VOICE OVER LTE STUDY AND TEST STRATEGY DEFINITION

by

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A Thesis Presented in Partial Fulfillment

of the Requirements for the Degree

Master of Science

Approved April 2014 by the
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ARIZONA STATE UNIVERSITY

May 2014

ABSTRACT

Voice and other circuit switched services in a LTE deployment can be based on a Circuit Switched Fall Back mechanism or on the upcoming Voice Over LTE option. Voice Over LTE option can be used with its SIP based signaling to route voice calls and other circuit switched services over the LTE's packet switched core. The main issue that is faced though is the validation of this approach before the deployment over commercial network. The test strategy devised as a result of this work will be able to visit corner scenarios and error sensitive services, so that signaling involved can be verified to ensure a robust deployment of the Voice Over LTE network. Signaling test strategy is based on the observations made during a simulated Voice Over LTE call inside the lab in a controlled environment. Emergency services offered are carefully studied to devise a robust test strategy to make sure that any service failure is avoided. Other area were the service is routed via different protocol stack layer than it normally is in a legacy circuit switched core are identified and brought into the scope of the test strategy.

DEDICATION

To my parents and my sister, who have always been by my side, constantly encouraged me to pursue my dreams and encouraged me to work towards the wellbeing of mankind.

ACKNOWLEDGMENTS

I am thankful to Prof Reisslein for giving me an opportunity to pursue this work in a direction that I wanted to take it in and also for all the guidance, advice and wisdom that he offered to help me finish this work.

I am thankful to my dear friends Hitesh Khunti, Mohit Shah, Prasanna Sattigeri for all the motivational and technical discussions without which this journey would have been difficult.

I am thankful to Mr. Bryan Mikesh, Mr. Miguel Garrido and my friend and colleague Axel Robinson for all their unconditional support and words of wisdom.

TABLE OF CONTENTS

CHAPTER PAGE

CHAPTER PAGE

LIST OF TABLES

LIST OF FIGURES

Chapter 1
INTRODUCTION

1.1 Background and Motivation

The idea of providing voice and video call services along with other circuit switched services over a LTE network using the VOLTE approach has been driving global telecommunication industry. The main advantages is the concept of a single radio access technology that supports all services which will be a cost efficient and radio resource efficient approach. Along with the added advantages of deeper penetration of LTE radio frequencies due to their lower channel frequency compared to the legacy UMTS and GSM channel frequencies. This will in turn add value to the existing emergency services by being more accessible in basements and enclosures. The main gap between the formalization and realization of the VOLTE deployment is the uncertainty that brews from changes in the stack and signaling architecture, backward compatibility, mobility etc. the uncertainty in the quality of already established and important services pose a serious challenge. The test strategy devised as part of this work is to close this gap and to add to the existing stability and performance test strategies to make sure the system is robust before deployment. This is of utmost importance as one failed emergency call can cost a life.

1.2 Prior Work

A lot of attempts have been made since the advent of LTE and that of VOLTE on the performance optimization of Random access procedures and algorithms to optimize the use of exponential back offs or other mechanisms in Random access procedures [8, 9, 14, and 15]. Also a lot of work has been done on specific protocol stack layers to point out their efficiencies and measure their performance.

1.2.1 The LTE Link Layer

As per the study by Anna Larmo, Magnus Lindström, Michael Meyer, Ghyslain Pelletier, Johan Torsner, and Henning Wiemann, Ericsson Research [6], the LTE link layer standardized by 3GPP release 8 has been pointed out as an efficient design both in terms of complexity and also performance. There is no extension into the interworking of this stack with other protocol stack elements as part of a complex system. The study talks about all the aspects of the link layer design like the random access procedure, handover functionality, discontinues reception, among other features.

1.2.2 Random Access Procedure

Lot of study has been done on optimization of Random access procedure optimization be it regarding the back off mechanism, performance related study among other. It has been discussed how an exponential back off [16. 17] mechanism performance is analyzed with respect to obtaining maximum through put and through analysis of medium access delay observed in a packet being transmitted across in the random access procedure [8, 9, 15, 27].

1.2.3 Downlink scheduling and prioritization

A study done on the need for prioritization of multimedia packets over the LTE network shows the deterioration of performance due to lack of prioritization. In this study the approach is to analyze the downlink schedulers have been studied and based on the results obtained the need for prioritization of multimedia packets is deemed essential to obtain good quality of service in VOIP and real time services like video calls [19].

1.3 Proposed Test Strategy to analyze VOLTE performance

In this literature we discuss an approach focusing more on the stability and call retention in case of a Voice over LTE radio network. The objective is to attain a more signaling based test approach to understand the issues that may be observed in case of concurrency scenarios, mobility scenarios and general quality of the service in question. This approach will enable anyone attempting to deploy a VOLTE on their existing LTE network to assess the performance of the VOLTE network and its interworking with various devices that may use the network to place and receive VOLTE based calls. The test approach is devised to be able to assess various layers of the protocol stack with the same scenarios.

It is a more system based approach rather than analyzing individual components separately.

1.4 Literature Outline

The literature is classified into various chapters each catering to a different segment under discussion. Chapter 2 discusses the LTE architecture and further briefly discusses different procedures like the attach procedure and the IMS registration procedure. It also discusses the SIP packet and the RACH request/ response packets. Chapter 3 discussed about various different CS service options explored over time in the LTE system. Chapter 4 discussed Voice Over LTE option in detail along with the ISIM application present on the UICC. Chapter 5 discussed the test setup used and Chapter 6 accounts for the observations made and the final test strategy proposed. Chapter 7 includes a few conclusions drawn from the literature.

Chapter 2

LTE ARCHITECTURAL DETAILS

2.1 LTE network architecture:

Basic E-UTRAN architecture is standardized in 3GPP36.401. The idea is to have multiple eNodeBs that will provide the last mile radio access to the users. These eNodeBs are connected via wired connections with the Evolved Packet Core (EPC) [11] as is shown in the fig 1. The eNodeBs are connected to the EPC via logical interface S1 and the eNodeBs are connected to each other via logical interface X2. [1, 2, 4]

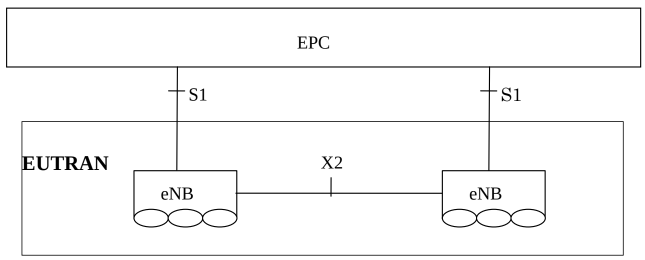

Figure 1 Basic EUTRA core network architecture [1]

The UE will be communicating with one or more eNodeBs in soft handover state. The signaling is differentiated as control plane and the user plane. The control plane describing the signaling phase while establishing communication and providing functions like power control once the call is established. So control plane activities and signaling concern with establishing, maintaining and terminating calls. The control plane is responsible for activities including session establishment, mobility, power control, radio resource management, Medium Access Control and Radio Link control protocol signaling among others. The user plane is representative of the actual user data and the involved communication including the various application level details. Hence once the connection or call has been established the data from the applications/ users flow over these established connections encapsulated. These procedures are handled in the user plane. Data performance related issues are to concern more with user plane than control plane, with a few exceptions.

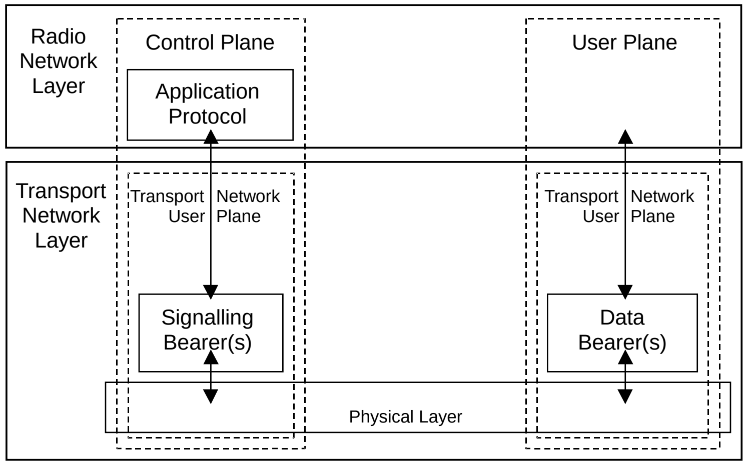

Figure 2 Control Plane and User plane interaction logical diagram. [1]

On a bird's eye logical view of the whole network the EUTRA network along with the other RAN technologies appear as in the fig 3.

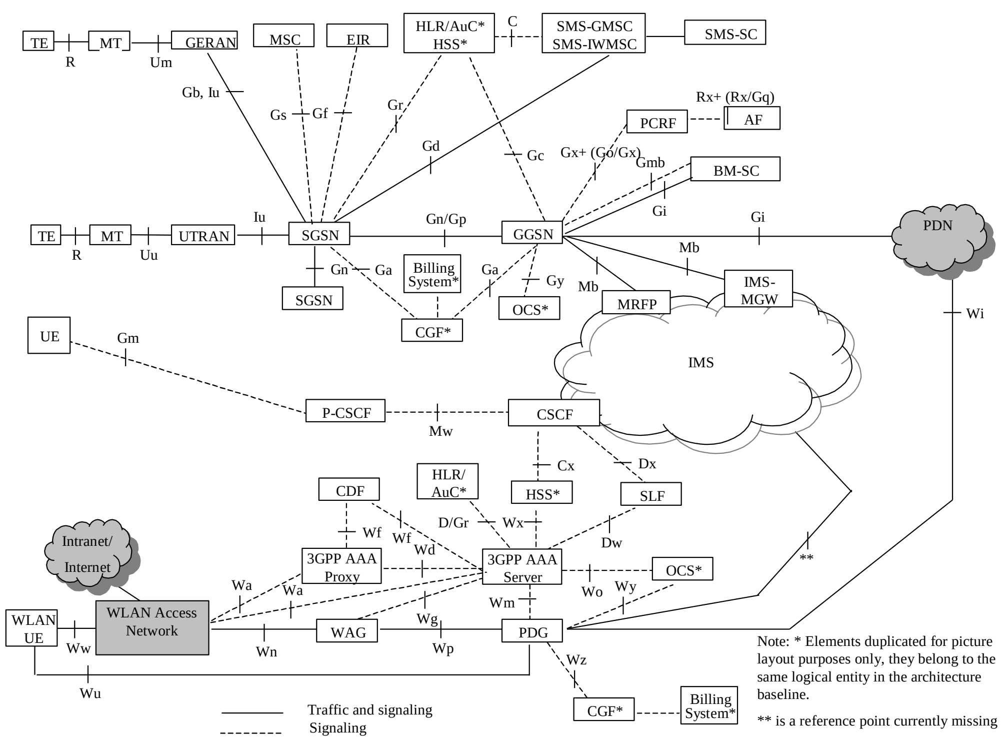

Figure 3 LTE Architecture logical representation Birds Eye View along with other RAN [2]

2.1.1 GERAN Section

The GERAN [21, 22] part of the network mainly contains the BTS as the Radio Access point connected to the MSC-HLR and MSC-VLR also containing the Authentication Center and the Equipment identity register elements. This is further connected to the GPRS or data network via the SGSN [12, 13] and GGSN further being interfaced with the global internet/ data access. This section contains the GSM, GPRS, EDGE radio access.

2.1.2 UTRAN Section

The UTRAN [21, 22] part consists of the eNodeB which is part of the UTRAN radio access. The eNodeB, further being connected with the SGSN which interfaces with the MSC HLR/ VLR, SMS center, Authentication center, Equipment Identity Register and GGSN network element. This enable the CS and PS access parts of the UTRAN network and provides mobility between the GERAN and UTRAN Radio Access Technologies. This section contains the UMTS, HSPA radio access.

2.1.3 EUTRAN Section

The EUTRAN section in this particular architectural representation is interfaced via the P-CSCF which is part of the IMS core network to provide the CS and PS services sought out by the EUTRA UEs. This section contains the Long Term Evolution (LTE) and going forward the LTE Advanced radio access.

2.2 Evolved Packet Core

EPC or Evolved Packet Core is the latest evolution of the 3GPP core network. The architecture of EPC is completely based on supporting IP as the network layer protocol for transferring all data and signaling. Hence the EPC is designed to have an all PS core and not to possess a circuit switched core. The EPC was first officially standardized and released in 3GPP Rel8 specifications. UE connects on a radio link with the eNodeB. The eNodeB further is connected via a wired connection to the EPC. [20, 21]

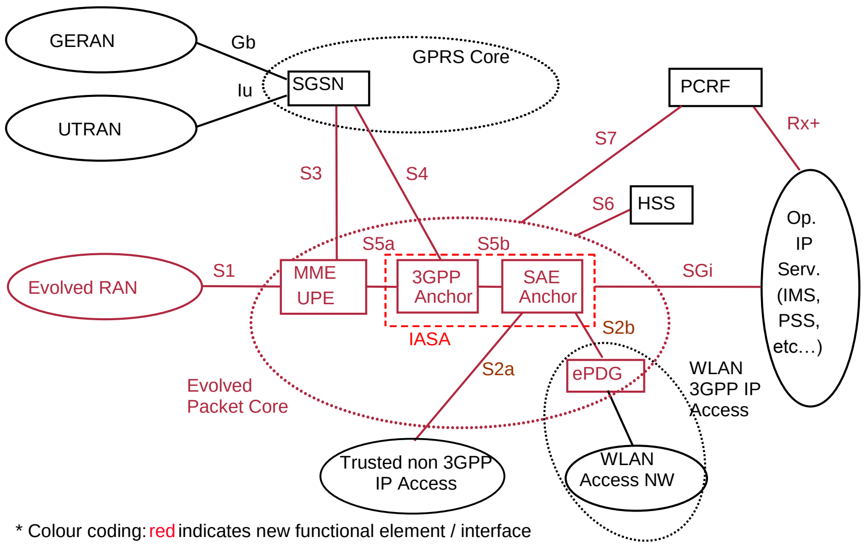

Figure 4 High Level Logical representation of SAE [2]

Fig 4 represents the logical interpretation of how the EUTRA network and the legacy GERAN/ UTRAN networks co-exist and are interconnected via the EPC which further connects to the IMS server and other access technologies. The EPC also connects to the Home Subscriber Server that contains all user related information and subscription details that help in authenticating the UE and also contains the present positioning of the UE so that an incoming call can be correctly routed to the UE.

2.2.1 Mobility Management Entity

Mobility Management Entity [2, 22] functionality includes Non-Access Stratum Signaling and security procedures. All inter core network signaling and 3GPP based mobility maintenance and facilitation is taken care of by this network element. Further this element maintains all Tracking Area list and individual Tacking Area Identities of various sectors/ cells. The functionalities of this network element further include the selection of Packet Data Network Gateways and Serving Gateways and also mapping of various TAIs to respective Time zones along with many other important functions mainly related to enabling mobility of the UEs.

2.2.2 3GPP Anchor

This element in fig 4 interconnects the MME with the SGSN from the Legacy GERAN/ UTRAN networks. This element facilitates in the mobility between the EUTRA and the GERAN/ UTRAN.

2.2.3 SAE Anchor

SAE refers to System Architecture Evolution. This is a core network evolution of 3GPP's GPRS core. The aim of this was to improve data performance by reducing latency and improve interoperability with other technologies while moving towards an all IP network design. This element in fig 4 interconnects the non 3GPP access technologies to the MME and the 3GPP anchor entity. This enables to provide a common access architecture for both 3GPP and non-3GPP access technologies and may later transcend into a seamless handover architecture between technologies and access techniques.

2.2.4 Home Subscriber Server (HSS)

This is a database server containing all information regarding the user which can be used for authenticating the user, subscription related information along with other user specific information.

2.2.5 IP Multimedia Subsystem (IMS)

IMS [4] is a network architecture framework. This is developed for and adopted by 3GPP standards as part of their evolution into higher data rate network architectures that will provide seamless services with a convergence of different Radio Access Technologies.

2.3 LTE Registration Procedure

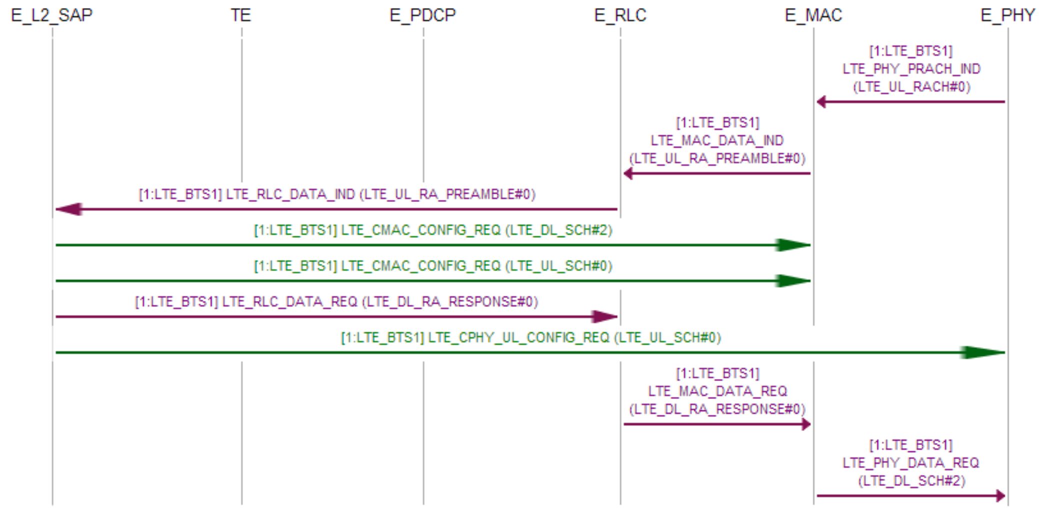

Figure 5 RACH procedure, Anritsu Exported Protocol Analyzer Traces, MSC View type log

The UE on being switched on and on observing the available EUTRAN RF, sends across a Random Access Request with a random number, on receiving the Random Access Response the UE goes ahead with sending across a RRC connection request to establish a RRC connection on which the rest of the signaling session will be built on. The network responds with a RRC connection setup message instructing the UE of all the radio resource availability. To this the UE responds with a RRC connection setup complete message acknowledging the successful establishment of a RRC connection. [23-26]

Once the RRC connection is established the UE proceeds with the attach procedure. The attach procedure begins with an attach request piggybacked with a PDN connectivity request. The network on receiving these requests begins an authentication procedure and a security mode procedure. This ensures the UE is recognized by the network and the network is recognized by the UE and a secure line of communication is established leading to the Attach procedure completion with attach complete message and the activation of the default EPS bearer. The initial RACH procedure along with the physical

RACH packet captured in a generic registration process on the test system is depicted in fig 5.

2.3.1 RACH Indication

The RACH indication sent across by the UE at physical layer view is as in Fig 6. The preamble index is used to avoid contention and collision if multiple UEs transmit RACH at the same time. The preamble index values are between 0 to 63 so there is a possibility of having a collision in case of many UEs are attempting to access the network at a crowded place. In which case the UE just traces back on not receiving any response and re-sends the RACH with a new pre-amble index. Usually on the re-transmission attempt the UE ramps up its power as per the retransmission power RAMP up specified in the system information.

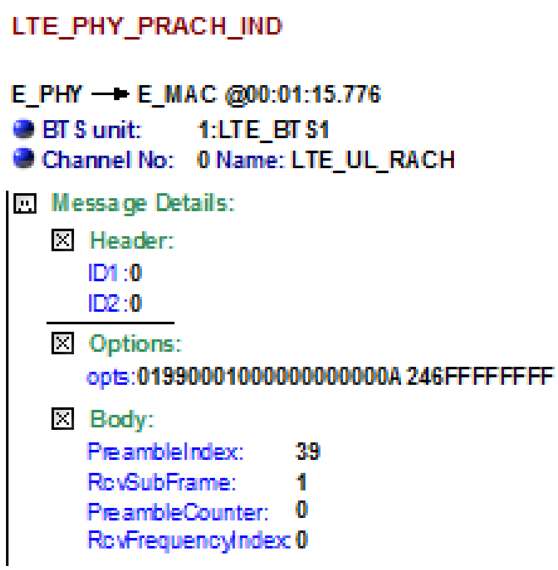

Figure 6 Random Access (RACH) Indication as received at Physical layer.

2.3.2 RACH Response

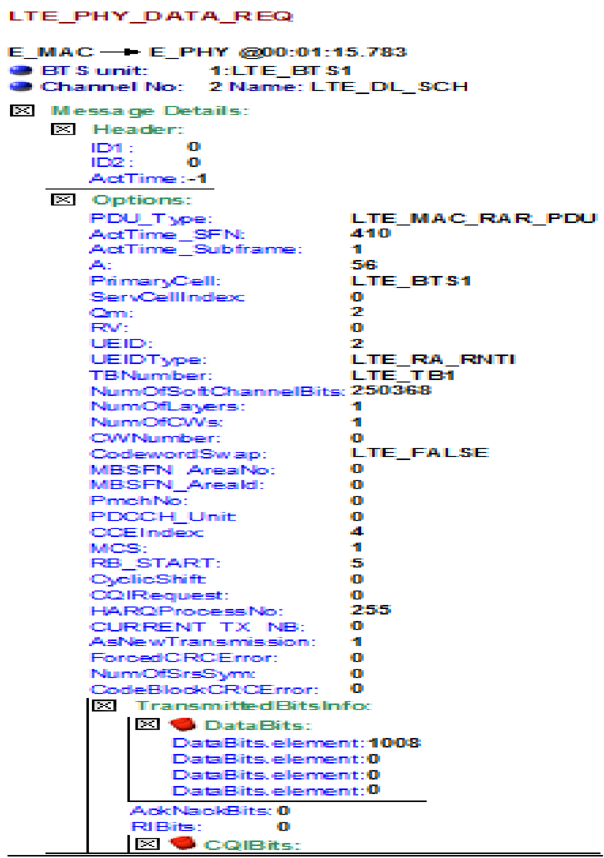

Figure 7 Random Access Response as perceived at Physical layer.

A RACH Response is sent by the network to the UE on receiving a RACH request. This message contains the grant information for further UL transmission from the UE. The signaling continues with the UE sending an RRC connection request to complete the registration procedure and establish a default EPS bearer. RACH response as perceived at Layer 1 is represented in Fig 7.

2.3.3 Attach Procedure

Figure 8 Attach procedure and Default EPS Bearer Activation, Anritsu Exported Protocol Analyzer Traces, MSC View type log

Prior to the establishment of the EPS bearer, the UE capability enquiry procedure is used to take account of the features and services supported by the UE. The UE responds with a complete set of supported Feature Groups and other supported services and capabilities. This procedure is not captured in the figure above. After the EUTRAN is made aware of the UE supported capabilities the EUTRAN attempts to establish an EPS bearer. This is the data connection with the LTE network on which the UE will further communicate. This is done by a RRC connection reconfiguration message from the EUTRA to which the UE responds with a RRC connection reconfiguration complete message. Once the

EPS bearer is established the proceeds with an Attach procedure by which the UE attaches to the packet core of the LTE network. Attach complete message is transmitted to the UE along with the activation of a default EPS bearer. The signaling is captured in Fig 8.

2.4 IMS Registration

Once the data connection is established in the form of a Default EPS bearer, the UE can obtain the S-CSCF IP address by interrogating the I-CSCF server via the P-CSCF server in the IMS core. The test simulation is in a controlled environment so the Application used to simulate the ISIM parameters has been configured with the S-CSCF ip address and as seen in the figure below the interrogation part is skipped and the UE directly proceeds to the Registration process with the S-CSCF SIP server. The registration process SIP messaging is as in the Fig 9.

No.	Time	Source	Destination	Protocol	Length	Info
4	1.157855	192.168.157.11	192.168.157.18	SIP	1009	Request: REGISTER sip:anritsu-cscf.com
5	1.161564	192.168.157.18	192.168.157.11	SIP	725	Status: 200 OK (1 bindings)
6	1.190862	192.168.157.11	192.168.157.18	SIP	906	Request: SUBSCRIBE sip:users1@anritsu-cscf.com
7	1.195517	192.168.157.18	192.168.157.11	SIP	459	Status: 200 OK
8	1.196741	192.168.157.18	192.168.157.11	SIP/XML	872	Request: NOTIFY sip:users1@anritsu-cscf.com
9	1.213852	192.168.157.11	192.168.157.18	SIP	398	Status: 200 OK

Figure 9 SIP signaling for registration with S-CSCF (IMS registration), Wireshark SIP traces.

Contents of a SIP message that is conveyed over a UDP message appear as in the Fig 9.

2.4.1 Session Initiation Protocol (SIP)

SIP signaling is used in most multimedia type communications. The use of SIP was adopted for VOLTE calls for this very specific reason. It is appropriate for the establishment of a voice/video call session over a packet core. SIP signaling is used for creation, maintenance and termination of communication sessions. The parameters defining the session are carried over SDP packets as discussed in the next section.

2.4.2 Session Description Protocol (SDP)

The SDP packet describes the various parameters that define the SIP session that is ongoing. All information required to describe the session like the Quality of Service, Media format, Activation time etc. are all part of this message.

2.4.3 SIP Header

The SIP header packet contains all “From” and “To” addresses at the various stack levels that is in SIP addressing scheme, UDP addressing scheme. UDP is represented as the Transport layer protocol used to send these SIP messages across the established data connection on the air interface.

```
⊟ Session Initiation Protocol
  ⊟ Status-Line: SIP/2.0 183 Session in Progress
      Status-Code: 183
      [Resent Packet: False]
  ⊟ Message Header
    ⊟ Via: SIP/2.0/UDP 192.168.157.18:60867;rport=60867;branch=z9hG4bKef47fd70385b4360bd07f23eafaa900a1b;transport=udp
        Transport: UDP
        Sent-by Address: 192.168.157.18
        Sent-by port: 60867
        RPort: 60867
        Branch: z9hG4bKef47fd70385b4360bd07f23eafaa900a1b
        transport=udp
    ⊟ From: <sip:0123456789@anritsu-cscf.com>;tag=1111111111
      ⊟ SIP from address: sip:0123456789@anritsu-cscf.com
          SIP from address User Part: 0123456789
          SIP from address Host Part: anritsu-cscf.com
        SIP tag: 1111111111
    ⊟ To: <sip:users1@anritsu-cscf.com>;tag=1396195325089
      ⊟ SIP to address: sip:users1@anritsu-cscf.com
          SIP to address User Part: users1
          SIP to address Host Part: anritsu-cscf.com
        SIP tag: 1396195325089
    ⊟ Contact: <sip:users1@192.168.157.11:57436;transport=udp>
      ⊟ Contact-URI: sip:users1@192.168.157.11:57436;transport=udp
          Contactt-URI User Part: users1
          Contact-URI Host Part: 192.168.157.11
          Contact-URI Host Port: 57436
        Contact parameter: transport=udp>
    Call-ID: 37fa9d9b85ac42c6ba84a07581af75df
```

Figure 10 SIP Packet Header

2.4.4 Session Description Protocol (SDP) Packet

Contents of a SDP message that is conveys the complete parameters describing the session established or ongoing appear as in Fig 11. The captured SDP message is from the actual VOLTE call simulation performed in the lab.

```
⊟ Message Body
  ⊟ Session Description Protocol
      Session Description Protocol Version (v): 0
    ⊟ Owner/Creator, Session Id (o): doubango 1983 678902 IN IP4 192.168.157.11
        Owner Username: doubango
        Session ID: 1983
        Session Version: 678902
        Owner Network Type: IN
        Owner Address Type: IP4
        Owner Address: 192.168.157.11
      Session Name (s): -
    ⊟ Connection Information (c): IN IP4 192.168.157.11
        Connection Network Type: IN
        Connection Address Type: IP4
        Connection Address: 192.168.157.11
    ⊟ Time Description, active time (t): 0 0
        Session Start Time: 0
        Session Stop Time: 0
    ⊟ Media Description, name and address (m): audio 25176 RTP/AVP 8 0 101
        Media Type: audio
        Media Port: 25176
        Media Protocol: RTP/AVP
        Media Format: ITU-T G.711 PCMA
        Media Format: ITU-T G.711 PCMU
        Media Format: DynamicRTP-Type-101
```

Figure 11 SDP Message Contents.

Chapter 3
CS SERVICES IN LTE

3.1 Simultaneous Voice LTE

SVLTE is one of the simplest and the oldest way of incorporating voice services on a LTE device. The basic mechanism is to have a device architecture with separate rf sub systems for LTE and voice. In this case the device is registered simultaneously over the LTE and the legacy 2G/ 3G networks. The LTE registration is used for data services and the 2G/ 3G legacy networks are used for voice and CS services including SMS, fax etc. This solution is specific to the handset and has no need for any modifications on the network side. This prompted the initial implementation with the advent of LTE to be focused on this particular implementation. The main disadvantage of this implementation is its power inefficient design. As the battery has to power two rf subsystems simultaneously there is a larger drain on the battery. Again, these devices are more expensive due to the use of two rf subsystems. Though this is not an efficient design, due to the ease of deployment and because of its independence from changing any network configuration and since no new expensive software being required on the network side, this implementation was preferred in the initial deployment.

3.2 Over The Top solution

Another approach that was investigated in the initial days for CS support on an LTE device is the Over The Top (OTT) solution. In this the use of applications like Skype or Google Talk for providing CS services was explored. This approach never gained a lot of momentum due to the restrictions on voice quality and mobility. The issue that rises from this approach is mainly when in a mobility scenario, on transitioning from LTE to a 3G network and further to a 2G network due to weak signal conditions the sustainability/ quality of the OTT call is not assured. Also, the switching of RAT (Radio Access Technology) becomes difficult due to the presence of multiple rf subsystems. A voice call with a low quality will be unacceptable to a market that has good CS services with the legacy 2G and 3G networks.

3.3 Voice Over LTE via Generic Access

VoLGA or Voice over LTE via Generic Access is to use the GAN (Generic Access Network) to route the voice calls over LTE in compliance with the GAN specifications from 3GPP. The GAN specification specifies incorporating the IP networks in to the generic cellular networks by offering services over the IP network. There is a possibility of a seamless handover in to and from the cellular network to a Wireless LAN. GAN enabled devices on detecting the Wireless LAN create a secure connection to a network element called the GANC. The GANC makes the telecom core network look like a generic base station through which the communication can be routed. The communication hence uses the GSM/ UMTS protocols over this secure interface to establish, release and continue conversation. This interface can be extended over to an LTE network by replacing the Wi-Fi access part with the LTE network. Hence the IP connection will be made on the LTE network though the general call establishment and continuation will happen over this interface using the existing GSM/ UMTS protocols. This will be possible because of the GANC interface which will interface with the core network, so that the device observes it as a base station.

3.4 Circuit Switched Fall Back

CSFB or Circuit Switched Fall Back is another approach to provide the Circuit Switched services (CS services) on a LTE network. The main idea behind this approach is to have the device registered with the LTE network only but to be also known to the legacy CS core. This would enable the devices to incorporate only one rf sub system, which will be an improvement over the SVLTE approach. Hence the device registers with the LTE network on powering ON. Further the CS core is informed on the current positioning of the LTE device by using a tunneling mechanism. CSFB mechanism makes the device to carry out the CS services over the legacy 2G/ 3G network. The approach is to keep listening to the paging channel for any incoming call while being registered on the LTE network. On seeing and incoming page addressed to the device or in case the user initiates a CS service, the device immediately falls back on to the legacy 2G/ 3G networks to carry out the requested/ incoming service. The main challenge in this approach is to keep the circuit core informed about the current position of the device.

This is required so that the incoming paging requests can be sent to the correct device location. The registration on both E-UTRAN and GERAN/ UTRAN (legacy) networks is required at all times to provide this service.

3.4.1 CSFB Reference architecture

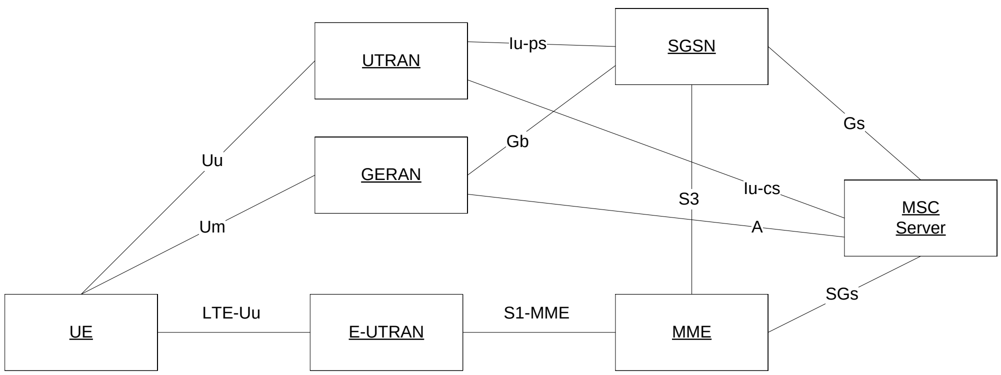

Figure 12 EPS architecture for CS fallback and SMS over SGs [3]

An interface 'SGs' as in Fig 11, has been added to the EPC in order to enable CS services in an EPS network. This interface is used not only for voice services but also to support SMS. SMS over SGs is another requirement that needs to be addressed to offer full range of CS services over a Packet only EPS network. The SGs interface as in the figure is used to interface the MSC Server with the MME. The MSC server is able to direct all CS paging originating in the GERAN or UTRAN domains and direct them towards the UE via the MME which has information of the UEs location and registration status.

3.4.1.1 Mobility Management Entity (MME)

Mobility Management Entity [7] of the EPS architecture makes sure that the UE is known in the CS domain using the registered Tracking Area Identity and mapping this to a Location Area Identity with which the UEs location will be identified and tracked in the CS domain. This ensures that any incoming CS service knows where to look for and page for the UE to deliver the incoming call. MME also makes sure that the mapping is appropriate and that the TAI the UE is registered to does not cross over the LA the UE identifies itself with in the CS domain. This is done by making a list of TAIs that the UE can be in so that there is minimal chance of the UE being forced to fall back on to a different LAI than the LAI that has been derived and the UE is identified with. This makes sure that the rate of call drops is to a minimum and the coverage is appropriate. The MME preserves the interface and association with the MSC VLR entities for UEs that are combined attached that is registered via both IMSI and EPS attach to be known in both the PS and CS domains simultaneously.

3.5 Voice Over LTE (VOLTE)

VOLTE is delivering voice/ video calls and other circuit switched services over the all IP LTE core network without making use of Over The Top approaches like VOIP. VOLTE calls will be established over the LTE EPS bearers making sure that the end user receives a quality of service that will be able to sustain a good voice call experience. VOLTE also can be extended to provide enhancements to the circuit switched services currently available by adding video telephony and HD voice type of services. The VOLTE call is essentially setup with the help of the IMS core. Once the LTE registration is finished, the UE goes on to a registration with the IMS core using SIP signaling. Once the IMS registration is completed, all circuit switched services are established using SIP signaling over EPS bearers having negotiated Quality of Service.

Chapter 4
VOLTE

The main elements required to process a Voice Over LTE call are discussed in this chapter.

4.1 IP Multimedia Subsystem (IMS)

IMS [10] is a telecom framework to integrate IP related services into a Telecom Core network. The protocols used in the general internet and on the Telecom network are usually different. IMS was designed to be a convergence point for the wired/ wireless and any other existing access technology. IMS acts as a fixed-mobile convergence providing a flexible architecture supporting all types of access technologies. The foundation of IMS operations are based on the P-CSCF, ICSCF and the S-CSCF. Which are collectively called the CSCF or Call Session Control Function.

4.2 Call Session Control Function (CSCF)

Call Session Control Function has three different functional divisions namely Serving-CSCF, Proxy-CSCF and Interrogating-CSCF. CSCF is the SIP server that after being connected to routes and provides the subscribed services to the UE. During the initial connection establishment prior to any SIP signaling, the UE is assigned a specific P-CSCF. This P-CSCF is usually located in the VLR region of the UE. Whenever the UE attempts to establish a SIP registration with the IMS core, the UE firstly sends its request to the P-CSCF which then forwards the request to the I-CSCF. I-CSCF now searches for an appropriate S-CSCF that can be assigned to the UE as the UE’s S-CSCF. After the S-CSCF is located and finalized, the communication is handed over to the S-CSCF which then responds to the UE via the P-CSCF, usually with a challenge nonce “number once used”. When the UE receives this challenge string, it responds with a hashed Shared Secret Key which is present in its ISIM and also in the HLR. On receiving the hashed SSD, the S-CSCF further reaches out to the I-CSCF to extract the SSD in the HLR pertaining to the UE’s record there. On confirming the UE’s identity the S-CSCF allows the registration of the UE and an IPsec communication is established between the two entities via the P-CSCF.

Other than the mentioned CSCF there is also an Emergency-CSCF that has been developed to route emergency calls efficiently. The decision, whether a received request is an emergency call or not is made by the P-CSCF on receiving the initial request from the UE. On sighting an emergency call establishment attempt from the UE the P-CSCF quickly routes the call to the E-CSCF which then routes it to the closest Public Safety Access Point to get the call handled. Most commercial networks today handle the emergency calls separately via CSFB or SV-LTE mechanisms to route them via the legacy 2G/ 3G networks depending on the network policy. But the use of VOLTE to establish an emergency call is very interesting due to the extremely fast set up time and the fact that many of the lower frequency LTE frequency bands have better coverage due to lower attenuation.

4.3 P-CSCF Discovery

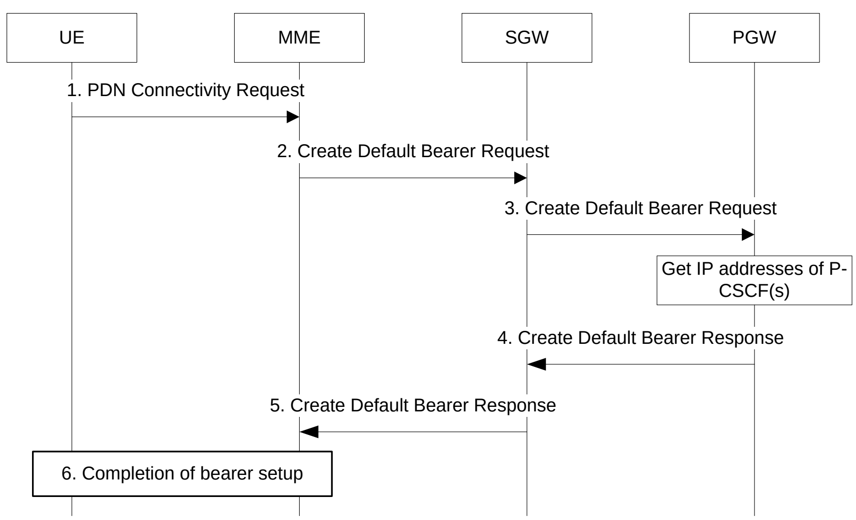

Figure 13 P-CSCF Discover process [4]

The P-CSCF discovery follows from the PDN connectivity request in the E-UTRA system by interrogating the PDN-Gateway for the P-CSCF IP address. The discovered address is later communicated to the UE in the bearer setup complete message. This P-

CSCF is used by the UE for all further IMS service requests and IMS communication. This procedure may be followed in the initial attach procedure itself and this makes the UE to be aware of the P-CSCF locally which further directs the requests to the UE's respective S-CSCF. The S-CSCF discovery is based on interrogating the I-CSCF as described earlier in this text.

4.4 IMS SIM

ISIM [5] is an application that is contained in the UICC card which allows the UE to perform IMS related procedures like identifying the UE to the IMS and being able to communicate with the IMS via SIP procedures, also making sure the authentication and security procedures with the IMS are taken care of. ISIM initialization on turning on the UE makes sure that the ISIM procedures are executed hence enabling the UE to be able to talk to the IMS and make the UE capable of using the services offered by the IMS.

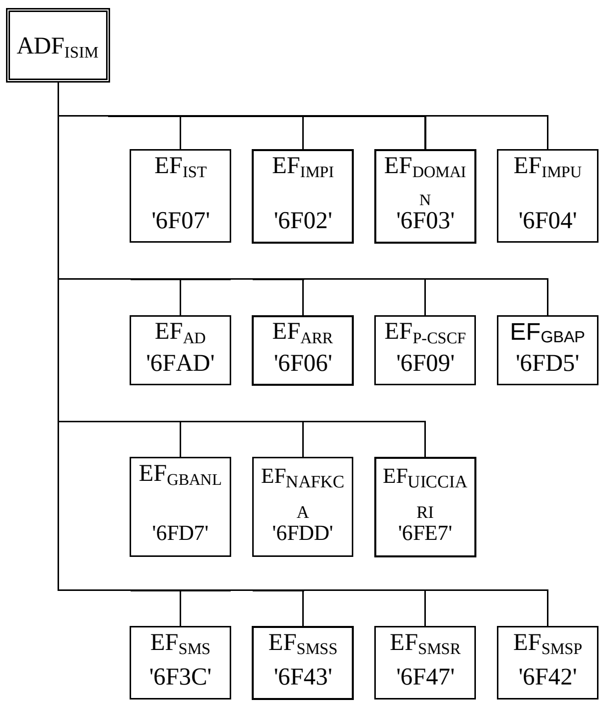

Figure 14 ISIM File structure [5]

4.4.1 ISIM initialization procedures

4.4.1.1 IMPI Request

The file EF_{IMPI} is IMS Private User Identity file, this is read from the ISIM application on the UICC. The EF_{IMPI} file contains the user's private identity with which the UE uniquely identifies itself with the IMS core.

4.4.1.2 IMPU Request

The file EF_{IMPU} is the IMS Public User Identity file, this is read from the ISIM application on the UICC. The EF_{IMPU} file contains the SIP identification of the UE. This is like the phone number in case of a SIM/ USIM application. The SIP identification is used by any other user to reach out to the UE in the public domain from another SIP related application. In case of Non-SIP application, the 'called' identification will be mapped in the IMS core to the IMPU of the UE so that the incoming service reached the device correctly.

4.4.1.3 SIP Domain Request

The file EFDOMAIN is the Home Network Domain Name file, this is read from the ISIM application on the UICC. The EFDOMAIN file contains the UE's home network related identification. The domain name of the UE's HPLMN is stored in this file.

4.4.1.4 ISIM Service Table request

This is the ISIM Service Table file, this is read from the ISIM application on the UICC. The EFIST contains the information regarding all the supported optional services. This enables the UE to understand the various services that it is subscribed to or not.

4.4.1.5 P-CSCF address request

The file EP-CSCF is read from the ISIM application on the UICC. This file contains the IPV4 IPV6 addresses of one or more P-CSCF. The priority of the various P-CSCF addresses existing on this file decrease from top to bottom in the file arrangement. These values are populated during the P-CSCF discovery procedure during the IMS registration process.

Chapter 5
TEST SETUP

5.1 Network Simulator configuration

Set Down Link Reference Power: -40dBm

Set Up Link Reference Power: -20dBm

Set EPRE: EPRE is the measure of the power contained in one Resource Element (RE). This power does not vary in case there exists, more than one RE under discussion or in case there are multiple RABs that we are talking about. This one kind of power measurement used. Further RS EPRE is the reference signal energy calculated per RE. This value is set to -67.8dBm/15KHz, considering the channel bandwidth to be 15KHz.

Set RSRP: RSRP is the average power of all the REs counted together inside one symbol of the reference signal. Now, RSRP value equals the EPRE value in case if there is no noise in the system. The value approaches the value of EPRE otherwise since it is an average of the power contained by the various REs inside a symbol.

Set Total Channel Power: This is another power representation. This is calculated as the sum of all the EPRE values inside one symbol. This value will differ from symbol to symbol due to their different channel combinations.

Set Packet Scheduling Mode: Packet scheduling mode was set to Static with Packet Rate as Best Effort and Transport Block Size pattern to Full Allocation.

5.2 Test Simulator Anritsu MD8475

Anritsu MD8475 signaling tester and base station simulator was used during to simulate the VOLTE call in the lab environment. MD8475 is a signaling tester which can simulate LTE FDD (2x2 MIMO), W-CDMA/HSPA/HSPA Evolution, GSM/(E)GPRS, CDMA2000® 1X/1xEV-DO Rev. A, TD-SCDMA/TS-HSPA. This simulator can be used for VOLTE tests as it has an IMS simulator with a built in working CSCF architecture.

5.3 Test Setup Diagram

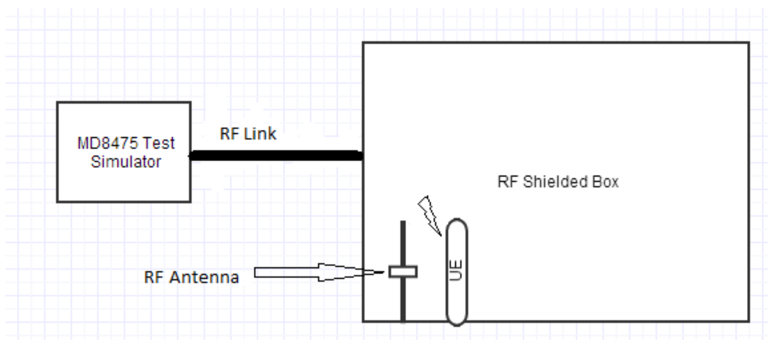

Figure 15 Test Setup Diagram

Test Setup consisted of a MD8475 Anritsu LTE signaling tester. This is connected via an rf cable with an rf shielded box. The rf shielded box houses an rf antenna, which creates a two-way connectivity with the UE under test. Once the setup is ready, the shielded box is closed to avoid any external commercial network interference. The radio of the MD8475 is switched ON or OFF modifying any parameters like the power level if required. On switching on the simulation, the radio starts broadcasting inside the rf shielded box and this allows the UE to attach to the simulated network.

5.4 IMS Droid App

IMS Droid is an open source app that simulates the IMS/ SIP stack along with other required ISIM parameters. Installation of this app enables the use of a regular LTE device as a VOLTE device. The SIP stack simulated by this app makes the device on which it is enabled to be registered directly to the S-CSCF server in the tester's IMS core. Once registered the app can receive calls from the Tester and make calls to the Tester via dialing the SIP identification instead of the phone number itself. From a signaling point of view the app suffices as there is no IP level information related to performance etc. that is being verified. The SIP signaling part and the protocol signaling during the initial registration phase stay unaffected. Furthermore the data connection established for the voice call with this app is not a different VOLTE bearer but on the same LTE bearer established. Additionally, the Quality of Service attained is often Best Effort. But regardless of these differences from an actual simulated VOLTE call, there is not effect on the pre-VOLTE LTE registration phase or the SIP signing in and signaling during voice call establishment that follows. This enables us to be able to simulate a VOLTE call on the MD8475 tester since the prime objective is to view signaling and not to measure performance.

Chapter 6

TEST RESULTS, OBSERVATIONS & PROPOSED TEST STRATEGY

6.1 SIP Signaling during a VOLTE Call on the MD8475 Simulator

24	50.299075	192.168.157.11	192.168.157.18	SIP	1028	Request: REGISTER sip:anritsu-cscf.com
27	50.304424	192.168.157.18	192.168.157.11	SIP	716	Status: 200 OK (1 bindings)
31	56.695952	192.168.157.18	192.168.157.11	SIP/SDF	1433	Request: INVITE sip:users1@192.168.157.11, with session description
32	56.717807	192.168.157.11	192.168.157.18	SIP	530	Status: 100 Trying (sent from the Transaction Layer)
33	56.778832	192.168.157.11	192.168.157.18	SIP	685	Status: 180 Ringing
34	56.786117	192.168.157.18	192.168.157.11	SIP	646	Request: PRACK sip:users1@192.168.157.11:47527;transport=udp
35	56.808809	192.168.157.11	192.168.157.18	SIP	563	Status: 200 OK
40	58.392726	192.168.157.11	192.168.157.18	SIP/SDF	1209	Status: 200 OK, with session description
41	58.410464	192.168.157.18	192.168.157.11	SIP	555	Request: ACK sip:users1@192.168.157.11:47527;transport=udp
396	62.376624	192.168.157.11	192.168.157.18	SIP	1003	Request: BYE sip:0123456789@192.168.157.18:55782;transport=udp
400	62.401672	192.168.157.18	192.168.157.11	SIP	528	Status: 200 OK
407	66.055444	192.168.157.11	192.168.157.18	SIP	880	Request: REGISTER sip:anritsu-cscf.com
408	66.058127	192.168.157.18	192.168.157.11	SIP	483	Status: 200 OK (0 bindings)

Figure 16 SIP signaling extracted from wireshark test logs

IMS Droid APP is used in this lab scenario to simulate the ISIM parameters and to enable a non-VOLTE UE to be able to talk to the IMS core and establish a SIP based VOLTE call. The scenario investigated is a VOLTE Mobile Terminated voice call. The initial registration with the IMS involves the first two messages observed in the wireshark traces above which are the REGISTER request to the anritsu-cscf.com which is the configured S-CSCF realm in the MD8475 simulator. Once the hashed request from the UE is matched and the UE is subsequently authorized correctly in the IMS core, the UE is successfully registered and recognized in the IMS core.

On attempting a Mobile terminated call from the simulator, we observe a subsequent establishment of a SIP based VOLTE voice call. SIP is an application layer protocol that can operate on TCP/ UDP as its underlying transport layer protocol. SIP is designed as a text based protocol that is independent of its underlying transport layer structure and can make use of the lower layer protocols as available. The log segment above is for a Mobile Terminated VOLTE call. The SIP signaling in case of a UE-IMS interaction during a MT VOLTE call setup begins an incoming INVITE message from the IMS core to the UE. The media identification and negotiation of the session parameters is done with the help of SDP. The parameters are described as the session profile or in general the governing parameters that define and establish the communication session. All the parameters to define the bandwidth, QOS, type of media (audio in this case), port number to be used for communication and all other defining parameters are described in the SDP message and these values are negotiated/ configured using this message. Further the UE sends a 100 Trying message and Ringing messages before actually connecting the VOLTE call. The UE responds with a Provisional Acknowledgement (PRACK) after the call gets connected. This Acknowledgement is forwarded to the end UE that actually originated the call. In this case the Simulated UE inside the MD8475 Simulator.

6.2 Multiple VOLTE Calls

Back to back multiple VOLTE calls were made to observe the average setup times which will provide inputs on how quickly the concurrency scenarios need to be developed in order to test the stability of the VOLTE call setup. The call signaling of multiple calls is as in the Fig below. Also the average setup time in different signal strengths is in the table below.

No.	Time	Source	Destination	Protocol	Length	Info
37	31.566486	192.168.157.18	192.168.157.11	SIP	673	Request: PRACK sip:users1@192.168.157.11:57436;transport=udp
39	31.586055	192.168.157.11	192.168.157.18	SIP	567	Status: 200 OK
40	31.587055	192.168.157.11	192.168.157.18	SIP	712	Status: 180 Ringing
41	31.614626	192.168.157.18	192.168.157.11	SIP/SDP	1308	Request: UPDATE sip:users1@192.168.157.11:57436;transport=udp, with session descripti
42	32.087072	192.168.157.11	192.168.157.18	SIP	712	Status: 180 Ringing
45	33.087042	192.168.157.11	192.168.157.18	SIP	712	Status: 180 Ringing
46	33.618743	192.168.157.18	192.168.157.11	SIP/SDP	1308	Request: UPDATE sip:users1@192.168.157.11:57436;transport=udp, with session descripti
47	33.701986	192.168.157.11	192.168.157.18	SIP/SDP	1061	Status: 200 OK, with session description
48	33.727353	192.168.157.18	192.168.157.11	SIP	559	Request: ACK sip:users1@192.168.157.11:57436;transport=udp
370	37.561400	192.168.157.18	192.168.157.11	SIP	598	Request: BYE sip:users1@192.168.157.11:57436;transport=udp
374	37.575899	192.168.157.11	192.168.157.18	SIP	565	Status: 200 OK
375	37.630046	192.168.157.18	192.168.157.11	SIP/SDP	1308	Request: UPDATE sip:users1@192.168.157.11:57436;transport=udp, with session descripti
376	39.701787	192.168.157.18	192.168.157.11	SIP/SDP	1432	Request: INVITE sip:users1@192.168.157.11, with session description
377	39.726855	192.168.157.11	192.168.157.18	SIP	529	Status: 100 Trying (sent from the Transaction Layer)
378	39.726862	192.168.157.11	192.168.157.18	SIP/SDP	1134	Status: 183 Session in Progress, with session description
379	39.738702	192.168.157.18	192.168.157.11	SIP	670	Request: PRACK sip:users1@192.168.157.11:57436;transport=udp
380	39.754847	192.168.157.11	192.168.157.18	SIP	566	Status: 200 OK
381	39.754851	192.168.157.11	192.168.157.18	SIP	710	Status: 180 Ringing
382	39.769999	192.168.157.18	192.168.157.11	SIP/SDP	1307	Request: UPDATE sip:users1@192.168.157.11:57436;transport=udp, with session descripti

Figure 17 SIP signaling extracted from wireshark test logs for multiple VOLTE calls

6.3 Observed Call setup times

S.No	Call Setup Time (seconds)	DL/UL Reference Power (dBm)
1	0.08	-40/-20
2	0.06	-40/-20
3	0.08	-40/-20
4	0.07	-40/-20
5	0.1	-40/-20
6	0.07	-40/-20
7	0.08	-40/-20
8	0.09	-40/-20
9	0.08	-40/-20
10	0.08	-40/-20

Table 1 VOLTE Call setup time in seconds

It was observed that the average call setup time with respect to the 10 call data collected in the table above is around 0.079 seconds. This gives insight into test scenario development while attempting to create concurrency test scenarios.

6.4 Observed Corner scenarios

The following observations were made in the process of this study and after viewing the established VOLTE call logs. Each observation is related to an area has to be included in the test strategy design to verify the robustness of the UE.

1) Supplementary Services: Supplementary services like call barring call, forwarding etc. are conventionally sent over the CS connection but in a VOLTE architecture their transmission and reception becomes an area of uncertainty, due to a SIP based signaling and the call being established on top of the PS network.

2) Interoperability with legacy networks: Signaling uncertainties creep up while moving from EUTRA coverage to GERAN/ UTRAN coverage area while on an established VOLTE call. This raises concerns regarding call retention and call quality. Also, on moving from just an IMS supported EUTRA coverage area to an area with no IMS support raises the same concerns regarding call retention and quality. The reason being the no IMS support area will force the UE to Fall back on the GERAN/ UTRAN network to maintain the CS service.

3) Emergency calls: One of the most affected areas will be that of emergency calls. Emergency calls are currently not supported via VOLTE on any of the live networks, and emergency calls are still routed through the legacy networks. Emergency call establishment in some cases may be routed particularly via the GERAN or UTRAN network in which case there may be delays due to two routing procedures, firstly from EUTRA to GERAN/ UTRAN and then next from the incorrect legacy RAN to the correct one.

4) Concurrency scenarios: Concurrency scenarios with respect to emergency call establishment will be an area of prime concern. The reason being that there is a lot of signaling involved for the various IMS based services offered. Also many protocols are interworking in this case. Once an emergency call has been initiated, there is a very good chance of the emergency call getting dropped or the call just facing delays and quality issues.

5) SMS: Short Message Service will also be affected due to the absence of a CS realm with an IMS core network. SMS send receive may affect warning broadcast message services that will be a compromise on the existing standard of warning message services etc. on the CS realm.

6.5 Test Areas Proposed to verify the robustness of a VOLTE deployment

As per the observations made and after the study of technological aspects of VOLTE, the following Test scenarios were identified to be used to measure the stability and performance of a VOLTE service.

1) **IMS Registration:** Verification of the device successfully registering on to the IMS server by reaching out to the SIP server S-CSCF. The use of correct APN will be verified and a successful registration is the desired outcome.

2) **Supplementary Services:** Supplementary services like call forwarding, barring, conference calls are to be verified on a VOLTE call. All other USSD related tests can be performed to validate the VOLTE call performance and stability.

3) **Mobility Scenarios:** Mobility scenarios between EUTRAN and GERAN/ UTRAN networks while on an established call verifying the call quality and stability. Also mobility between LTE coverage areas with IMS support to areas without IMS support.

4) **Emergency calls:** Emergency call establishment, establishment time, call quality, stability to be verified in mobility scenarios, low observed signal strength area, concurrency scenarios, handover scenarios. Automatic redial mechanisms, back to back call stability and long duration calls to be verified.

5) **SMS:** SMS related test scenarios to be explored, due to the absence of the CS realm. Transmission and reception of SMS needs to be verified. Different classes of SMS and with different coding schemes also need to be verified.

Chapter 7
CONCLUSIONS

Advent of VOLTE in the market has created a lot of open issues regarding the quality of already stable circuit switched services over the air. CS services offered over VOLTE must be at least as good as the existing CS services. The proposed test strategy will enable the VOLTE devices to be tested in specific critical areas to ensure quality and dependability. The issues identified will eventually be instrumental in the successful stable roll out of the VOLTE enabled networks and devices. Multiple VOLTE calls were simulated to verify different signaling areas and identify critical area that needed additional validation.

7.1 Future Work

One of the main challenges will come in case of FDD-LTE and TDD-LTE handover agreements or roaming feature. This will be a new critical area if un-interrupted CS services are being rolled out with area from FDD-LTE to TDD-LTE and vice versa. With more emphasis on international roaming this may soon be a possibility on international borders. Further with the enabling of HD voice and video call services, many more critical areas will need to be verified. Critical Area ranging from signaling, mobility and concurrency scenarios to more user experience related areas like performance, usability will have to be tested.

VOLTE is the future of voice and video calls on the mobile networks as it ensures uniformity by avoiding multiple Random Access Technologies co-existing, due to its range, penetration among many other features. The main hurdle is the verification of the robustness and performance of a VOLTE deployment which can be achieved to some extent using the proposed test strategy. Further extension of the strategy will be done to cover more area as more and more features are planned to be added to the existing VOLTE network.

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APPENDIX A

ABBREVIATIONS

3GPP	3RD Generation Partnership Project
AS	Access Stratum
CDMA	Code Division Multiple Access
CS	Circuit Switched
CSFB	Circuit Switched Fall Back
E-CSCF	Emergency-Call Session Control function
EDGE	Enhanced Data rates for GSM Evolution
EGPRS	Enhanced General Packet Radio Service
eNodeB	Evolved Node B
EPC	Evolved Packet Core
EPRE	Energy Per Resource Element
EPS	Evolved Packet System
EUTRAN	Evolved UMTS Terrestrial Radio Access Network
GANC	Generic Access Network controller
GERAN	GSM EDGE Radio Access Network
GGSN	Gateway GPRS Support Node
GPRS	General Packet Radio Service
GSM	Global System for Mobile communication
HD	High Definition
HLR	Home Location Register
HSPA	High Speed Packet Access
HSS	Home Subscriber Server
I-CSCF	Interrogating-Call Session Control function
IMPI	IP Multimedia Private Identity
IMPU	IP Multimedia Public Identity
IMS	IP Multimedia Subsystem
IMSI	International Mobile Subscriber Identity
IP	Internet Protocol
ISIM	IP Multimedia Services Identity Module
LTE	Long Term Evolution

MAC	Medium Access Control
MIMO	Multiple-Input Multiple-Output
MME	Mobility Management Entity
MSC	Mobile Switching Center
NAS	Non-Access Stratum
OTT	Over The Top
P-CSCF	Proxy-Call Session Control function
PDN	Packet Data Network
PS	Packet Switched
QoS	Quality of Service
RACH	Radom Access Channel
RAN	Radio Access Network
RF	Radio Frequency
RLC	Radio Link Control
RRC	Radio Resource Control
RSRP	Reference Signal Received Power
RSSI	Received Signal Strength Indication
SAE	System Architecture Evolution
S-CSCF	Serving-Call Session Control function
SDP	Session Description Protocol
SGSN	Serving GPRS support Node
SIP	Session Initiation Protocol
SMS	Short Message Service
SVLTE	Simultaneous Voice LTE
TAI	Tracking Area Identity
TCP	Transmission Control Protocol
TD-SCDMA	Time Division synchronous Code Division Multiple Access
UDP	User Datagram Protocol
UE	User Equipment
UICC	Universal Integrated Circuit Card

UMTS	Universal Mobile Telecommunication System
USSD	Unstructured Supplementary Service Data
UTRAN	UMTS Terrestrial Radio Access Network
VLR	Visitor Location Register
VOIP	Voice Over IP
VOLGA	Voice Over LTE via Generic Access
VOLTE	Voice Over LTE