane becomes part of the antenna radiating structure affecting the electrical size that defines the antenna.

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[53] S.M. Ali and H. Gu, *Effects of chassis currents on hearing aids compatibility in the handset,* IEEE Transactions on Electromagnetic Compatibility, vol. 53, no. 4, Nov. 2010, pp. 837-842

[54] H.A. Wheeler, *Fundamental limits of small antennas,* Proceedings of the I.R.E (IEEE), Dec. 1947

[55] L.J. Chu, "Physical limitations of omni-directional antennas," J. Appl. Phys., vol. 19, pp. 1163-1175, 1948

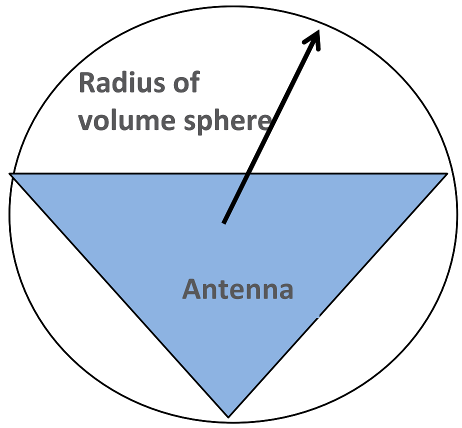

Figure 28: Illustration of the sphere defining the antenna size within the sphere volume

Table 6 shows the required sphere radius enclosing the antenna as a function of the operating frequency band that the antenna is expected to support and its bandwidth for the best efficiency possible. A lower frequency of operation implies a larger antenna.

Table 6: The radius of the volume sphere enclosing the antenna as a function of the operating frequency band and the supported bandwidth

Frequency (MHz)	Bandwidth supported (MHz)	Radius of volume sphere (mm)
700	48	28.3
850	70	25.0
900	69	22.8
1500	68	11.3
1700	44.5	20.6
1800	169	13.0
1900	140	10.7
2100	250	12.5
2600	190	7.70

Figure 29 shows the required antenna separation in terms of the device form factor as a function of the operating frequency. Techniques to increase the electrical separation between antennas continues to be an active research area and should relax the challenges for MIMO implementation below 1.4 GHz. At these frequencies, we can expect quad-band GSM, penta-band WCDMA, and single-antenna LTE. Early results indicate that the efforts to overcome the MIMO implementation challenges for lower frequency bands in handsets are bearing fruit.

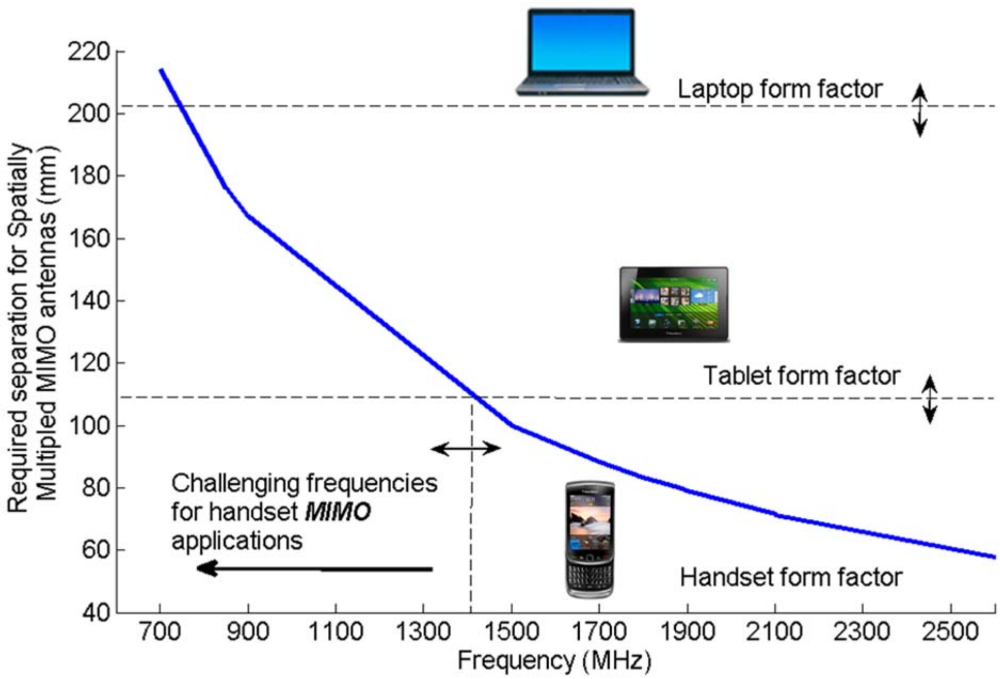

Figure 29: Required separation for MIMO antennas as a function of the carrier frequency

The discussed challenges have stimulated considerable efforts to find novel solutions towards facilitating the coexistence of multiple antennas and multiple transceivers in the handset. For example, there is currently significant drive for more flexible RF front end components, the RF filtering tasks have become more difficult while at the same time achieving significant miniaturization. Tunable components promise performance improvements in the order of (0.8-1) dB when use in tunable power amplifiers and filters; 0.3 dB in tunable switches; and (1.5-3.0) dB in tunable antenna matching circuits[56]. Another application for these tunable components is in the design of tunable antennas. With this, the antenna size could be significantly reduced without compromising its performance or the number of bands it supports[57]. Generally, tunable technology enables the reuse of portion(s) of the RF transceiver and/or the antenna. Thus, it relaxes the required real-estate and the bill of material on the handset PCB; however, where simultaneous operation is not required.

The benefits of tuning components have been understood for decades, but cost–effective technology capable of practically realizing this vision is only now reaching the market. The technology yet has to address issues such as cost and integration, certification, mature sensing capabilities in the handset, and the needed changes in the radio architecture to support them.

7.2 SUPPORT FOR LTE UE PROFILE DEFINED BY IR.92 FOR VOICE

The capabilities of a VoLTE device for IMS Voice and MMTel services are defined within GSMA, IR.92 which defines the minimum mandatory set of features such a device need to be capable of.

These capabilities are categorized in four areas:

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[53] Wispry Inc., *RF-MEMS for Wireless Communications,* IEEE Communications Magazine, Aug. 2008, (http://www.wispry.com)
[57] Agile RF (http://www.agilerf.com), *Analog Tunability in the RF Front End*, The International Wireless Industry Consortium (IWPC), Feb. 2008

1. IMS Basic and supplementary services for telephony
2. Real-time media negotiation, transport, and codec's
3. LTE radio and evolved packet core capabilities (e.g. GBR EPS bearer (QCI1) for voice, Non-GBR EPS bearers for SIP and XCAP etc.)
4. Functionality that is relevant across the protocol stack and subsystems (IP Version, Emergency call etc.)

Additionally, the capabilities related to early VoLTE deployments where LTE coverage is complemented with CS coverage are also defined (domain selection, SRVCC etc.).

A VoLTE UE based on above specification should support basic IMS capabilities as P-CSCF discovery, Registration and Authentication (IMS AKA), Addressing, session negotiation and call establishment according as well as defined codec (AMR NB and WB Codec & payload format) and media capabilities.

It is also defined that for IMS voice, only an "IMS" APN shall be used and this needs to be supported by the UE.

The list of mandatory supplementary services support for the UE is:

- Originating/Terminating Identification Presentation/Restriction 3GPP TS 24.607
- Communication Forwarding Unconditional/not logged in/Busy/not Reachable/No Reply 3GPP TS 24.604
- Barring of All Incoming/Outgoing Calls 3GPP TS 24.611
- Barring of Outgoing International Calls 3GPP TS 24.611
- Barring of Outgoing International Calls – ex Home Country 3GPP TS 24.611
- Barring of Incoming Calls - When Roaming 3GPP TS 24.611
- Communication Hold 3GPP TS 24.610
- Message Waiting Indication 3GPP TS 24.606
- Communication Waiting 3GPP TS 24.615
- Ad-Hoc Multi Party Conference 3GPP TS 24.605

The UE shall also be capable of managing the services over Ut/XCAP.

Besides general MMTel and VoLTE capabilities, the UE needs to be prepared (aligned with IR.92) for CS and PS coexistence. This implies support in the UE for:

- CSFB
- Emergency call support over CS (CSFB)
- SMSoSGs
- SRVCC capability

8 EVOLUTION TO COMMUNICATION WITH DEVICES

The communications industry and consumers are evolving through different phases:

- Initially, people communicating with people (phones, smartphones)
- Now, people communicating with machines (cloud computing)
- Evolving to machines communicating with machines (hospital devices with computers)

These additional forms of communication are in addition to those occurring today on 3G WCDMA/HSPA networks, and will occur very rapidly on new LTE networks. Therefore, it is important that Machine-to-Machine (M2M) is implemented on both 3G and LTE networks, and that these networks and devices are interoperable.

M2M devices such as sensors or meters are used to capture events such as temperature, fluid levels or other event. The information is then relayed through communication networks to a point of collection and analysis. These devices will need to coexist with the existing device ecosystem.

M2M equipment could be a device that is fully self-contained or an embedded device with interfaces to attach to other devices, for example, sensors and health care monitoring tools.

However the definition of M2M is evolving to include some level of human interaction as in embedded M2M module with encapsulated gadget. This type is commonly known as machine to human (M2H) or human to machine (H2M) communication. These gadgets will be geared to vertical markets such as healthcare, transportation, advertising and other segments where there is some human interaction.

Standardization will focus on the base technology layers and allow innovation at the application layer. Key areas for standardization include: security, remote management, network components, and device intelligence. Device reliability is also an important area since some of these devices are expected to survive in extreme conditions.

MTC devices will proliferate once standardization is completed. This will result in higher volumes, more reliable, interoperable, and lower cost devices. Figure 30 represents an architectural overview of M2M ecosystem and various components involved.

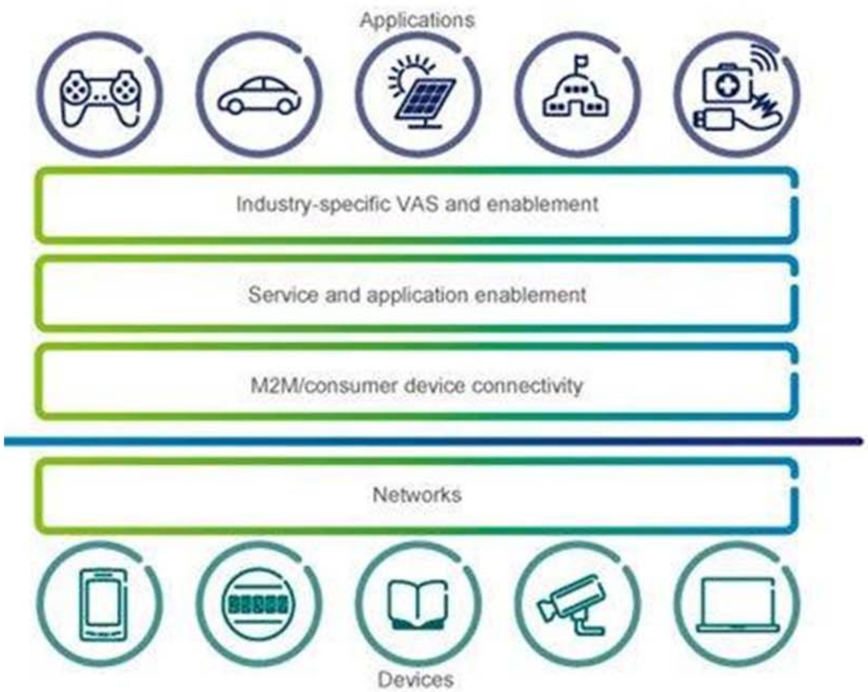

Figure 30: M2M Ecosystem Architecture

Connected devices are those devices which are used outside of the traditional use such as voice or data communications. These devices include e-readers, tablets, GPS, alarms, telematics and picture frames that exchange content with the Internet. An analysis shows that connected device subscriptions are growing at 18.7% and 10.6% compounded annual growth rate while traditional subscriber growth rate is 2.9% and 1.2% respectively for the same subscriber base.

Machine Type Communication (MTC) ranges from low bandwidth, low transaction devices to high bandwidth, high transaction devices. MTC is different from traditional services in terms of how the services will be marketed. In smart metering LTE modules will be embedded into metering devices attached to homes to capture and deliver meter readings. It can also be used in tracking reporting temperature readings of solution or an environment.

Figure 31: Machine Type Communication (MTC) Network Architecture

MTC applications also have the potential to include large number of communicating terminals with small amounts of traffic per terminal.

The primary issue with M2M devices is the lack of standardization[58] on a global level. Currently 3GPP, ETSI, ITU, IEEE and others are working on M2M standards. There is ongoing standardization in these areas. Only baseline requirements have been defined today.

3GPP has defined service requirements that are common to all MTC devices (e.g. MTC device triggering, addressing, identifiers, charging), and requirements that are specific to MTC use cases including:

- Low mobility MTC devices
- Time controlled communications
- Mobile originated only communications
- Group based policing and Addressing

8.1 NETWORK SECURITY

No specific requirements relative to security have been defined in 3GPP for MTC devices.

ETSI introduces in TS 102 671 specific form factors for M2M UICC and addresses UICC constraints and aspects related to M2M environments such as life time, temperature, vibration, pairing etc. Remote Management of USIM application is as specified in 3GPP in TS 31.115[59] and TS 31.116[60].

8.2 NETWORK CONSIDERATIONS

The deployment of active M2M devices will increase network signaling and traffic load. Failure to understand the impact of MTC devices on network load can potentially result in complete network degradation. For example, if a large number of MTC devices go down and all try to reconnect at the same time. The standard will define different use cases to address some unique requirements including various loading requirements.

Other potential exists for operators to extend the Home Subscriber Server (HSS) to support machine type devices. This will mean adding new M2M profiles to address different device types, specifications, priority levels and owners. This will also mean automation and authentication of key management for sensors and similar device types. The current standards does not yet provide the details on how should be done.

One potential solution is to schedule the sending of data from non-real-time M2M applications during non-peak hours.

8.3 DEVICE IMPLICATIONS (MAN-MACHINE INTERFACE OF UE)

Device capabilities will impact what kind of network architecture will be required. For example, for high power/processing devices existing macro cell architecture may be suitable. While, for low power/complexity devices mesh/relay architecture may be suitable. Given the range of M2M use cases

[58] 3GPP Rel-10 TS-22.368, TR 22.988 and TR 22.889 define and provide high level description and some use cases for MTC. Subsequent work is being done in 3GPP Rel-11 and future iterations of the standards.

[59] Secured packet structure for (Universal) Subscriber Identity Module (U)SIM Toolkit applications (http://www.3gpp.org/ftp/Specs/html-info/31115.htm)

[60] Remote APDU Structure for (Universal) Subscriber Identity Module (U)SIM Toolkit applications (http://www.3gpp.org/ftp/Specs/html-info/31116.htm)

(Metering, Tracking; Health monitoring etc.), device requirements on a single network architecture (e.g., cellular; mesh; relay etc.) may not be sufficient.

Figure 32: M2M Communications for different applications

There is also discussion that the mobile phone could act as a hub for multiple M2M devices. It can be used to aggregate traffic from low-powered networks such as ZigBee and transport the traffic over the cellular network.

One of the most significant problems with MTC devices will be how to uniquely identify and track usage of each device. ETSI proposed using E.164 for the short and medium term numbering scheme with migration to IPv6 over the long term. In some cases, there are regulatory needs to differentiate M2M devices from traditional devices.

Each device will need a unique IP. The current shortage of IPv4 addresses will mean that IPv6 will be required to support the billions of MTC devices forecasted. In the interim, devices may use tunneling of IPv6 over IPv4.

It has to be noted that proper protection of USIM credentials requires the security endpoint for remote management to be in the UICC rather than in the device. This is not provided by device management protocols[61].

[61] MTC devices will also be managed under OMA DM specifications and the USIM by OTA as specified in 3GPP TS 31.115 and TS 31.116.

8.4 MTC IMPLICATIONS FOR COEXISTENCE OF GSM, HSPA, LTE

There are several factors relevant to MTC coexistence on multiple 3GPP technologies. Some include lower chipset cost for existing technologies, low bandwidth requirements and low network impact of existing solutions. As the numbers of MTC applications increases and bandwidth requirements change, operators have to consider which applications are best supported on legacy networks verses LTE.

Applications that require higher bandwidth and lower latency, such as medical images, HD video or live video chat, are better supported by LTE.

Another factor to consider is the length of time that an application or a solution will be in place. Many MTC applications are expected to last for long periods of time with minimal change. Operators have to consider the impact of changing a large number of devices once they retire their legacy networks or using LTE solutions even with the higher costs.

MTC standardization in 3GPP will lead to more mass market solutions thus reducing MTC long term costs. Standardization should also include interoperability between the different technologies resulting in smother migration between legacy solutions and LTE.

CONCLUSION

This whitepaper addresses the issues and technical solutions related to the successful coexistence of GSM, HSPA and LTE networks. The RAN (radio access network) issues are explored. The MSR (Multi-Standard radio) is explained as a solution to maximizing spectrum reuse. Roaming issues related to the RAN, Core and Devices are touched upon in their respective sections. Core network considerations including, migration from 3GPP Rel-7 to Rel-8 architecture are then reviewed. An overview of QoS for Rel-7 vs. Rel-8 and beyond is presented. Then LTE services including Voice over LTE is presented along with coexistence solutions including CSFB (circuit switched fallback) and SRVCC (Single Radio Voice Call Continuity). IMS Centralized Services are presented. The chapter concludes with a discussion on SMS and IMS Messaging. Regulatory Issues related to Lawful Intercept, Emergency Services, Telecommunications Device for the Deaf (TDD), Non-Voice Emergency Services (NOVES), Priority Services and Commercial Mobile Alert Service (CMAS) are presented at length. Devices for seamless migration are discussed with a focus on multiband and multimode devices. The MIMO capability in these devices and challenges are presented. Finally, an entire chapter is devoted to Machine Type Communication (MTC and M2M).

The 3GPP ecosystem will entail the coexistence of GSM, HSPA and LTE technologies in the coming years. This whitepaper addresses the intricate issues and solutions related to making the migration successful in this multi-technology environment, and take advantage of the economies of scale for years to come.

GLOSSARY

3GDT	3GPP Direct Tunnel
3GPP	3rd Generation Partnership Project
CMAS	Commercial Mobile Alert System
CN	Core Network
eNodeB	evolved NodeB
eUTRAN	evolved UTRAN (also known as LTE)
EPC	Evolved Packet Core, previously known as SAE
EPS	Evolved Packet System
GBR	Guaranteed Bit Rate
GGSN	Gateway GPRS Support Node
GPS	Global Positioning System
GERAN	GSM Radio Access Network
HSDPA	High Speed Downlink Packet Access
HSPA	High Speed Packet Access
HSS	Home Subscriber Server
IETF	Internet Engineering Task Force
IMS	IP Multimedia System (3GPP's SIP-based service architecture)
IP	Internet Protocol
LTE	Long Term Evolution
LEA	Law Enforcement Agency
M2M	Machine to Machine
MTC	Machine type Communication
MBR	Maximum Bit Rate
MME	Mobility Management Entity
MMTel	Multimedia Telephony Service
MSC	Mobile Services Switching Centre
MSR	Multi Standard Radio
NOVES	Non-Voice Emergency Services
OTA	Over The Air
P2P	Peer to Peer
PCC	Policy and Charging Control
PCRF	Policy and Charging Rules Function
P-CSCF	Proxy CSCF
PDN-GW	Packet Data Network Gateway
PDP	Packet Data Protocol
PDSN	Packet Data Serving Node

POTS	Plain Old Telephone Service
PS	Packet Switched
PSAP	Public Safety Answering Point
PSTN	Public Switched Telephone Network
PTT	Push To Talk
QCI	QoS Class Identifier
QoS	Quality of Service
RAN	Radio Access Network
RNC	Radio Network Controller
SACC	Service Aware Charging and Control
SAE	System Architecture Evolution, now known as EPC
SAPC	Service-Aware Policy Controller
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SRVCC	Single Radio Voice Call Continuity
UE	User Equipment
UTRAN	UMTS Terrestrial Radio Access Network
VCC	Voice-call continuity
VoIP	Voice over IP
VoLTE	Voice over LTE
WCDMA	Wideband Code Division Multiple Access
WPS	Wireless Priority Service

APPENDIX

The following table denotes the 3GPP Frequency Bands from 3GPP TS 36.101.

Table 7: LTE FDD/TDD Frequency Bands

FDD		
Band	"Identifier"	Frequencies (MHz)
1	IMT Core Band	1920-1980/2110-2170
2	PCS 1900	1850-1910/1930-1990
3	GSM 1800	1710-1785/1805-1880
4	AWS (US & other)	1710-1755/2110-2155
5	850	824-849/869-894
6	850 (Japan #1)	830-840/875-885
7	IMT Extension	2500-2570/2620-2690
8	GSM 900	880-915/925-960
9	1700 (Japan)	1750-1785/1845-1880
10	3G Americas	1710-1770/2110-2170
11	1500 (Japan #1)	1428-1448/1476-1496
12	US 700	698-716/728-746
13	US 700	777-787/746-756
14	US 700	788-798/758-768
17	US 700	704-716/734-746
18	850 (Japan #2)	815-830/860-875
19	850 (Japan #3)	830-845/875-890
20	Digital Dividend	832-862/791-821
21	1500 (Japan #2)	1448-1463/1496-1511

TDD		
Band	"Identifier"	Frequencies (MHz)
33 34	TDD 2000	1900-1920 2010-2025
35 36	TDD 1900	1850-1910 1930-1990
37	PCS Center Gap	1910-1930
38	IMT Extension Center Gap	2570-2620
39	China TDD	1880-1920
40	2.3 TDD	2300-2400

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