



TAMPEREEN TEKNILLINEN YLIOPISTO  
TAMPERE UNIVERSITY OF TECHNOLOGY

**MERJA HAIKKA**  
**ANALYSIS AND TESTING OF LTE VOICE SOLUTIONS**

Master's Thesis

Examiner: Professor Pekka Loula  
Examiner and topic approved by the  
Faculty Council of the Faculty of  
Computing and Electrical Engineer-  
ing on 15.08.2012

## ABSTRACT

**MERJA HAIKKA:** Analysis and Testing of LTE Voice Solutions

Tampere University of Technology

Master of Science Thesis, 51 pages

July 2015

Master's Degree Programme in Information Technology

Major: Telecommunication Technology

Examiner: Professor Pekka Loula

Keywords: Telecommunication, LTE, CSFB, VoLTE, SRVCC, IOT

The goal of this thesis is to assess the long term evolution (LTE) voice solutions and to analyse the performance testing and measurement results. The main focus is on circuit switched fallback (CSFB) and single voice call continuity (SRVCC). Thesis is divided into two parts, a literature study and the performance testing results. The theoretical approach is been supplemented by the real measurement results. This gives a more dimensional view for the theoretical part. The CSFB measurements are based on interoperability tests executed by Microsoft Mobile Oy and published measurement results from Qualcomm. The SRVCC results are based on published measurements from Ericsson.

The measurement results show that with LTE it is possible to gain similar service level as with current legacy network. Especially it is expected that with new enhancements performance figures will improve. Important is to maintain at least the same service level for subscriber as what they currently have. Even though it is essential to do testing in lab environment to gain information how UE and networks are working together. Still the real results are received by measuring the live networks. There are already recommendations available for how to optimize for example network configuration to decrease the CSFB call setup delay.

The study indicates the significant changes that are ongoing and will come in mobile telecommunication. It describes the LTE voice solution techniques and features that are used to handle the call continuity when user equipment (UE) is moving between 2G/3G and LTE networks. It outlines the importance of usability and importance of nationwide common rules as standards.

## TIIVISTELMÄ

**MERJA HAIKKA:** LTE puheratkaisujen analyysi ja testaaminen

Tampereen teknillinen yliopisto

Diplomityö, 51 sivua

Heinäkuu 2015

Tietotekniikan diplomi-insinöörin tutkinto-ohjelma

Pääaine: Tietoliikenneteknologia

Tarkastaja: professori Pekka Loula

Avainsanat: Tietoliikenne, mobiiliverkot, LTE, VoLTE CSFB, SRVCC, IOT

Tässä opinnäytetyössä käsitellään LTE:n CSFB- ja VoLTE-puheratkaisuja. Näillä puheratkaisuilla taataan puheluiden toimivuus piiriyhteyksien ja LTE-verkon välillä. Pääpaino työssä on esitellä puheratkaisuihin sekä suorituskykyyn liittyviä asioita teoriaosuudessa sekä julkaistuja mittaustuloksia. Työn alussa käydään läpi matkapuhelinteknologioiden kehittyminen.

CSFB Suorituskykymittaustulokset perustuvat Microsoft Mobile Oy:n suorittamiin IOT-mittauksiin sekä Qualcommin julkaisemiin materiaaleihin. SRVCC suorituskykymittaustulokset ovat Ericssonin julkaisemia lukuja.

Käyttäjien näkökulmasta on tärkeää, että palvelun taso ei heikkene uusien teknologioiden ja ratkaisujen käyttöönoton myötä. Mittaustulosten perusteella on mahdollista saavuttaa vähintään sama palvelutaso kuin mitä on 2G- ja 3G-verkoissakin. Todelliset suorituskykymittaukset saadaan tuotantoverkosta jossa on paljon käyttäjiä ja eri tietoliikenneverkkoja. Tänä päivänä onkin jo olemassa CSFB-mittauksista tuloksia, jonka pohjalta on olemassa suosituksia miten esim. CSFB-puhelun syntymiseen liittyvää viivettä voidaan lyhentää verkon konfiguroinnilla. Työssä tuodaan esille myös maailmanlaajuisen yhteisten standardien tärkeys.

## **PREFACE**

This work was done for Microsoft Mobile Oy company.

I would like to thank my instructor from Microsoft Mobile Oy Mr. Markku Tarkiainen and my supervisor professor Mr. Pekka Loula for their support and excellent proficiency with this work.

Oulu 29<sup>h</sup> July 2015

Merja Haikka

## CONTENTS

1	INTRODUCTION .....	1
2	LONG TERM EVOLUTION (LTE) .....	6
	2.1 LTE Releases .....	8
	2.2 LTE coverage .....	9
3	LTE VOICE SOLUTIONS .....	11
	3.1 LTE Voice Evolution .....	11
	3.2 Circuit Switched Fallback (CSFB).....	13
	3.2.1 CS Fallback network architecture.....	13
	3.2.2 CSFB voice call procedures.....	15
	3.2.3 Network acquisition mechanisms .....	19
	3.2.4 Emergency calls.....	20
	3.3 Dual Radio .....	21
	3.4 VoLTE.....	22
	3.4.1 VoLTE system architecture .....	23
	3.4.2 SRVCC .....	24
	3.4.3 UE return back to LTE.....	26
4	INTEROPERABILITY TESTING (IOT).....	28
	4.1.1 IOT process.....	28
5	PERFORMANCE AND MEASUREMENT .....	30
	5.1 CSFB .....	30
	5.1.1 Data interruption time .....	32
	5.1.2 Mobile originated call setup time .....	33
	5.1.3 Mobile terminated call setup time .....	36
	5.1.4 Optimization for CSFB setup delays .....	37
	5.1.5 Call setup reliability.....	38
	5.2 SRVCC.....	39
	5.3 IOT .....	41
	5.3.1 Test environment .....	41
	5.3.2 Test results .....	42
6	CONCLUTIONS .....	45
	BIBLIOGRAPHY .....	48

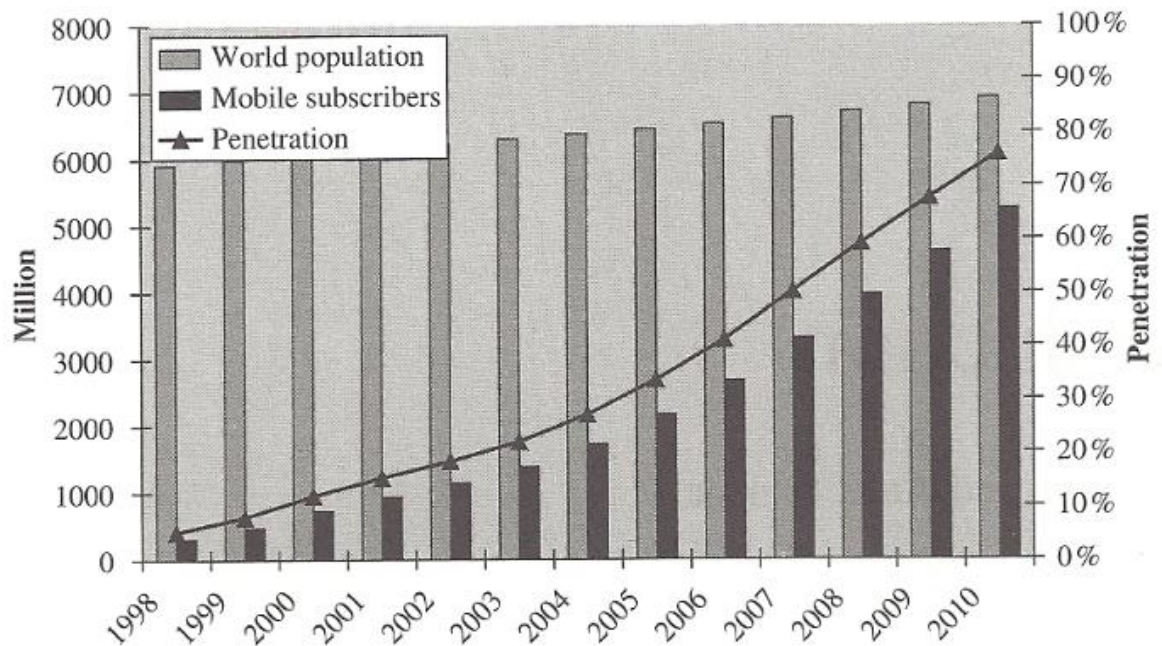
## LIST OF ABBREVIATIONS

2G	Second generation
3G	Third generation
4G	Fourth generation
3GPP	3 <sup>rd</sup> Generation Partnership Project
CDMA	Code Division Multiple Access
CN	Core Network
CS	Circuit-Switched
CSFB	CS Fallback
DL	Downlink
E2E	End to end
EDGE	Enhanced Data Rates for GSM Evolution
EPC	Evolved Packet Core
ESR	Extended Service Request
E-UTRAN	Evolved UTRAN
FDD	Frequency Division Duplex
GERAN	GSM/EDGE Radio Access Network
GPRS	General Packet Radio Service
GSM	Global System for Mobile Communications
HSPA	High Speed Packet Access
HSS	Home Subscriber Server
IMS	IP Multimedia Subsystem
IMT	International Mobile Telecommunication
IoE	Internet of Everything
IoT	Internet of Things
IRAT	Inter Radio Access Technology
ITU	International Telecommunication Union
IP	Internet Protocol
LAU	Location Area Update
LTE	Long Term Evolution
LTE-A	LTE Advanced (LTE R10 onwards)
M2M	Mobile to Mobile
MIMO	Multiple-input multiple-output
MME	Mobile Management Entity
MO	Mobile Originated Call, outgoing call
MSC	Mobile Switching Center
MSS	MSC Server
MT	Mobile Terminated Call, incoming call
NAS	Non-Access Stratum
OFDMA	Orthogonal Frequency Division Multiple Access

OTT	Over The Top
P-GW	PDN Gateway
PDP	Packet Data Protocol
PSTN	Public Switched Telephone Network
RAB	Radio Access Bearer
RCSE	Rich Communication Services Enhanced
rSRVCC	Reverse SRVCC
RAT	Radio Access Technology
RAU	Routing Area Update
RRC	Radio Resource Control
SAE	System Architecture Evolution
S-GW	Serving Gateway
SIB	System Information Block
SRVCC	Single Radio Voice Call Continuity
SVDO	Simultaneous Voice and Data Only
SVLT	Simultaneous Voice and LTE
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
UE	User Equipment
UMTS	Universal Mobile Telecommunication System
UTRAN	Universal Terrestrial Radio Access Network (UTRAN)
VLR	Visited Location Register
VoLTE	Voice over LTE

# 1 INTRODUCTION

The number of mobile subscribers for wireless network has grown significantly during the past 12 years. In 2002 the first billion subscribers were reached and five billion subscribers during year 2010. By 2011 the penetration of mobile phone reached 75% and network coverage was 90% globally. Figure 1.1 below illustrates the growth of mobile subscribers during years 1998-2010. These figures very clearly indicate that mobile communication is nowadays the preferred method for communicating, globally. The low prices in mobile device markets have been one of the enablers for this kind of development.

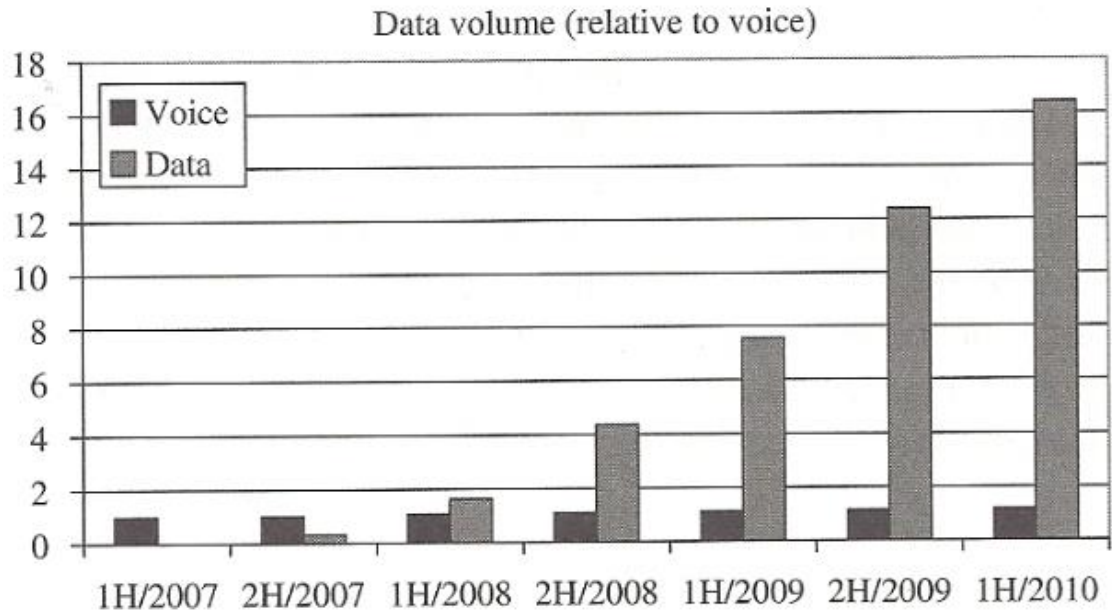


**Figure 1.1.** Growth of Mobile subscribers (Holma & Toskala 2011, p.2)

Taking the example of a subscriber who talks on the phone 300 minutes per month. With 12.2 kbps voice data rate the 300 minutes equals approximately 30 MB of data. This example subscriber has a friend who is a broadband user. The broadband user can easily use 1 GB of data per month. Sometimes user becomes a heavy broadband user by using 10-100 times more network capacity compared to the voice subscriber. This difference naturally sets high requirements for data network capacity and efficiency.

Developed techniques in telecommunication technology have impacted significantly in growing usage. One of these big technology changes happened in year 2007 when the

high speed downlink packet access (HSDPA) service was launched to the market. HSDPA in practice turned the mobile networks from voice-dominated to packet-data-dominated networks. Figure 1.2 illustrates the difference between data and voice based communication volumes in Finland (in 2007 the voice traffic is scaled to one). It can be seen that data traffic volume exceeded the voice volume during 2008 and by 2009 the data volume was already ten times bigger compared to the voice volume.

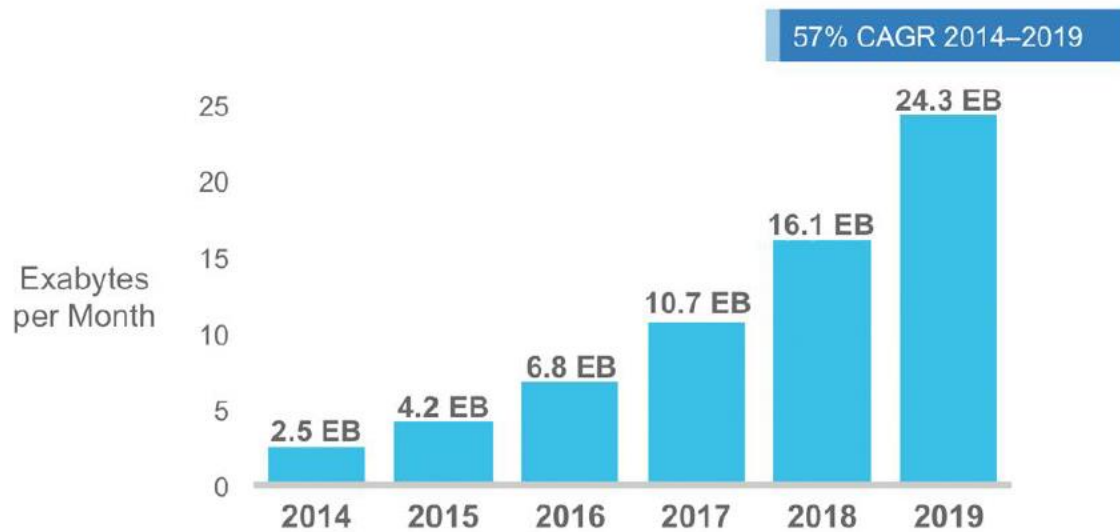


**Figure 1.2.** HSDPA data volume exceeds voice volume (Holma & Toskala 2011, p.2)

HSDPA is a modulation technique that is used to speed up the downlink communication, from network to user equipment (UE). HSDPA is used in universal mobile telecommunications system (UMTS) based 3G networks. UMTS is a third generation mobile phone technology and successor for global system for mobile communications (GSM).

In September 2013 there were approximately 6.6 million subscriptions worldwide. As many people may have multiple subscriptions the estimated real total subscription number is closer to 4.5 billion. Smartphone purchases are on the increase with an estimated 25-30% of mobile phone purchase now for smartphones. (Ericsson Mobility report 2013). Smartphones enable subscribers to use more applications e.g Facebook, Twitter and internet video streaming that require more network capacity and efficiency.

According to Cisco's forecast the mobile data traffic globally is expected to grow to 24.3 exabytes per month by year 2019. Additionally the Compound Annual Growth Rate (CAGR) for mobile data traffic is estimated to be 57% between 2014 and 2019. This forecast is presented in below Figure 1.3. (Cisco 2015)



**Figure 1.3.** Cisco Forecasts 24.3 Exabytes per Month of Mobile Data Traffic by 2019 (Cisco 2015, p.5)

From this thesis point of view the following analysis are interesting in Cisco's report. Smart devices represented 21% of all mobile devices and connections globally in 2013. 4G connections only represent 2.9% of mobile connections, but they already represent 30% of mobile data traffic. (Cisco 2014)

The concepts internet of everything (IoE) and internet of things (IoT) are being discussed more often. IoE means in conceptual level bringing together data, people, processes and things enabling for example new capabilities and business possibilities to rise. (Cisco, 2013)

The IoT means network of physical objects that are connected to internet. The type of object can be manufacturing floor, coffee machine or a car, for example, connected to internet with their own unique identifier. The digital identifiers enable bi-directional communication, such as remote control for the objects. When these mentioned concepts will be implemented it will naturally result in more data through the network. (Cisco, IoT).

When thinking about the huge increase in the mobile usage and especially the consumer requirements for higher and higher network capacity it is obvious that telecommunication technologies and networks need big changes in near future. The Long Term Evolution (LTE) and especially LTE-Advanced (LTE-A) have been developed to meet these growing demands.

Chapter two gives an overview of the LTE development during the past years. It will walk through the most important LTE releases that are relevant for the thesis. LTE services are

already deployed in many countries and are in commercial use. Figure 2.3 LTE Service Map illustrates this topic.

Chapter three discusses the LTE voice solutions. It will provide an over view of the three phases of LTE solutions, followed by a deeper dive into the world of voice solutions.

Circuit switched fallback (CSFB) enables voice calls and data traffic between the circuit switched (CS) and fourth generation (4G) long term evolution (LTE) networks.

The CSFB architecture is covered to give background information for the CSFB based incoming and outgoing voice call procedures. The network acquisition is also covered as this occurs whenever a UE switches from LTE to a 3G or 2G network. The acquisition can be done through a handover or re-direction procedure. These two procedures are also described more in detail in chapter three, including the differences between these two procedures.

Emergency calls are also briefly described in this chapter. It is critical to be able to establish an emergency call irrespective of which network a UE is located. Emergency call standardization has introducing demanding requirements of the network. An exception is emergency call establishment, where only the parts essential for this work are covered.

Dual radio solutions are covered in chapter three. A dual radio solution is where a UE has two radios always connected. One radio is used for packet data (LTE) and one for circuit switched voice calls (GSM, UMTS). There are pros and cons for this solution that will be highlighted in the chapter.

Voice over LTE (VoLTE) is IP Multimedia Subsystem (IMS) based VoIP, meaning that all voice and data traffic is IP based. IMS is the architectural framework that enables delivering the IP multimedia services for mobile users. The architecture including each of the three functional elements is described. This architecture brings several benefits as it is less complex compared to earlier architectural solutions, reduces latency and it increases data throughput.

One significant improvement that VoLTE brings is the high voice quality. During the network transfer the Single Radio Voice Call Continuity (SRVCC) is used to transfer an active voice call when a UE changes location from LTE to non-LTE network area. In real life the single radio voice call continuity happens for example, when subscriber is talking on the phone while moving. When the call was established the UE was located in LTE network but when UE moves, there is a signal measurement executed and in case the signal is not anymore strong enough in the LTE network the call is handed over to CS network and estill keeping the active voice call alive. One benefit of SRVCC is that it reduces the number of dropped voice calls.

Chapter five handles the CSFB and SRVCC performance. There are three important performance factors related to the CSFB that have direct impact on user experience. The subjects are handled based on the Qualcomm's published test results.

The performance of SRVCC plays a very important role as it keeps the calls active when a UE moves from a LTE to non-LTE network. There are two factors that impact the performance: the voice interruption time, and the call retention probability. The SRVCC performance is based on Ericsson's published test results.

Interoperability testing (IOT) will be described in chapter 4. Interoperability can have many meanings. In this work the IOT for mobile phones is covered.

LTE voice solution measurements and results are presented in chapter five. Firstly I will give a high level description of the test environment, finishing with the actual IOT measurement results. The measurements were executed in Microsoft Mobile Oy company's IOT lab.

Chapter six will draw final conclusions of the work. The different mobile technologies that are in everyday life often referenced as different generations 2G, 3G and 4G. These are also used in this thesis. GSM is referred to as 2G, Universal Mobile Telecommunication System (UMTS) as 3G and LTE as 4G.

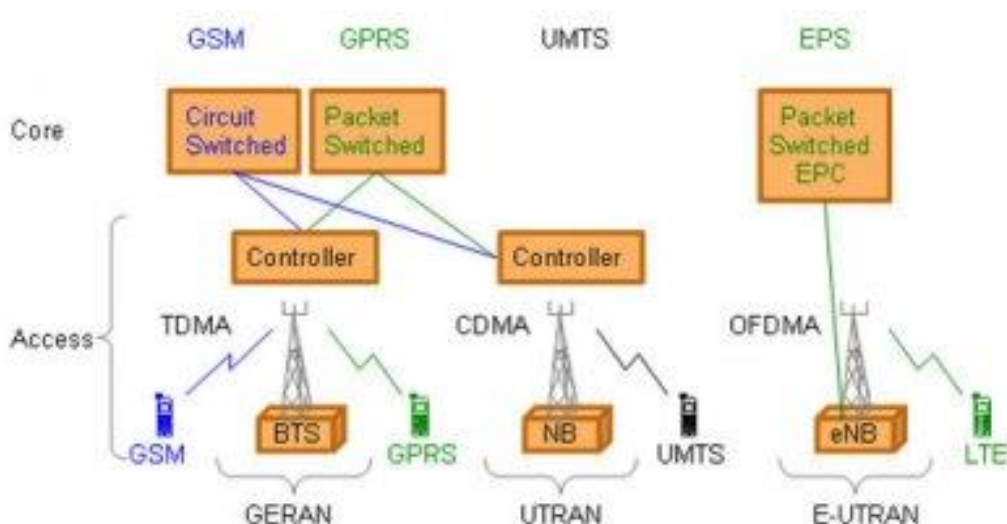
## 2 LONG TERM EVOLUTION (LTE)

Long term evolution (LTE) is a new radio technology that enables high data speed with low latency. Instead of Circuit Switched (CS) domain functions, LTE provides all services by using Internet Protocol (IP). In practice this means that voice calls and SMS services that have traditionally been provided over the circuit switched (CS) network, will be replaced by, for example Voice over IP (VoIP). Before this can happen, a new service control platform IP Multimedia Subsystem (IMS) needs to be deployed. IMS will be explained more in detail in chapter 3.

The long term evolution (LTE) standardization work is done by the International Telecommunication Union (ITU). Third Generation Partnership Project (3GPP) is the working group that develops the functionalities for the LTE. The 3GPP consists of six telecommunication standard bodies, known as organizational partners. 3GPP provides a stable environment for their members to produce the 3GPP specifications and reports. The detail list features and study items for each release can be found from 3GPP Internet page. (Wannstrom 2012; 3GPP)

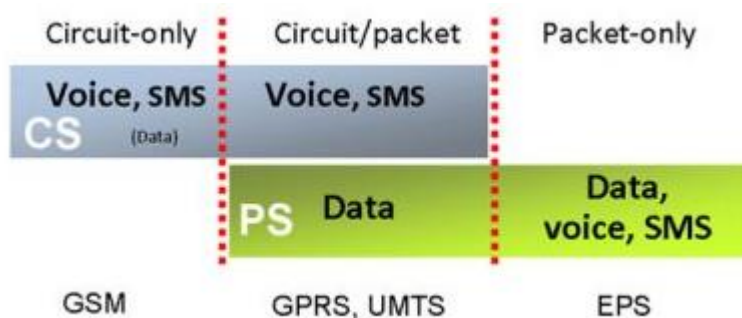
The work on LTE began already back in year 2004. LTE has been built on the universal mobile telecommunication system (UMTS). Figure 2.1 visualizes the development of network solutions from global systems for mobile communications (GSM) to LTE. On the bottom are mentioned the radio access networks GSM/EDGE radio access network (GERAN), universal terrestrial radio access network (UTRAN) and evolved UTRAN (E-UTRAN). LTE's access network is E-UTRAN. (Nohrborg)

Enhanced data rates for GSM evolution (EDGE) is a technique for packet switched communication for mobile phones. EDGE is based on GSM communication standard general packet radio service (GPRS). GSM/GPRS is 2G network, UMTS is 3G and LTE is 4G network. The radio access methods are different forms of multiplexing: time division multiple access (TDMA), code division multiple access (CDMA) and orthogonal frequency division multiple access (OFDMA).



**Figure 2.1.** Network solutions from GSM to LTE (Nohrborg)

Figure 2.2 supplements Figure 2.1 with a simple picture of circuit and packet domains. Here it can be clearly seen that in 2G network the circuit switched was used for voice calls and text messages. In 3G network the packet switched data transfers came parallel with the CS voice and text messages. The 4G network will transfer all traffic in packets.



**Figure 2.2.** Circuit and packet domains (Firmin)

The LTE is developed in phases called releases. More about releases in section 2.1. There are many factors that have been driving the development of Long Term Evolution (LTE). The obvious factors are the need for more wireless capacity but there are also other factors. For example the wireless technology development has to keep up with the wireline technology development to be able to ensure that applications also work fluently in the wireless domain. Also the competition from other wireless technologies is bringing the pressure for LTE development. As an example the IEEE 802.16 is providing wireless technology with high data rates. Similar alternative technologies are market competitors for the 3GPP technologies. (Wannstrom 2012; Holma & Toskala 2011)

The LTE is bringing many advantages. For example the data rate transfers will be faster, the traffic communication will be decreased and also more subscribers will use the same cell frequency that will increase the number of mobile subscribers.

However the new technology needs investments. The operators need to invest, for example, for the new antennas and subscribers need LTE supported UEs to be able to gain benefit of the new technology.

## 2.1 LTE Releases

A number of LTE releases have already been published. The LTE release 8 was frozen in December 2008 and is currently available on the market. The LTE-Advanced (LTE-A), also known as LTE release 10, will provide higher data rates in a cost-efficient way. LTE-A also fulfills the requirements set by the ITU-R for the IMT Advanced. (Wannstrom 2012)

Release 8 includes the following key features

- High spectral efficiency that is gained by using orthogonal frequency division multiplexing (OFDM) in downlink and DFTS-OFDM (“Single-Carrier in (frequency division multiple access) FDMA”) in uplink and by using multi-antenna application.
- Very low latency can be reached by short set-up and transfer delays. Also handover latency and interruption times are short.
- Support for bandwidths: 1.4, 3, 5, 10, 15 and 20 MHz.
- Simple protocol architecture compared to earlier solutions
- Simple architecture where eNodeB is the only E-UTRAN node, with fewer radio access network (RAN) interfaces.
- Backwards compatible with earlier 3GPP releases, including CSFB
- Interworking with other systems , for example cdma2000
- frequency division duplex (FDD) and time division duplex (TDD) within single radio access technology
- Self-Organizing Network (SON) operation
- CSG/HeNB (Home eNode B)

Release 9 has following small enhancements compared to release 8:

- HeNB enhancements
- SON enhancements
- Evolved-Multimedia Broadcast Multicast Service (E-MBMS)
- Location Service (LCS)
- Commercial Mobile Alert System (PWS/CMAS) (Takehiro 2010).

Release 10 is also known as LTE advanced (LTE-A). It includes especially improvements to increase the wireless capacity:

- To increase peak data rate, for downlink to 3 Gbps and for uplink to 1.5 Gbps.

- To gain higher spectral efficiency. In release 8 the maximum is 16bps/Hz and target is to gain 30 bps/Hz in release 10.
- To increase number of simultaneously active subscribers
- To improve performance at cell edges, for example for downlink (DL) 2x2 multiple-input multiple-output (MIMO) at least 2.40 bps/Hz/cell.

(Wannstrom 2012)

Release 11 includes one special new VoLTE feature, reversed single radio voice call continuity (rSRVCC), that is part of enhanced SRVCC. rSVCC feature is briefly covered in section 3.4.2.

## 2.2 LTE coverage

Figure 2.3, below, shows the LTE networks worldwide that are currently in service. This gives an overview of how widely LTE is already in use. It is important to notice that there are differences in the used frequency bands depending on the country and operators. This is also the reason why, as an example, the iPhone was compatible with 3G networks first in the USA, and then later in Europe.



**Figure 2.3.** *In Service LTE map (LTE Maps 2014)*

Network operators in Finland, and South Korea that provide LTE service can be seen in Table 1 below. South Korea is an interesting country to follow, as it is one of the leading countries where LTE advanced (LTE-A) and voice over LTE (VoLTE) is in use. VoLTE is described more in detail in the next chapter. (LTE Maps 2014)

**Table 1.** *LTE Service providers in Finland and South Korea (LTE Maps 2014)***Finland**

<b>Operator/Network Name</b>	<b>Deployment Status</b>	<b>LTE Band</b>
TeliaSonera	In Service Nov 2010	LTE-800, LTE-2600
Elisa	In Service Dec 2010	LTE-800, LTE-2600
DNA Finland Oy	In Service Dec 2011	LTE-800, LTE-2600

**South Korea**

<b>Operator/Network Name</b>	<b>Deployment Status</b>	<b>LTE Band</b>
LG Uplus	In Service July 2011, VoLTE Aug 2012, LTE-A July 2013	LTE -850
SK Telecom	In Service July 2011, VoLTE Aug 2012, LTE-A June 2013	
KT Corp	In Service Jan 2012	LTE-1800

According the LTE Maps information 30 network operators provide LTE services in USA in September 2014.

## 3 LTE VOICE SOLUTIONS

LTE is designed to be pure IP based communication. At present most of the voice networks are still GSM/WDCMA networks and voice calls are driving operators' revenue. However, the data tsunami is coming. This has largely been driven by the recent developments on the user equipment (UE) side and especially on the rapid software application development.

Skype calls have been in use already for some years, perceived to offer higher value to subscribers due to the reduced cost, but at the same time the operators are losing revenue. With voice over LTE (VoLTE) solution operators can get money from the calls and this explains why operators are driving to get VoLTE in use.

This chapter covers theory for LTE evolution, circuit switched fallback, VoLTE, single radio voice call continuity (SRVCC). Also smaller topics are covered that are seen important for this work. LTE evolution roadmap gives an overview for the changes that are needed. This also includes voice solutions circuit switched fallback (CSFB) that enables voice calls between LTE and legacy networks. The single radio voice call continuity (SRVCC) is the solution to transfer an active VoLTEe calls when UE moves from LTE to legacy network and to ensure voice call continuity. The CSFB voice call procedures are explained in detail. This is important as procedures will be referred in the performance chapter.

### 3.1 LTE Voice Evolution

The LTE evolution roadmap can be divided into three different phases as shown in below Figure 3.1. The first evolution phase was started in year 2011. The second phase of evolution was started couple of years later and third evolution phase is planned to be ready during year 2015.



**Figure 3.1.** The 3 phases of LTE voice evolution (Qualcomm 2013 p.2)

In the first phase of LTE evolution the data communication is handled by the LTE packet-switched (PS) networks if available. If the LTE PS network is not available, then data traffic is handled by the legacy networks. All voice traffic is handled by the circuit switched legacy networks. (Qualcomm 2013)

Dual and single radios are different options for how to handle the PS and CS traffic. In dual radio solutions the UE has two radios always enabled. One radio is used for packet data (LTE) and the other for circuit switched voice calls (GSM, UMTS). The dual radio solution will be covered more in detail on chapter 3.3. The circuit switched fallback (CSFB) functionality enables voice calls and data traffic between the 2G/3G circuit switched (CS) and LTE networks. With CSFB the UE in LTE network is switched to use 2G or 3G network instead of LTE. This way UE can utilize the CS voice calls while camping on LTE. However, SMS can be used still in LTE network. (Qualcomm 2013)

Second phase of LTE evolution includes LTE voice solution called VoLTE. It comes from words Voice over IP (VoIP) over LTE. VoLTE covers the IP Multimedia Subsystem (IMS), which is the concept for delivering IP multimedia for mobile users, and the multimedia telephony service (MMTel). MMTel is a standardized multimedia service suite. With IMS/MMTel it is possible to enrich voice services with video, or other such media content. Additionally voice services can be combined with other enhanced IP based services like HD voice and Rich Communication Suite (RCS). RCS enables activities such as instant messaging, sharing phonebooks, location and video. On the second phase the Single Radio Voice Call Continuity (SRVCC) functionality is used to maintain the call in progress while UE moves from LTE to non-LTE network coverage. (Qualcomm 2013; Poikselkä et al. 2012)

The target of the LTE evolution is that all network traffic is IP based. At this stage all communications, including voice and video, will be delivered over IP supported by RCS.

Different access methods and operators will be interoperable (Qualcomm 2013; Poikselkä et al. 2012).

## **3.2 Circuit Switched Fallback (CSFB)**

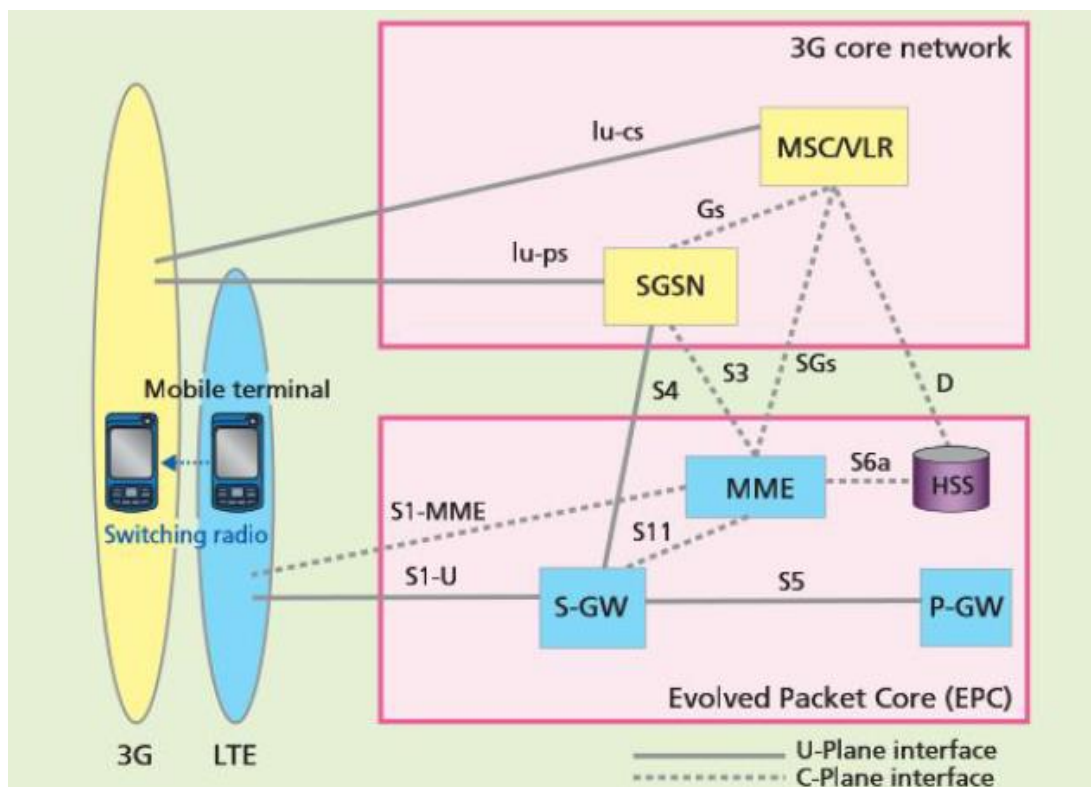
The transfer to LTE voice service network does not happen over a night. Meanwhile the networks are migrated there are different solutions to enable phone connections still working in the multi network environment. Circuit Switched Fallback (CSFB) solution is an intermediate solution that is used to enable voice calls and data traffic between the 3G Circuit Switched (CS) and LTE networks.

This work covers the CSFB solution in cases where call is established from mobile to land-lines (MO), from land-lines to mobile (MT) and from mobile to mobile (M2M). CSFB is used with single radio solutions. In single radio solution one radio is used to handle both packet switched (PS) and CS traffic. A special network signal is used to define which network is used at the time; PS or CS network (Qualcomm 2013).

CSFB mechanism to UMTS (third generation, 3G) and GSM (second generation, 2G) is described in this work. The CSFB mechanism description for other networks such as CDMA2000 and 1xRTT is excluded. The CSFB solution requires an upgrade to the operators network and also the UE needs to support this solution. (Nokia Siemens Networks 2012).

### **3.2.1 CS Fallback network architecture**

To support the CS fallback (CSFB) solution network changes and updates are needed. The UE also must support the CSFB. Figure 3.2, below, illustrates the most important network elements that are described in this chapter.



**Figure 3.2.** CS Fallback network architecture (Itsuma et al. 2009)

Mobility management entity (MME) is the main control element of the LTE core network. The LTE core network is called as evolved packet core (EPC). With the MME the LTE core network is connected to a mobile switching center (MSC) server and to visited location register (VLR). VLR is located in the 3G CS domain. The MME connection to MSC server enables the CSFB signaling and SMS transfer for the LTE devices. MSC server is also known as MSS. (Holma & Toskala 2011; Itsuma et al. 2009; Qualcomm 2013)

SGs is a new interface between mobility management entity (MME) and mobile switching center (MSC) server. The SGs reference point is similar to Gs reference point that changes signals between MSC and serving general packet radio service support node (SGSN). Messages in SG interface enable MME to register UE in 2G/3G network and provide CS based voice services for the UE. MME is serving UEs that are accessing from LTE network. In 2G/3G network SGSN is serving UEs for data services and MSC server is serving UEs for voice services. (Holma & Toskala 2011; Itsuma et al. 2009; Qualcomm 2013)

The SGs has important role as it is an interface to transfer the mobile terminating call request from CS domain to LTE. The combined mobility management between the 3G CS domain and LTE is also done by SGs. It is also possible to deliver CS pages and SMS via LTE access even if the UE is still connected to LTE network. (Holma & Toskala 2011; Itsuma et al. 2009; Qualcomm 2013)

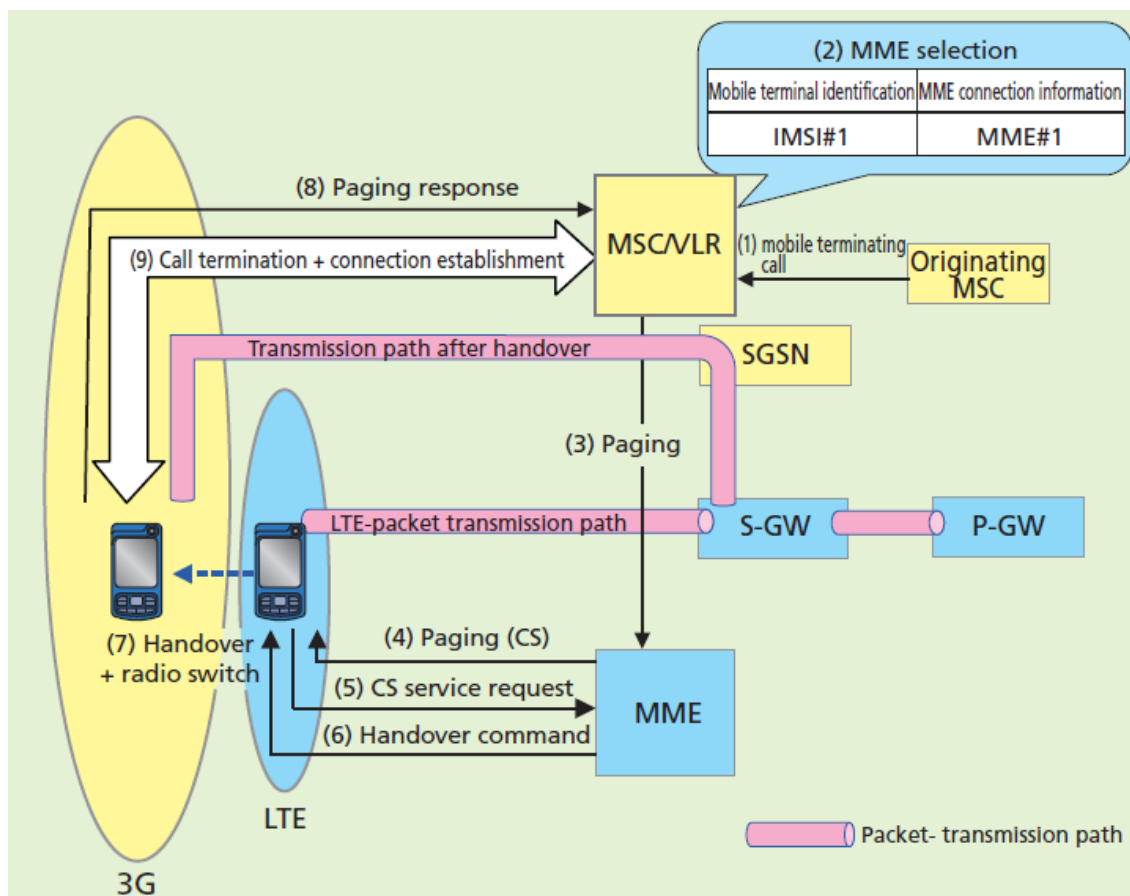
### 3.2.2 CSFB voice call procedures

The circuit switched fallback (CSFB) based incoming and outgoing voice call procedures are described in this chapter.

Figure 3.3 illustrates the communication exchange for establishing the mobile terminating (incoming) call. In this case the call is coming from legacy network to user equipment (UE) that is located in LTE.

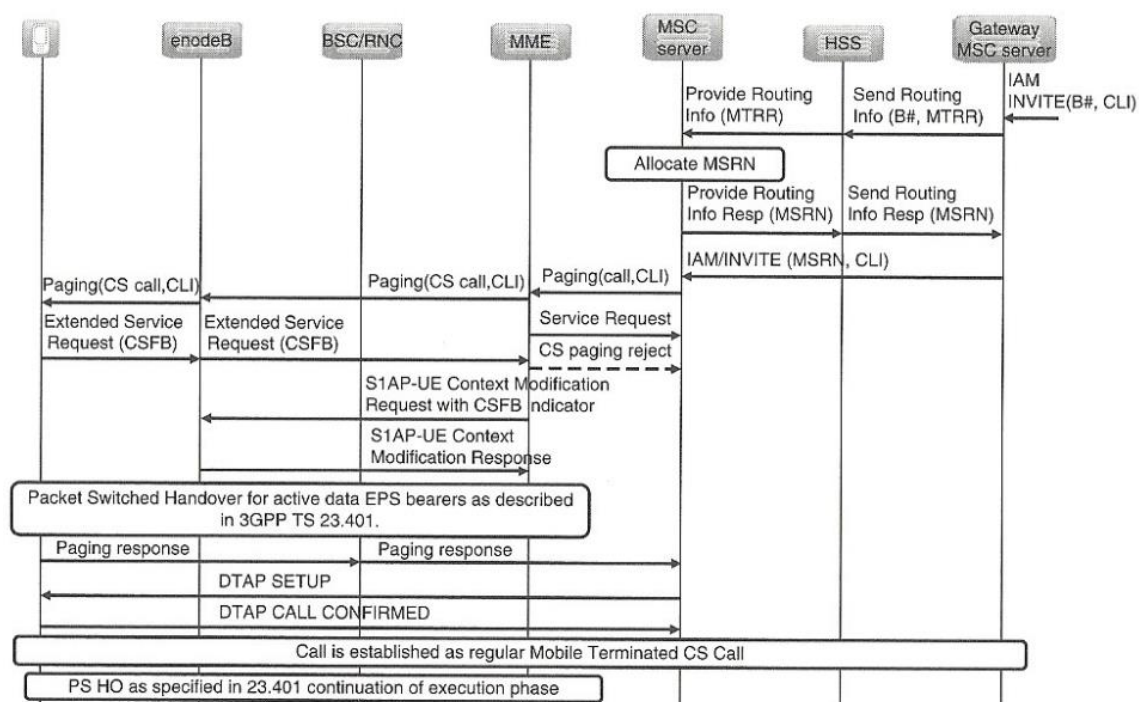
- (1) When a mobile terminating call is coming from the legacy network to the user equipment (UE), mobile switching center and visitor location register (MSC/VLR) receives an IAM/INVITE message indicating an incoming voice call.
- (2) The MSC/VLR identifies the mobility management entity (MME) from where the incoming call is initiated.
- (3) The MSC/VLR sends a paging message to the MME.
- (4) The MME sends a paging message to the UE that is located in the LTE network. The paging message includes information that a CS call is coming.
- (5) Now that coming voice call is CS the UE sends CSFB service request back to MME.
- (6) In evolved packet core (EPC) the packet communication transmission path (bearer) must be able to provide the always-on connection. This means that also the bearer has to be switched to 3G, as a result the MME sends a handover message back to the UE and initiates a handover procedure.
- (7) The UE switches its radio from LTE to 3G.
- (8) The UE sends a paging response message to the MSC/VLR. The terminating voice call can now be executed and CSFB is completed. All call control is handled on the 3G side during the voice call.

(Itsuma et al. 2009; Qualcomm 2013; Poikselkä et al. 2013)



**Figure 3.3.** Mobile terminating call (Itsuma et al. 2009, p.17)

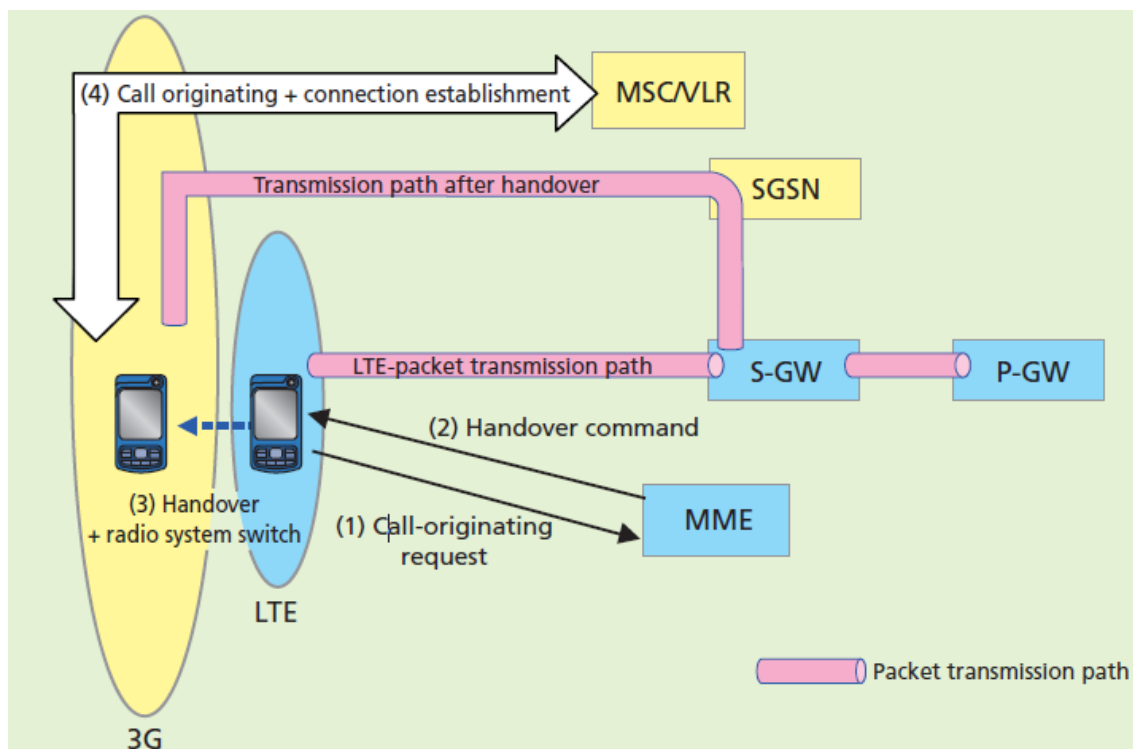
Establishing a voice call from legacy network to UE in LTE network requires several communication exchanges. Figure 3.4 is a summary of the communication. It shows the signaling and communication steps from beginning until the call is established and for example, in this case, the packet switched (PS) handover (HO) as the last step. This will be helpful also in coming performance chapter 4 where it will be presented more in detail how long time the call setup will take and what factors are influencing to the performance.



**Figure 3.4.** Signaling flow for mobile terminating call with CSFB (Poikselkä et al. 2012, p.180)

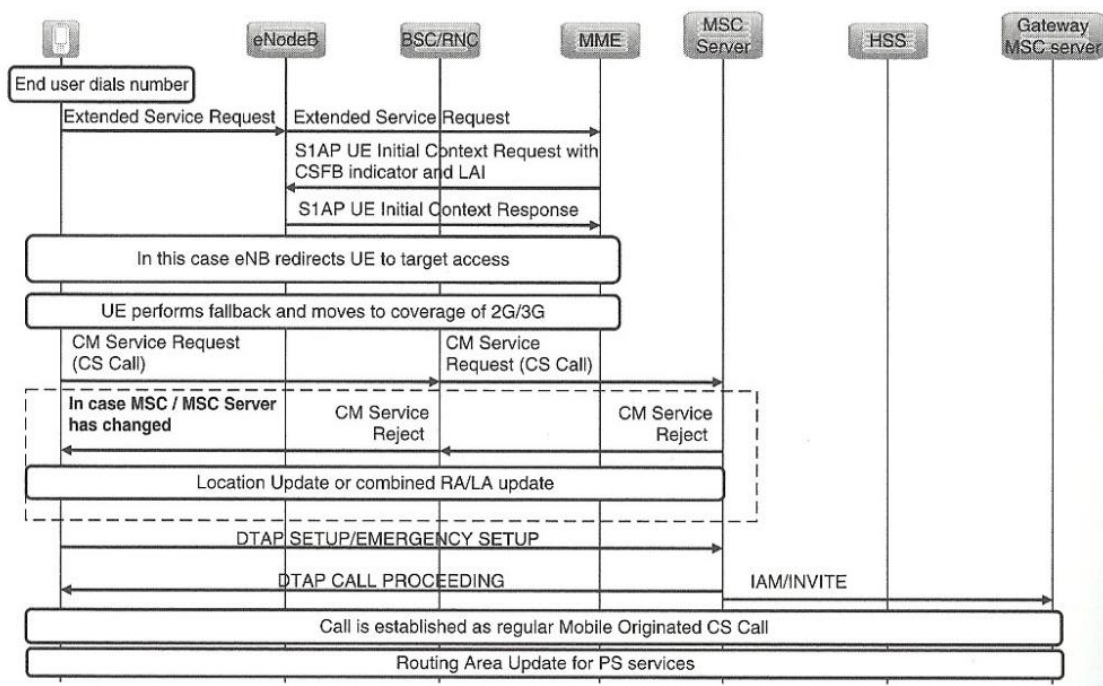
The mobile originating (outgoing) voice call is slightly simpler compared to the mobile terminating call. The UE that makes the voice call also needs to fall back from LTE network to 3G. The main difference is that the paging steps are not required in this scenario. The below Figure 3.5 describes in detail the steps for the mobile-originating voice call:

- (1) The UE, located in the LTE network, sends a call-originating request message to the MME. This message includes the CSFB service request. Also, in this case the packet-communication transmission path (bearer) has to be switched to the 3G side.
- (2) The MME sends a handover command to the UE and initiates a handover procedure.
- (3) The UE switches its radio from LTE to 3G and handover takes place.
- (4) As completion of the handover procedure the UE sends a call originating request to the MSC/VLR. The call is established and CSFB is completed.



**Figure 3.5.** Mobile originating call with CSFB (Itsuma et al. 2009, p.17)

Figure 3.6 is similar to Figure 3.4. Here the message exchange is described starting when UE dials the number. Here is described also the case when UE is moving and MSC server is changed. Then routing area (RA) and location area (LA) update is needed. This is also supportive figure for performance chapter 4 when discussing the mobile originated CSFB performance.

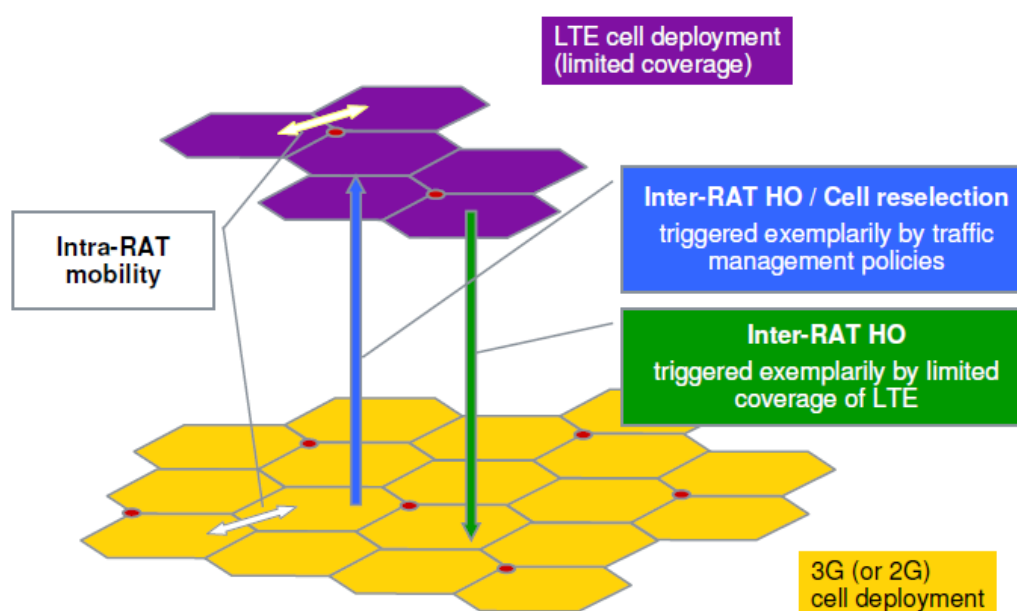


**Figure 3.6.** Signaling flow for mobile originating call with CSFB (Poikselkä et al. 2012, p.176)

### 3.2.3 Network acquisition mechanisms

When a UE switches from LTE to 3G or 2G network, acquisition of the voice network is required to take place. There are two different ways the acquisition and setup of the voice call, handover or redirection procedures can be performed.

With the handover procedure the target cell is prepared in advance and the UE can connect to the cell directly. Before the handover is performed, the inter-radio access technology (IRAT) measurement of the signal strength may be required. Figure 3.7 illustrates what is meant by the inter-RAT. It also describes the difference between inter- and intra-RAT.



**Figure 3.7.** Inter-RAT and Intra-RAT handover (Wegmann 2010, p.3)

The intra-RAT handover and cell change is performed between different cells within the same network. The inter-RAT handover and cell change is performed between the different networks. The inter-RAT from LTE to 2G/3G can be initiated by for example limited LTE network coverage. The traffic management policies are normally initiating the inter-RAT from LTE to 3G. (Wegmann 2010)

When a redirection procedure is used, only the signal frequency is dedicated to the UE and the UE can use any cell of the dedicated frequency. In case no cell is found from the dedicated frequency, then the UE can try other frequencies. With redirection procedure no IRAT measurement of signal strength is required prior the redirection.

The most interesting difference between the handover and redirection procedures is the requirement for signal strength measurement. The IRAT signal strength measurement is needed with handover procedure, but not with redirection. In practice this means that CSFB with redirection may take less time to identify the cell than compared to the handover procedure. The difference between the handover and the redirection procedures can be significant from the user experience point of view. This will be covered more in detail in the Performance chapter 4.

Redirection based CSFB has different call set-up speeds depending on which LTE release is used for the solution. Today the most commonly used solution is the System Information Block (SIB) skipping that is part of the release 8 redirection specification. There are 13 SIBs used in LTE. Each block carries specific information for the UE. Based on the information the UE is able to access a cell or to do a cell re-selection. For example SIB-1 includes cell access related parameters and scheduling of other SIBs.

Under SIB skipping the UE reads only the mandatory SIBs; 3, 5 and 7:

- SIB-3: Cell-reselection parameters for intra-frequency, inter-frequency and inter-RAT
- SIB-5: Cell-reselection parameters for inter-frequency neighboring cells (E-UTRA)
- SIB-7: Cell-reselection parameters inter-RAT frequency (GERAN cells)

All other SIBs are skipped prior to access. Measurement control messaging is used to deliver the SIB-1 after the UE is connected to the target cell.

Release 9 includes redirection SI tunneling. With this feature UE does not have to read any SIB on the target cell. The SIB information is tunneled through the Radio Access Network (RAN) and sent via the core network to the source RAN. SIB information is included in the redirection message that is sent to the UE. It is expected that release 9 redirection SI tunneling solution will bring improvements for the redirection call setup times as there is no need to read SIBs on the target cell. (Qualcomm 2013; Satpathy 2012)

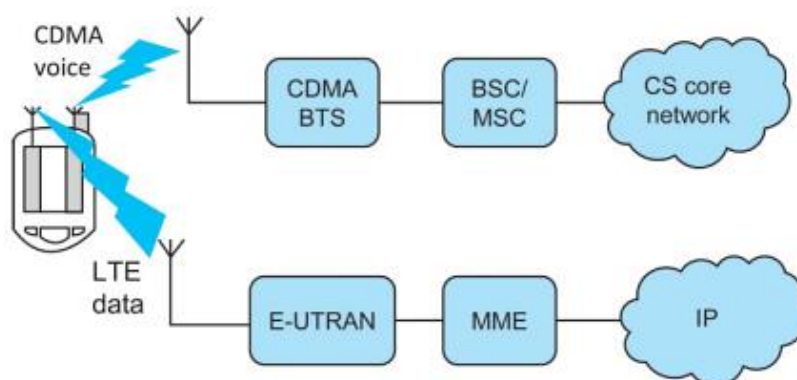
### **3.2.4 Emergency calls**

The CSFB procedure works also for emergency calls. In case a UE has MO emergency call in progress and CSFB needs to be performed, the UE informs the main control element (MME) of the LTE core network that CSFB request is needed for the emergency call. The MME informs E-UTRAN about the emergency call. (3GPP 2012)

The E-UTRAN will indicate the CSFB priority level to the RAT where a PS handover is initiated. This enables RAT to prepare the radio resources for example RAB resource priority allocation. Based on the emergency indication the E-UTRAN may also select certain radio access 2G or 3G network to handle the CS emergency call. (3GPP 2012)

### 3.3 Dual Radio

In dual radio solutions the UE has two radios always enabled. One radio is used for packet data (LTE) and the other for circuit switched voice calls (GSM, UMTS). Figure 3.8, below, illustrates the solution.



**Figure 3.8.** Dual radio solution (Moray 2013 p. 489)

The dual radio related technical standard and protocol is Simultaneous Voice and LTE (SVLTE), functioning only with 4G networks. The 3G network has a similar standard in use called Simultaneous voice and data only (SVDO) (Phonoscope SVLTE).

The role of SVLTE is important as 4G networks were initially designed for data use only. Without SVLTE it is not possible to use both data and voice networks simultaneously. Over time SVLTE will be replaced by VoLTE that enables the use of both data and voice seamlessly on the LTE network.

The dual radio solution has a number of benefits when compared to CSFB solution. First of all dual radios introduces no latency, where CSFB solutions do. Additionally the solution is very straight forward from the perspective of the network operator, as it does not require any changes to the network side; the only required changes are made by device manufacturer on the UE. Some downsides exist with dual radio solutions, including that it is more expensive, a more complex solution and incurs higher power consumption (Qunhui 2011).

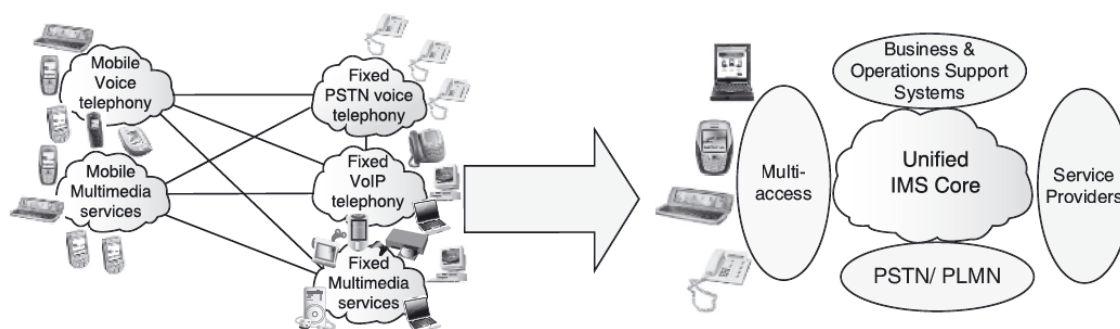
Currently at least the American network operator Verizon is using dual radio for CDMA and LTE with Qualcomm's chipsets (Klug 2011). According to Ericsson (Ericsson 2011 p.6) the dual radio solution for CDMA and LTE smartphones was launched in the US in 2010.

### 3.4 VoLTE

Over the top (OTT) Voice over IP (VoIP) applications, for example SKYPE have been available for many years. OTT applications are using the same network as other IP data traffic, which can lead to heavy data loads and compromise the OTT voice Quality of Service (QoS). Nowadays many subscribers are using OTT VoIP instead of CS. The operator's mobile broadband data services along with the wide coverage of LTE and HSPA networks are enabling the use of free OTT VoIP services. Other well-known OTT applications include Facebook, Whatsapp and GoogleTalk.

One of the significant improvements voice over LTE (VoLTE) brings is high quality voice. VoLTE is a native application on UE, enabling that voice traffic can be prioritized over the other data traffic and this way it is possible to ensure consistent quality of service for end users. VoLTE is based on two standards introduced by 3GPP; IP Multimedia Subsystems (IMS) architecture and Long Term Evolution (LTE). Session Initiation Protocol (SIP) is the enabling protocol.

When discussing about LTE it is important to understand meaning of IMS. As shown in Figure 3.9 the IMS is a unifying platform for all IP based communication. One example benefit from the network convergence is that all new services are accessed in a consistent manner. (Poikselkä et al. 2004)

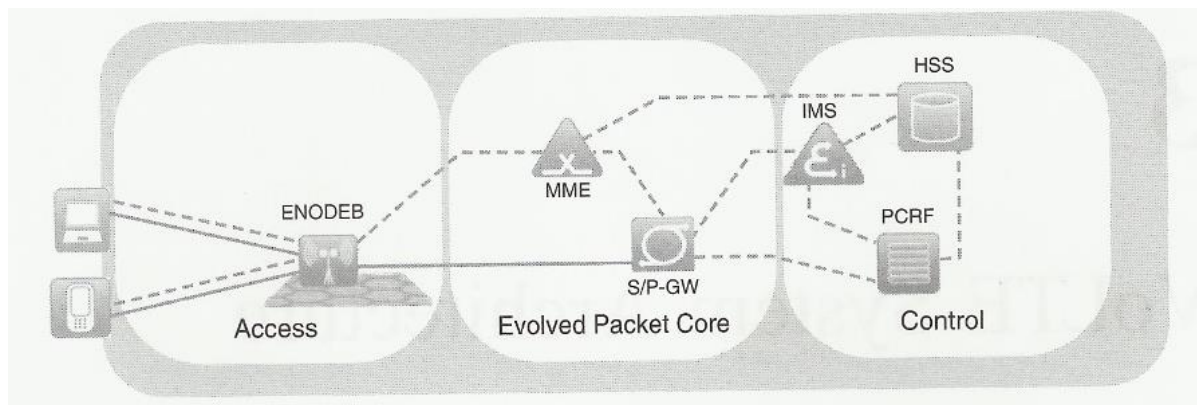


**Figure 3.9.** Convergence of network (Poikselkä et al. 2004, p. 6)

The work required to migrate from CS based voice to purely IMS architecture based VoIP will take a long time. The network operators are building the LTE networks in stages. During the network migration process, the most important thing is to ensure that voice calls are handed over to the CS network when LTE network coverage is insufficient. Single radio voice call continuity (SRVCC) is the solution that is used to guarantee the call continuity. (Poikselkä et al. 2012)

### 3.4.1 VoLTE system architecture

The VoLTE system architecture can be divided into three different functional elements: access, evolved packet core (EPC) and control, as mentioned in the below Figure 3.10.



**Figure 3.10.** VoLTE system architecture (Poikselkä et al. 2012 p.10)

The access domain includes only evolved NodeB (eNodeB) network element. eNodeB is connected to the network and directly communicates with the UE. It is the base station for the LTE radio. In earlier 3GPP architectures there have been two network elements defined, for example NodeB and radio network controller (RNC). They were defined for UMTS terrestrial radio access network (UTRAN). Simplified VoLTE network architecture brings benefit by reducing costs, complexity, latency and increasing the data throughput. (Holma & Toskala 2011; Poikselkä et al. 2012; Itsuma et al. 2009)

Evolved packet core (EPC) is the latest evolution of the 3GPP core network architecture. An EPC domain consists of three functional elements, the mobility management entity (MME), the serving gateway (S-GW) and the packet data network (PDN) gateway (P-GW). The MME is the main control element in the EPC. The EPC does not include any circuit switched domain and therefore has no connectivity to CS networks. The EPC enables seamless integrated usage of different radio access systems for example wireless LAN, 3G and LTE. EPC also provides Mobility Management among the mentioned systems. (Holma & Toskala 2011; Poikselkä et al. 2012; Itsuma et al. 2009)

The user plane uplink and downlink packets are going through the S-GW. The S-GW can also forward the packets during a handover procedure. The P-GW allocates IP addresses to UE and also provides an interface towards packet data networks (PDNs) like IMS and the Internet. The P-GW also includes the policy and charging enforcement function (PCEF). The role of the PCEF is to detect the service data flows, flow based charging and also policy enforcement, meaning the discarding of packets. (Holma & Toskala 2011; Poikselkä et al. 2012)

The control domain includes IMS, home subscriber server (HSS) and policy and charging rules function (PCRF). IMS has several elements which are needed to control the voice session. The HSS stores all subscriber and service related information of the IMS, EPC and access data. The PCRF is a bridge between the IMS and EPC domains. One of the PCRFs tasks is to setup the EPC bearer for a new voice session. (Holma & Toskala 2011; Poikselkä et al. 2012)

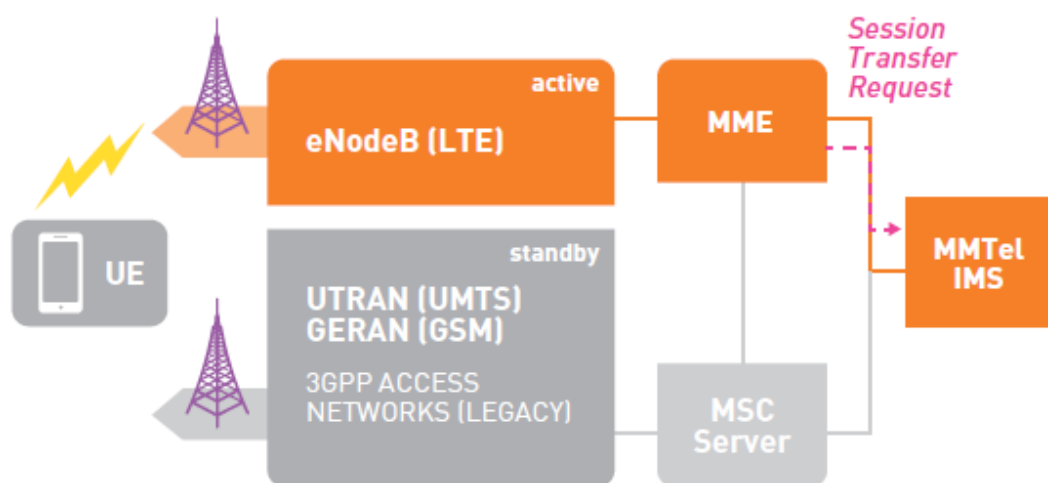
### 3.4.2 SRVCC

The operators are building the LTE networks gradually and meanwhile a solution is required to ensure ongoing call continuity between LTE and legacy networks. Single radio voice call continuity (SRVCC) solution transfers the active VoLTE calls from LTE to legacy networks. A pre-requisite for the SRVCC is that software updates need to be made to the MSS subsystem, IMS subsystem and the LTE/EPC subsystems. SRCVV is also known as packet switched-circuit switched (PS-CS) access domain transfer.

The call handover is performed when a UE on an active VoLTE call moves away from the LTE network coverage area and therefore handover to CS network is required. Signal strength measurement result indicates when the signal gets too weak and domain transfer is required. The handover includes two steps: IRAT handover and the session transfer. The IRAT handover follows the normal handover procedure from LTE radio access to WCDMA/GSM radio access. The session transfer is a new mechanism and it transfers the access control and voice media anchoring from the EPC to the legacy CS core.

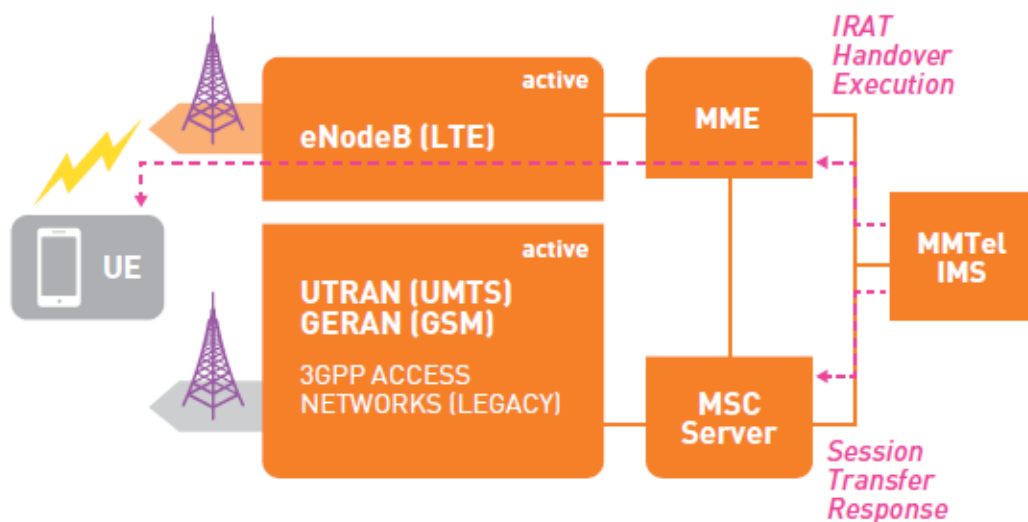
User control is retained by the MMTel/IMS during the whole handover process from LTE to 2G/3G. (Qualcomm 2012)

The handover is initiated by the session transfer request. As shown in the below Figure 3.11. The session transfer request is initiated from MME to MMTel/IMS.



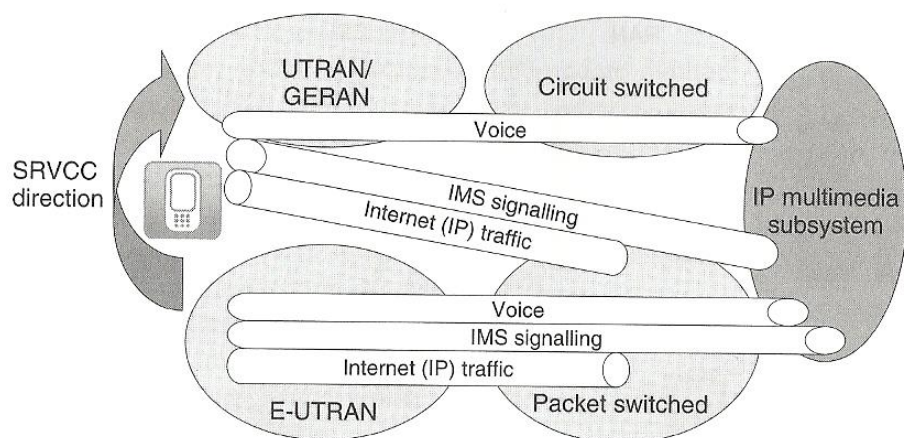
**Figure 3.11.** Session transfer request (Qualcomm 2012).

The MMTel/IMS responds to the session transfer request with two parallel commands as shown in the Figure 3.12. The IRAT handover execution command is sent to the LTE network through the MME to LTE RAN to instruct the UE to prepare to move the voice call to the CS network. The session transfer response command is sent to MSC server to CS network and it prepares the CS network to accept the active call. (Qualcomm 2012). After the commands have been executed, the MMTel/IMS and UE switch to CS network to continue the call in progress. (Qualcomm 2012; Ericsson 2012)



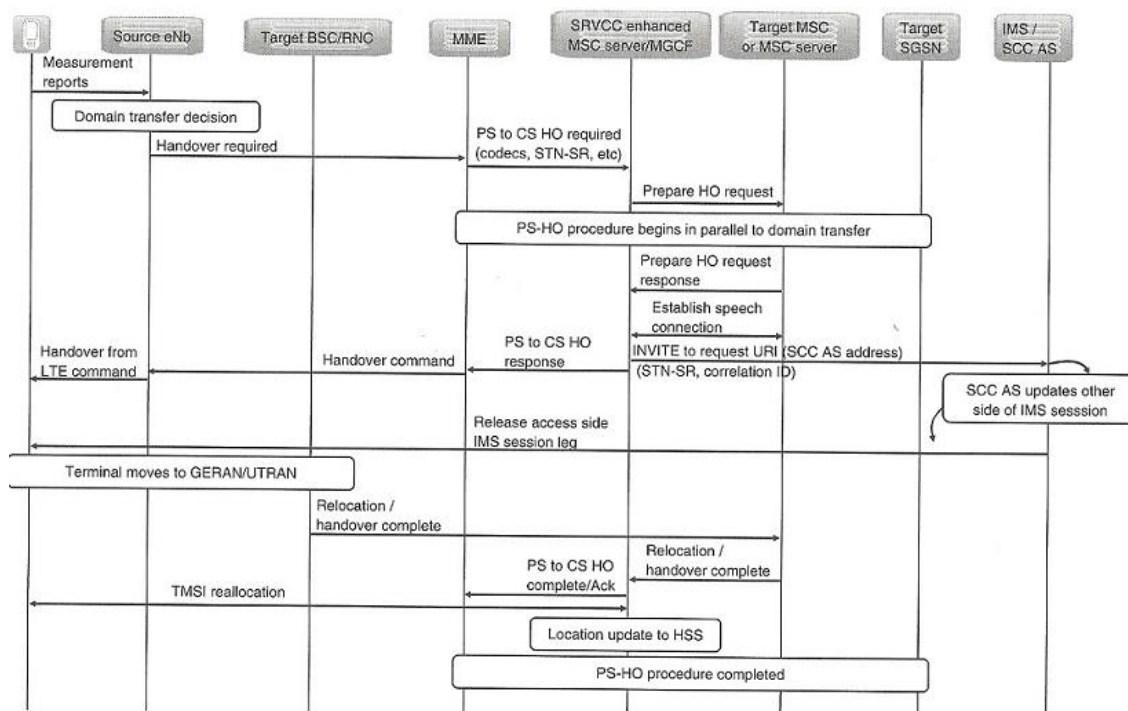
**Figure 3.12.** Simultaneous IRAT handover and session transfer commands (Qualcomm 2012).

As mentioned above, with SRVCC only the voice communication is moved from LTE to circuit switched side. All other communication is still remained in the LTE network. Figure 3.13 below illustrates the SRVCC procedure, where voice communication path is presented separately.



**Figure 3.13.** SRVCC principle in high level (Poikselkä et al. 2012, p146)

The SRVCC signaling flow is presented in Figure 3.14. It includes all required steps to transfer the active VoLTE call to legacy network. First thing that happens is the signal strength measurement and based on the report it is decided if the domain transfer is needed to be done. The signal flow will support in coming chapter 4 where SRVCC performance topic is covered.



**Figure 3.14.** SRVCC signaling flow (Poikselkä et al. 2012 p.150)

With the SRVCC solution it is possible to significantly reduce the number of dropped voice calls when UE is moving in and out of LTE network coverage. The SRVCC will also work for emergency calls. (Ericsson 2012; GSMA 2013)

When UE moves from non-LTE to LTE network the voice call continuity is enabled with reverse SRVCC (rSRVCC) functionality. The rSRVCC is included in the LTE release 11. From service continuity point of view the maximum interruption time for established voice call is 300 ms. (3GPP 2011)

### 3.4.3 UE return back to LTE

When voice call is completed, there are two options for how to guide UE to return to LTE PS access as soon as possible. These options require implementation on the legacy RAN. The first option is valid for the SRVCC. In this case legacy RAN network sends LTE system information to the UE and the UE will reselect the LTE cell. Alternatively the UE

can be redirected to connect to the LTE at the same time the connection is released for the earlier voice call. (Qualcomm 2012).

## 4 INTEROPERABILITY TESTING (IOT)

The interoperability can have different meanings depending of the context. Interoperability testing (IOT) is used to verify whether all different technical components, software and configurations are working together. In this work user equipment (UE) is tested against the network and vice versa. IOT de facto standard in cellular industry in order to avoid any product liability and quality issues.

There are two types of cellular testing scenarios, lab testing and field testing. In lab testing implementors, such as mobile phone vendors, have built a testing network to check how new UEs are working. In some cases the vendors are visiting operators test labs to execute the tests. The field testing is performed in a real commercial network where it is possible to cross check how UEs are working also with other type of UEs, and in multivendor network configurations. The main focus of the IOT is in lab testing.

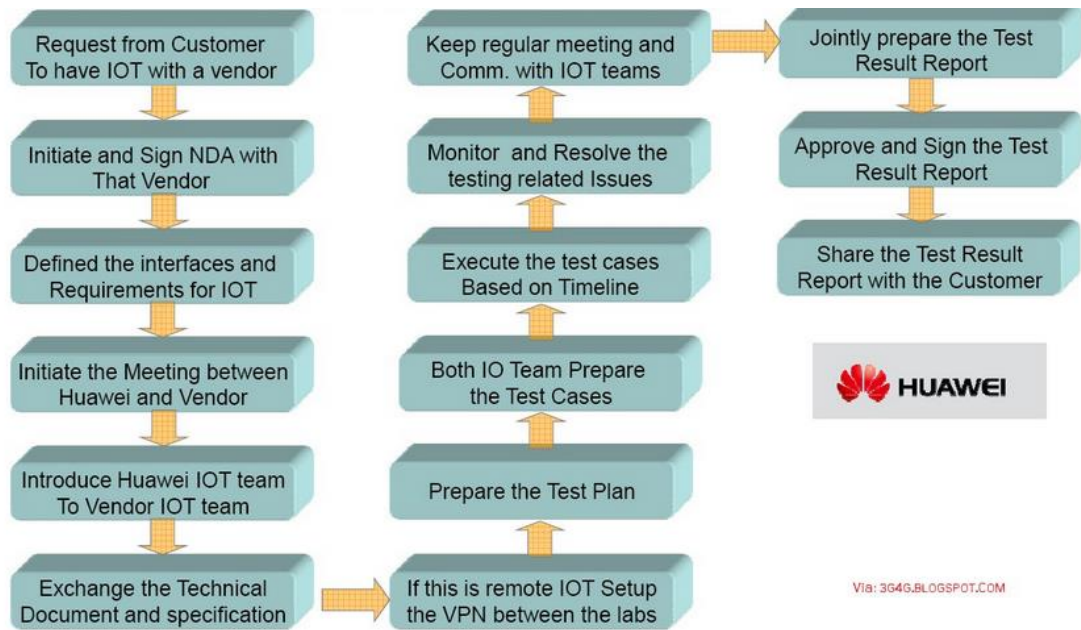
One important part of the testing is to identify the problem cases. The testing allows the possibility of debugging the issues for example handover issues, throughput rates, inter radio access technology (IRAT) performance and mobility cases as well. (Panigrahi 2012)

Issues identified in the early phase by software developers can easily be fixed, and with almost with no additional cost. Where the issues are identified after a commercial launch it may require user software updates or even product recalls, therefore there is a strong emphasis on comprehensive testing in the early development phase, as the costs grow when development work goes further. (Moray 2013)

### 4.1.1 IOT process

Figure 4.1 is Huawei's example of the IOT process flow. In general the work starts by asking customer for a IOT with a vendor. As we can see from the figure, there are many steps just to do the preparation work for the testing session for example non-disclosure agreements have to be taken care of, the interfaces has to be defined, introducing the testing team members and setting up follow-up meetings. The planned test cases are executed and the result report is jointly generated. The results will be assigned by both parties, with last task being to share the test results with the customer.

The same kind of IOT process is in use general.



**Figure 4.1.** Huawei's IOT Process Flow (Huawei's IOT Process Flow 2010)

## 5 PERFORMANCE AND MEASUREMENT

This chapter describes different performance related KPIs for the circuit switched fallback (CSFB) and voice over LTE (VoLTE), including example performance results for CSFB executed by Qualcomm and VoLTE performance results executed by Ericsson.

The measurement for SRVCC was carried by Ericsson with its commercial products and with test phones from Qualcomm.

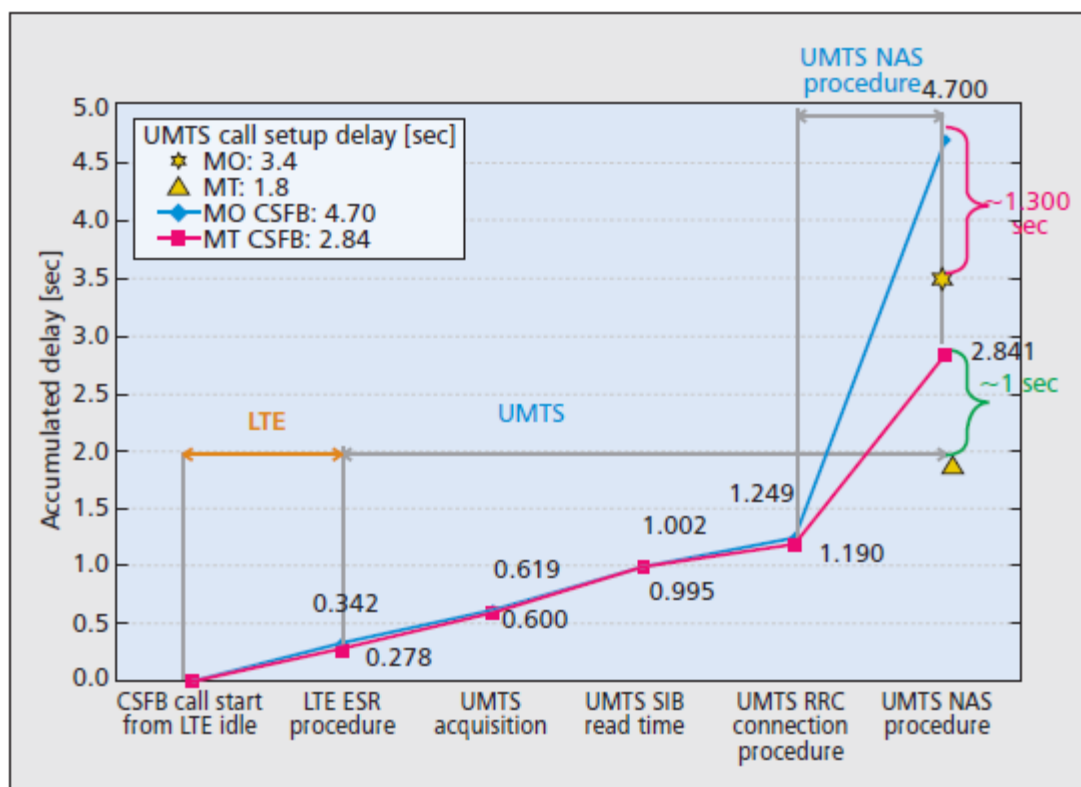
### 5.1 CSFB

Two different performance measurement results are presented for the CSFB. They both handle mobile originated (MO) and mobile terminated (MT) call-setup time for CSFB from LTE network to UTM network and MO and MT calls toward land lines to be able to compare the results. The first performance measurement handles the CSFB overall performance more in general. This part also includes the mobile to mobile (M2M) CSFB performance. The second measurement highlights the three important performance factors and gives an understanding what is the role of each performance factor, especially from user experience point of view. These performance factors are presented each in its own section. The CSFB call setup failure is opened up.

First presented performance figures are based on Qualcomm's published information. The data was collected in a real LTE commercial network with CSFB functionality, where LTE had one frequency overlapping with several UTM frequencies. CSFB voice call procedures were explained in chapter 3.2.2.

Figure 5.1 shows the accumulated CSFB call setup delay for the mobile originated and mobile terminated voice call from LTE to UTM network. The call starts from LTE idle mode. In this case the LTE release 8 with redirection was used.

The information in the figure shows that mobile terminated CSFB voice call takes approximately 1 second longer compared to the legacy network. In mobile originated case the difference is approximately 1.3 seconds. (Bautista et al. 2013)



**Figure 5.1.** Accumulated call setup delay for LTE CSFB to UMTS (Bautista et al. 2013 p. 138)

With UMTS SIB tunneling (release 9) it is expected that excess delay time for the CSFB MO and MT call setup time will decrease by 0.5 seconds for MO and MT CSFB voice calls. This is shown in table 2.

Mobile to Mobile (M2M) call setup delay is the most challenging one. Reason for this is that M2M includes both, MO and MT, procedures. As shown in table 2, the M2M CSFB call takes 2 seconds longer than compared to the legacy UMTS network call setup. This means that overall call setup time is 8.5 seconds. With release 9 it is expected that setup time will decrease with one second.

**Table 2.** Average call setup times in live networks (Bautista et al.2013 p.140)

Call Type	Legacy (UMTS) [sec]	Excess delay to Legacy [sec]	
		CSFB (Rel8)	CSFB (Rel9)
MO to Land-Line	3.4	1.3	0.5
MT from Land-Line	1.8	1	0.5
Mobile to Mobile	6.5	2	1

Later on similar figures will be presented in section 4.1.2 and 4.1.3. When comparing information between these three figures it can be seen that there are some differences. In above figure 4.1 the LTE and UMTS networks were well optimized. The network optimization is often the reason for the differences.

There are three performance factors that that directly impact on user experience:

- data interruption time
- mobile originated (MO) and mobile terminated (MT) call setup times
- call setup reliability

Each of these are explained in detail below. It will give an overview understanding how each performance factor impacts for the end to end call setup and what differences there are between LTE releases and what network acquisition mechanism (for example handover or redirection) is used. The measurement results are from Qualcomms published material. The information has been collected in live 3G commercial network. The radio conditions varied from good to poor. The performance measurement results for 2G have been collected in a lab environment. (Qualcomm 2013)

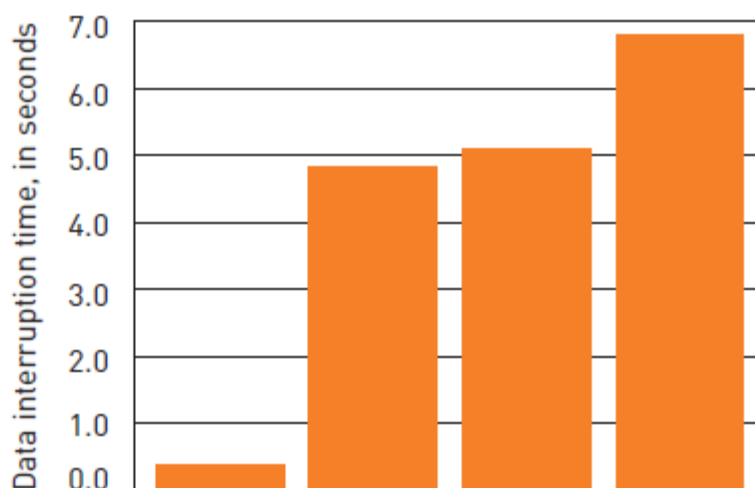
### **5.1.1 Data interruption time**

The data interruption time is related to the situation where UE has an active for example streaming session ongoing and a voice call is initiated. The streaming data transfer is interrupted by the IRAT transition and routing area update. The interruption time depends on the LTE version and also what network acquisition mechanism (handover or redirection) is used.

Figure 5.2, below, from Qualcomm's measurement illustrates the possible data interruption times. From the user experience point of view the smallest impact is when handover based CSFB is used. In practice the subscriber does not notice the 0.3 seconds data interruption time. The interruption times are much longer with redirection based CSFB. Redirection based interruption times are varying between 4.8 seconds to 6.8 seconds. Redirection based interruption times vary from 4.8 seconds to 6.8 seconds. This longer data interruption time can be noticed by the subscriber. It may be that the importance of this interruption time is quite low, as in practice the subscriber is concentrating on receiving the incoming call or making the outgoing call and therefore the interruption time may not be that critical. (Qualcomm 2013)

## UMTS

	Handover	Redirection		
	Rel-8 / Rel 9	Rel 9	Rel-8	
		SI Tunnel	Skip SIBs	Basic
Handover	<b>0.3</b>			
RRC Release		<b>0.2</b>	<b>0.2</b>	<b>0.2</b>
Acquisition on UTRAN		<b>0.2</b>	<b>0.2</b>	<b>0.2</b>
Read MIB & SIBs			<b>0.4</b>	<b>2.0</b>
Camp on Cell		<b>0.1</b>	<b>0.1</b>	<b>0.1</b>
Connection Setup		<b>0.3</b>	<b>0.3</b>	<b>0.3</b>
Optional RAU Procedure		<b>4.0</b>	<b>4.0</b>	<b>4.0</b>
<b>Total Data Interruption Time</b>	<b>0.3</b>	<b>4.8</b>	<b>5.1</b>	<b>6.8</b>



**Figure 5.2.** Data interruption time by voice call setup by setup method. All units are in seconds. (Qualcomm 2013 p.7)

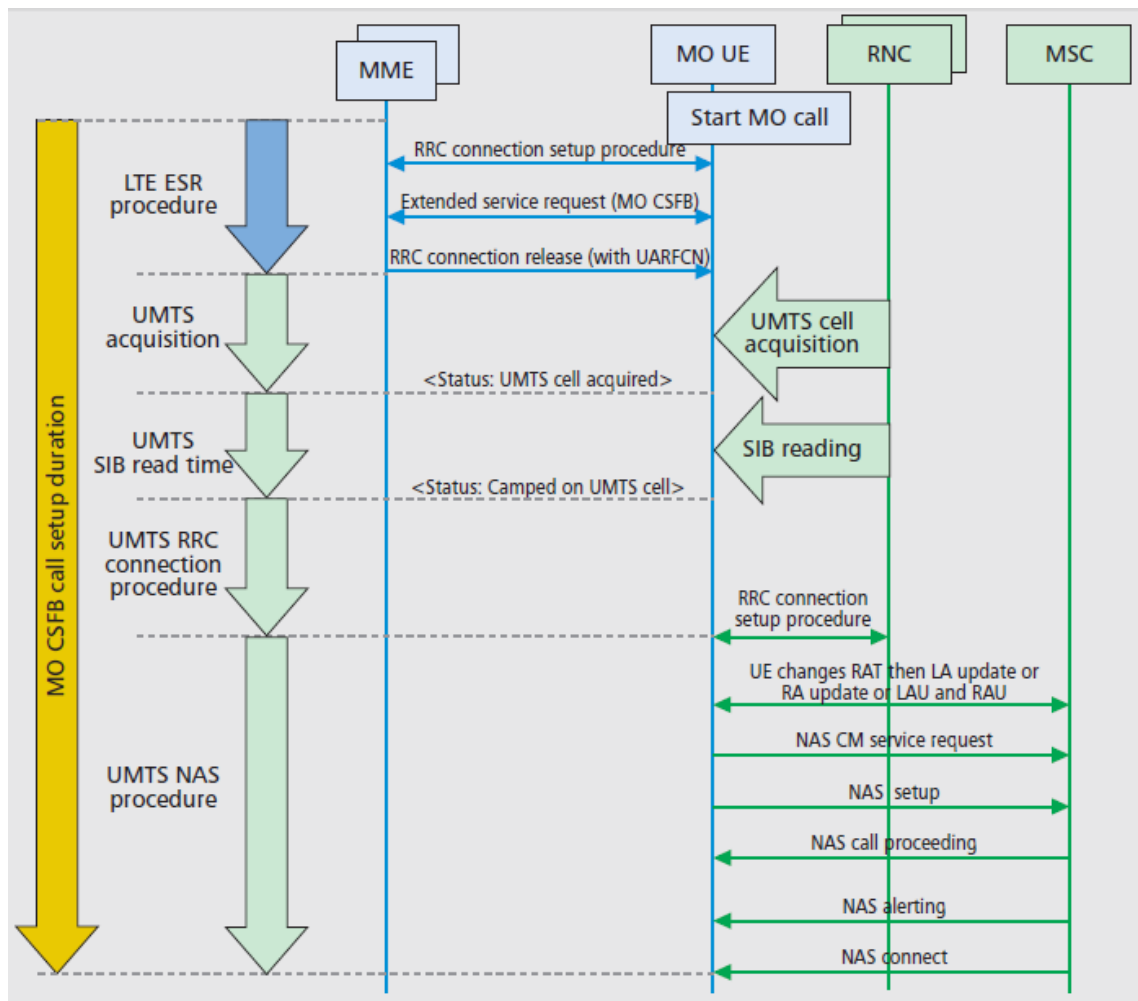
### 5.1.2 Mobile originated call setup time

Mobile originated (MO) call means the outgoing call. When CSFB is used with a mobile originated voice call, there is a delay due to switching from LTE network to 3G network. This has impact on the call setup time.

Figure 5.3 illustrates the difference procedure phases for MO call with CSFB.

The below figure illustrates the circuit switched fallback (CSFB) procedures when a CS call is established from LTE network. It

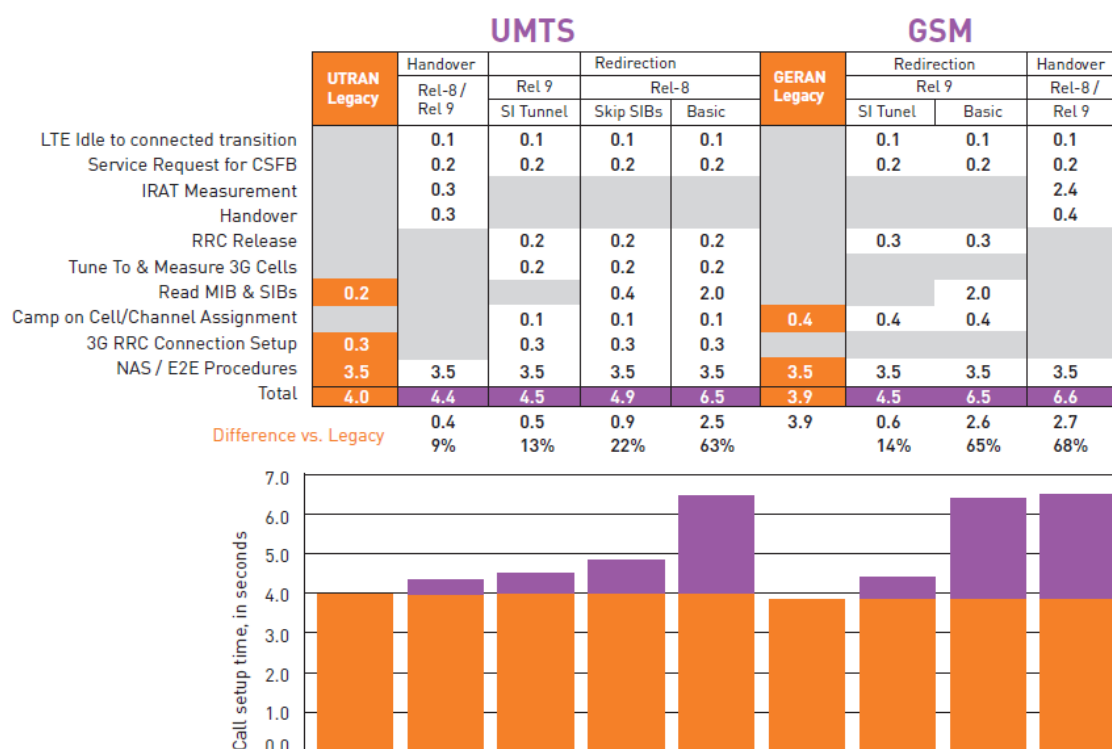
Figure 5.3 illustrates the difference procedure phases for MO call with CSFB. It shows what communication exchanges belong to different procedures. The ESR procedure happens in LTE network and all others in UMTS network.



**Figure 5.3.** Procedures for MO call with CSFB (Bautista et al. 2013 p.137)

MME that belongs to LTE network includes the following components: S-GW, P-GW and eNodeB. RNC that belongs to 3G UMTS network includes SGSN and NodeB.

The below Figure 5.4 from Qualcomm's measurement shows the differences between when the fallback is executed from LTE to UMTS (3G) and GSM (2G) networks and how significant a role the LTE version and network acquisition mechanism (handover, redirection) play. Legacy 3G network call setup values are provided for comparison. The CSFB call setup values should be as good as the legacy values to guarantee a good user experience. The 3G network call setup values were measured in a real commercial network. The 2G network call setup values were measured in a lab environment.



**Figure 5.4.** Mobile originated call setup times, LTE to 3G CSFB by setup method, with comparisons to legacy 3G. All units are in seconds. (Qualcomm 2013 p.5)

From the Figure 5.4 it can be seen that CSFB to UMTS (3G) is the quickest with handover mechanism taking in total 4.4 seconds (only 0.4 seconds longer than the 3G legacy network). In a CSFB handover the target cell is prepared in advance, no SIB reading is required, but instead the IRAT measurement for the signal strength is required. In this below case the IRAT measurement has taken 0.3 seconds.

The redirection based SI tunneling and SIB skipping are also quite close to handover call setup time. The worst connection time is with release 8 basic. In this scenario the call setup takes 6.5 seconds in total and 2.0 seconds is spent reading all the SIBs prior to access. All together the call setup time requires 2.5 seconds longer compared to the legacy network.

Where the CSFB is performed to the GSM (2G) network the redirection based SI tunneling is the quickest option. The call setup time in total is 4.5 seconds and only 0.6 seconds longer compared to the legacy value.

The redirection based basic mechanism and the handover mechanism call setup times are quite close to each other. When redirection basic mechanism is used the call setup time is 6.5 seconds in total, which is 2.6 seconds longer compared to the legacy. The difference is caused by the time that is spent on reading all the SIBs prior to access. With handover the total call setup time is 6.6 seconds that is 2.7 seconds longer than the legacy value. It

can be seen that 2.4 seconds is used for IRAT measurement. It can be also seen that same IRAT measurement for UMTS (3G) takes only 0.3 seconds. The difference is explained by less efficient synchronization and cell identification on the GSM (2G) network. (Qualcomm 2013)

### 5.1.3 Mobile terminated call setup time

Mobile terminated (MT) call means the incoming call. The below Figure 5.5 from Qualcomm test cases illustrates the differences for mobile terminated call setup times when fallback is done to UMTS (3G) networks. Also in this case the LTE version and network acquisition mechanism (handover, redirection) play important role. The legacy 3G network call setup time values are also presented to provide comparison. The CSFB call setup values must be as good as the legacy values to guarantee a good user experience. The 3G network call setup values were measured in a real commercial network. (Qualcomm 2013).

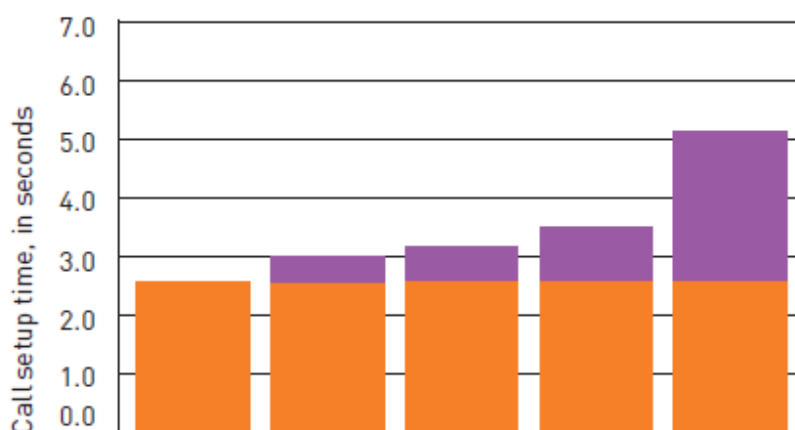
Based on the below Figure 5.5 the quickest call setup value comes by using the handover mechanisms where total call setup time is 3.0 seconds which is only 0.4 seconds longer than corresponding legacy value. Also the redirection based SI tunneling and skip SIBs mechanism are in the same range with legacy value with difference less than one second. The redirection basic based CSFB takes much longer time and total call setup time is 5.2 seconds. As noticed also with outgoing call setup time the basic mechanism spends 2.0 seconds for reading the all the SIBs prior to access.

When comparing the MO and MT test results, can be noticed that NAS/E2E Procedures take shorter time when MT (incoming) call is made. The value for MO is 3.5 seconds and for MT 1.5 seconds. Reason for this is that in MT case there is no paging delay, nor setup delay on the outgoing device. (Qualcomm 2013)

Non-Access Stratum (NAS) is set of protocols in EPS. It is used to convey non-radio signaling between the MME and UE for LTE access (Firmin).

## UMTS

	UTRAN Legacy	Handover	Redirection		
		Rel-8 / Rel 9	Rel 9	Rel-8	
			SI Tunnel	Skip SIBs	Basic
Paging (Assuming 1.2s DRX Cycle)	0.6	0.6	0.6	0.6	0.6
LTE Idle to connected transition		0.1	0.1	0.1	0.1
Service Request for CSFB		0.2	0.2	0.2	0.2
IRAT Measurement		0.3			
Handover		0.3			
RRC Release			0.2	0.2	0.2
Tune To & Measure 3G Cells			0.2	0.2	0.2
Read MIB & SIBs	0.2			0.4	2.0
Camp on Cell			0.1	0.1	0.1
3G RRC Connection Setup	0.3		0.3	0.3	0.3
NAS / E2E Procedures	1.5	1.5	1.5	1.5	1.5
<b>Total</b>	<b>2.6</b>	<b>3.0</b>	<b>3.2</b>	<b>3.5</b>	<b>5.2</b>
<b>Difference vs. Legacy</b>		<b>0.4</b>	<b>0.5</b>	<b>0.9</b>	<b>2.5</b>
		<b>14%</b>	<b>20%</b>	<b>33%</b>	<b>96%</b>



**Figure 5.5.** Mobile terminated call setup times, LTE to 3G CSFB by setup method, with comparisons to legacy 3G. All units are in seconds. (Qualcomm 2013, p. 6)

### 5.1.4 Optimization for CSFB setup delays

As seen in earlier sections, there are many parts that impact on the call setup delay. Qualcomm has done a study in commercial network and analyzed the root causes that caused the delay. First thing was to understand what is the shortest possible CSFB call setup time in a network. This time was compared with the average call setup time. The call setup stages were carefully studied to identify the root cause that was causing the excess delay. (Bautista et al.2013)

Table 3 presents what are the most common delay sources for CSFB call setup in a live network and recommendation how they could be optimized.

**Table 3.** Optimization possibilities for CSFB call setup delays (Bautista et al.2013, p.140)

Delay Source	% of excess delay relative to best call	Possible Optimization
SIB Read Time in UMTS	36	Increase the periodicity of mandatory UMTS SIBs; introduce SIB tunneling and SIB skipping
Paging Delay in LTE	15	LTE DRX cycle settings: Tradeoff call setup time vs. battery life
Core Network Delay	14	Optimize NAS message handling for concurrent CS/PS multi-RAB establishment
WCDMA Access	9.5	Optimize WCDMA layer, especially RACH parameters
LTE Access	6.8	Tune LTE RACH parameters, ensure good LTE coverage

As shown in the table 3 one significant additional call setup time delay is related to the SIB read time in UMTS. This could be optimized by introducing the SIB skipping or SIB tunneling. Paging delay in LTE could be optimized by changing the LTE discontinuous reception (DRX) parameter. This parameter defines how often the paging is done. However there is a tradeoff as the paging consumes the battery life. The core network delay could be improved by optimizing the NAS message handling. The last two delay sources are related to the WCDMA and LTE access.

The optimization possibilities regarding the CSFB call setup delay can be categorized under three areas. First one is network and parameter optimization. Second one is taking new CSFB enhancements into use (for example SIB skipping). Third area covers improvements for the NAS message handling and interwork with LTE and 3G core networks. (Bautista et al.2013)

### 5.1.5 Call setup reliability

In practice the call setup reliability means the percentage of successful calls established a first attempt within a timeout period. If the call setup time is too long the call is recorded as a failure. With legacy networks the call setup reliability is in the range of 98%. The CSFB should have at least the same target to be able to fulfill the user requirements. (Qualcomm 2013)

When CSFB functionality is used the fallback is performed between 3G and LTE networks. There are two important factors that are relevant for the call setup reliability.

First is the handover based CSFB IRAT measurement and related network acquisition. In this case the IRAT measurement value can change before the network acquisition is made and the connection will end in failure. This kind of issue normally comes in high mobility cases. The second is related to the mismatches on the geographic LTE and 3G signal coverage areas, requiring updates on the Mobile Switching Center (MSC) server side. (Qualcomm 2013)

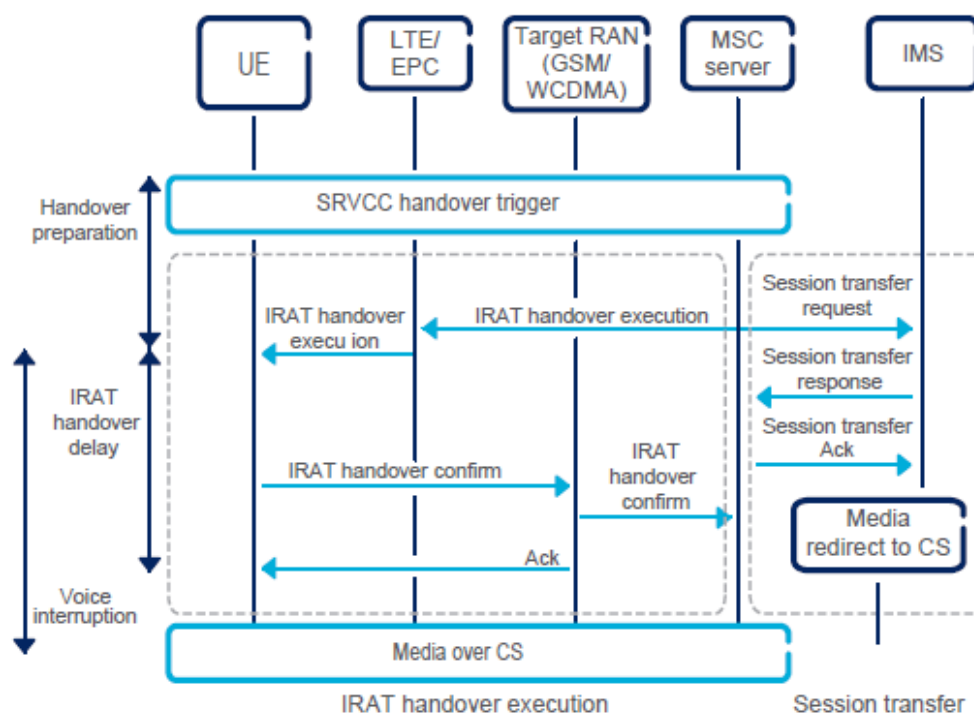
## 5.2 SRVCC

This chapter describes the single radio voice call continuity (SRVCC) performance test results executed by Ericsson (*Ericsson 2012*). There are two important key performance indicators (KPIs) from the point of view of user experience; the voice interruption time and call retention probability.

The voice interruption time is measured from the last voice packet that sent over LTE and the time when voice media is sent over CS access. Voice interruption time consists of IRAT handover execution and session transfer times.

When an in-progress call is handed over from an LTE to GSM/WCDMA network it happens with two steps. IRAT handover moves the UE to GSM/WCDMA network cell and session transfer moves the access control and voice media anchoring from EPC to MSS.

In legacy network these two steps, the RAN IRAT handover and session transfer, impact on the voice interruption time as connections are broken and reconnected. The SRVCC on the other hand initiates these two steps parallel as shown in below Figure 5.6, and this way minimizes the overall interruption time.



**Fig 5.6.** SRVCC handover procedure (Ericsson 2012, p. 6).

The IRAT handover delay is defined as the time between when a UE receives the IRAT handover execution command from the network and when UE is synchronized on the new radio access, and it has received the Ack message. (Ericsson 2012)

According to Ericsson's test results the session transfer process is quick and takes 0.01 seconds. The IRAT handover delay has bigger impact on the voice interruption time.

Figure 5.7 shows the test results that were executed on Ericsson commercial products and with test phones from Qualcomm. The given results are mean times. The mean handover preparation time was 0.52 seconds, handover –procedure delay 0.18 seconds and voice-interruption time 0.19 seconds. All these results were within the range of normal CS IRAT handover times.

PERFORMANCE INDICATOR	RESULTS (AVERAGE)
Handover preparation	0.52s
Handover-procedure delay	0.18s
Voice-interruption time	0.19s

**Figure 5.7.** Test measurement by Ericsson (Ericsson 2012 p. 7)

The call-retention probability is an important factor as it affects the service-interruption time. The 3GPP has defined that voice service interruption time should be less than 0.3 seconds.

Ericsson collected data from commercial deployments and based on the data the actual retention failure rate was in most cases less than 0.5%. Similar statistical characteristics are expected to be identified for SRVCC. To obtain the 98% voice call retention rate, the SRVCC handover has to have very high success rate (Ericsson 2012).

### 5.3 IOT

This work is concentrating on the interoperability testing (IOT) for circuit switched fallback (CSFB) that includes interoperability between networks and UE. The voice over LTE (VoLTE) functionality testing is out of scope for this work, as there was no test environment yet ready that could have been used.

#### 5.3.1 Test environment

Test environment was similar kind of configuration as in Figure 5.8. The CSFB functionality was tested between LTE and 3G networks.

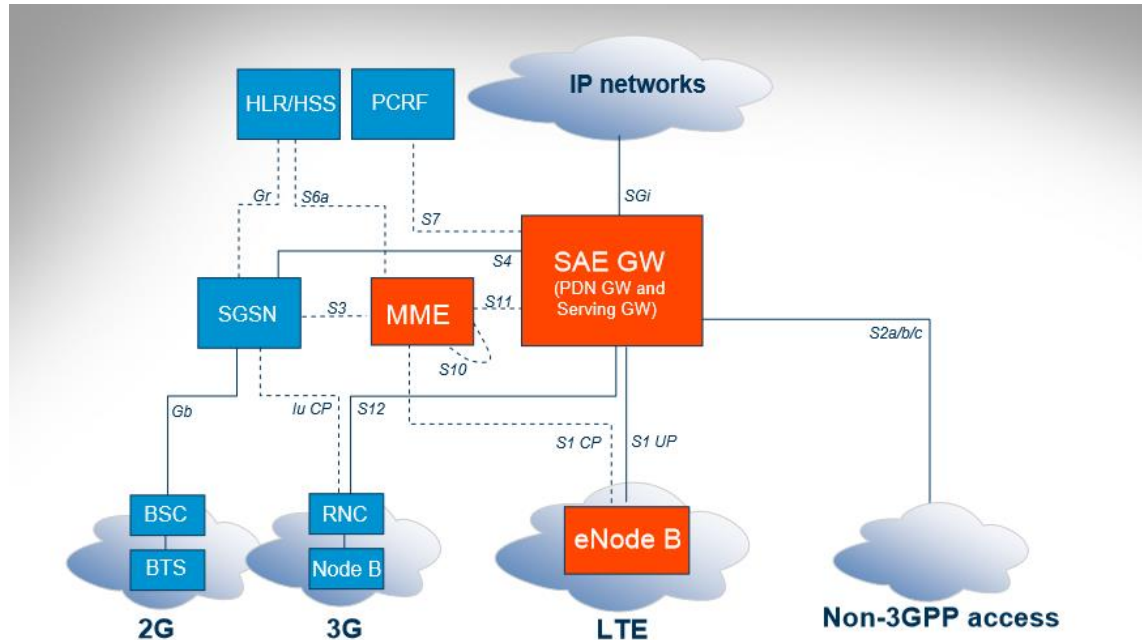


Figure 5.8. LTE/SAE Architecture (Fritze 2008).

### 5.3.2 Test results

The following test cases were executed in Microsoft Mobile Oy company's IOT lab environment:

- MO CSFB in idle mode with only default bearer
- MO CSFB in connected mode with data transfer ongoing on only default bearer
- MT CSFB in idle mode on default bearer
- MT CSFB in connected mode with data transfer ongoing on default bearer
- Emergency CSFB call in connected mode
- Emergency CSFB call in idle mode

The test cases are defined by the network vendor and each test includes many steps. The more detailed information about the test cases is confidential information and therefore not included in this thesis.

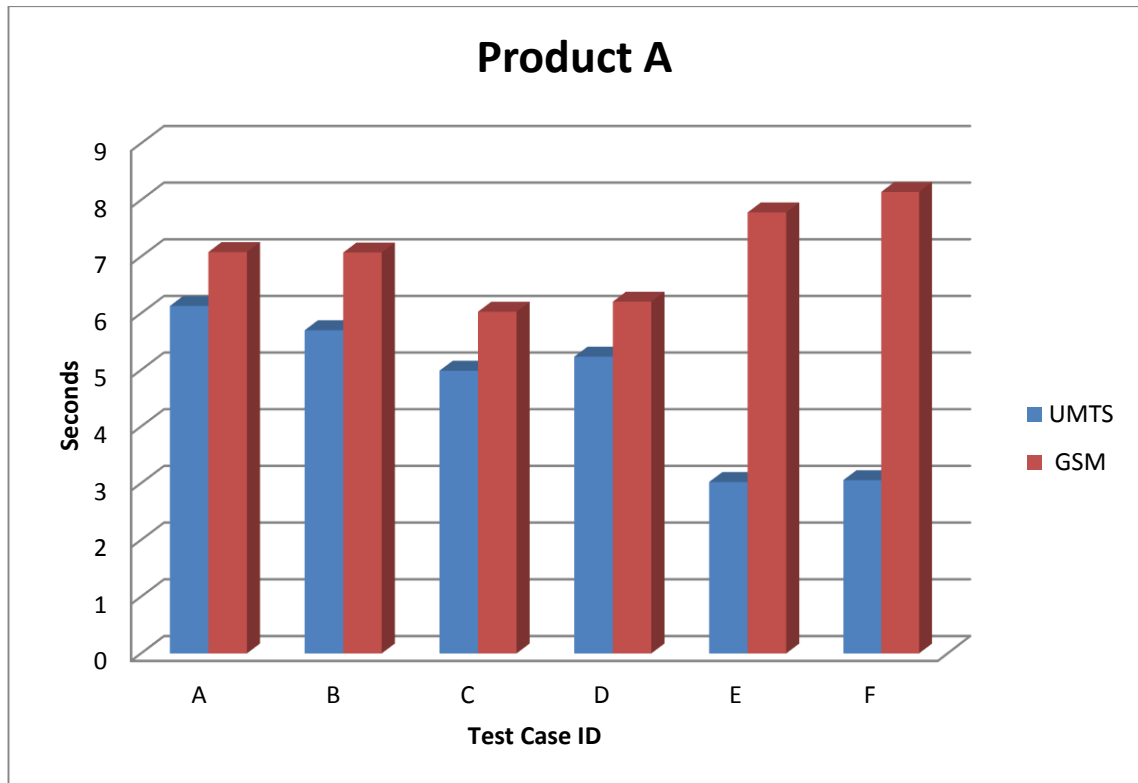
The call setup time was calculated from the time between the extended service request and alerting messages. Each of the cases was executed ten times in a row. The below results are the mean figures of the test runs. It is important to notice that if there is for example one longer or failed test run, it can have significant impact to the mean result time. The test cases had network acquisition mechanism redirect basic (release 9) in use. The network device vendor gives recommendations as to configurations (for example network acquisition mechanism) that should be used with their devices. It is up to the operator to decide the final configuration that is to be used.

In Figure 5.9 the test results for the Product A are, with the exception of test case F: emergency CSFB call in idle mode, at an acceptable level. In the GSM scenario the call set up for this test case takes over 8 seconds, taking over 2 seconds longer than compared to product B in Figure 4.10. The difference can be explained with network configuration and with failed authentication. In general 2G network has more variation in test results due to the system information scheduling. This explains also why the 2G results for product A has a lot variation. An example of the system information scheduling; in the test lab the location update has to be done prior the call setup.

#### Test Case / Product A

ID	Test Case Description	UMTS	GSM
A	MO CSFB in Idle Mode with only Default Bearer	6,129	7,081
B	MO CSFB in Connected mode with data transfer ongoing on only default bearer	5,702	7,073
C	MT CSFB in Idle mode on default bearer	4,986	6,028
D	MT CSFB in Connected mode with data transfer ongoing on default bearer	5,234	6,205
E	Emergency CSFB Call in Connected Mode	3,022	7,781
F	Emergency CSFB Call in Idle Mode	3,053	8,141

*All units are in seconds.*



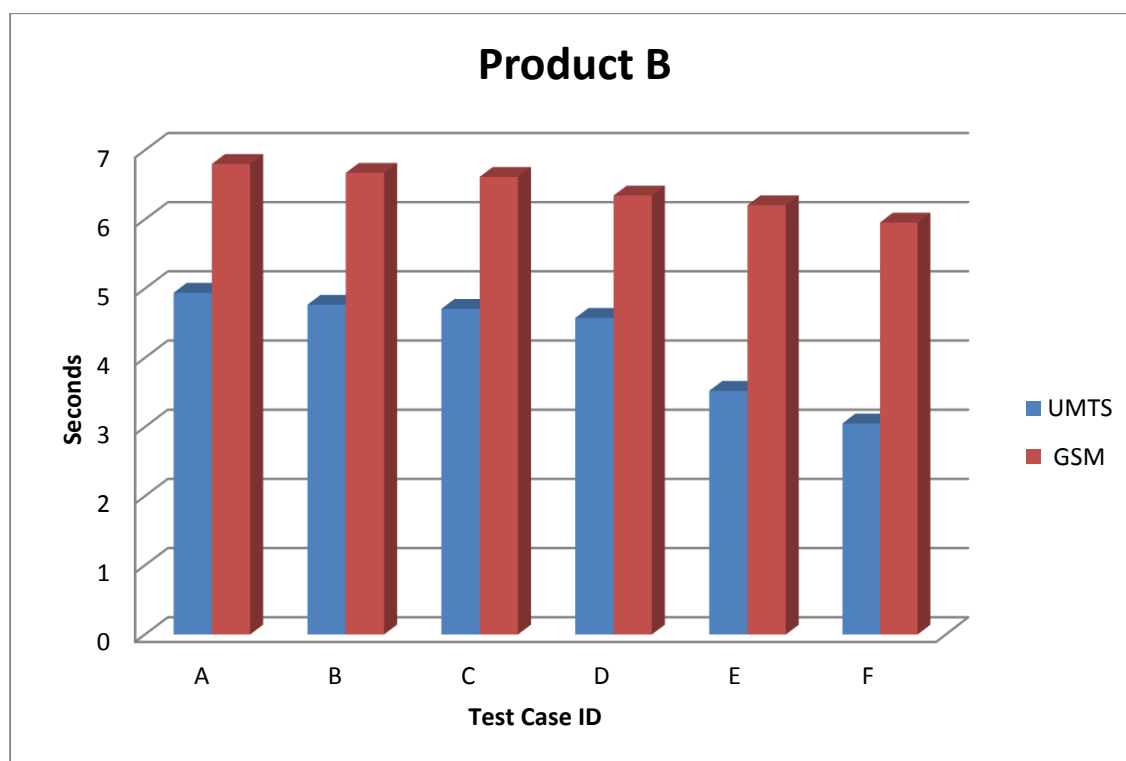
*Figure 5.9. Product A test results*

IOT results for product B are presented in figure 5.10. The values are in acceptable level.

#### **Test Test Case / Product B**

ID	Test Case Description	UMTS	GSM
A	MO CSFB in Idle Mode with only Default Bearer	4,940	6,802
B	MO CSFB in Connected mode with data transfer ongoing on only default bearer	4,767	6,673
C	MT CSFB in Idle mode on default bearer	4,707	6,615
D	MT CSFB in Connected mode with data transfer ongoing on default bearer	4,576	6,345
E	Emergency CSFB Call in Connected Mode	3,523	6,205
F	Emergency CSFB Call in Idle Mode	3,051	5,955

*All units are in seconds.*



**Figure 5.10.** Product B Test results

In both test scenario can be seen that emergency call setup times are quicker compared to normal call setup times. Reason for this is that the emergency call's priority is set higher compared to normal calls. The emergency call setup time for CSFB for LTE was found to be between 3.0 seconds and 3.5 seconds. In the case of emergency call location area update (LA), routing area update (RA), modify packet data protocol (PDP) context (2G connect mode) and authentication, procedures are made after the call release. Hence why call establishment is faster than in normal CS calls, the downside is that call the release procedure takes longer and therefore the return back to LTE is much slower.

There can be different situations that can happen during the testing.

In cases where the test results seem to differ a lot, the only way to find the reason for the behavior is to investigate more in detail the test run log files.

The following reasons can explain the long voice call setup times:

- authentication failure
- location update failure
- packet data tracking area update failure
- system information scheduling in 2G, for example, in the test lab the location update has to be done prior the call setup. (In a commercial network location update can be done also after the call setup in which case the call setup time is shorter).

## 6 CONCLUSIONS

It is obvious that importance of mobile telecommunication has increased dramatically during the last ten years and it still keeps growing in the coming years. In subscriber markets there is high demand for higher data rates, bigger capacity and also wider network coverage. When thinking the big picture, there are many different frequencies in use but they are not necessarily used for the same purpose all over the world. The network operators are mainly driving the network building, but there are other parties involved for example network vendors and device manufacturers. The standardization work for the LTE is done by the ITU. The 3GPP working group does the LTE development work. The development is published in different releases. This thesis mainly has performance results related to the LTE release 8 and 9.

Nowadays there are different mobile network solutions, such as 2G, 3G and 4G, used in parallel. This thesis covers the mobile network technologies development from GSM to LTE. With the LTE it is possible to gain high data rates as it uses wider spectrum per connection, multiple antenna paths and efficient coding of the sent and received data. The LTE coverage map illustrates how widely the LTE was already used in year 2014. The LTE evolution is divided into three phases. In the first phase the data communication is handled by the LTE packet switched (PS) networks if available. Second phase of LTE evolution includes voice over LTE (VoLTE) that brings improvement for the voice quality. It also introduces the single voice call continuity (SRVCC). In the last phase all network traffic is IP based.

Goal of the thesis was to assess the LTE voice solutions and to analyse the performance measurement results. The thesis introduces the LTE voice solutions circuit switched fallback (CSFB) and single radio voice call continuity (SRVCC). The CSFB enables voice calls and data traffic between circuit switched and LTE networks. SRVCC is VoLTE's solution. With SRVCC the active voice call is moved from LTE network to legacy network.

The performance is important part as it has direct impact to the subscriber experience. The performance figures are often compared to the legacy network figures. The LTE performance figures should be at minimum as good as the legacy figures to fulfill the subscriber requirements.

CSFB has following three important performance KPIs:

- data interruption time
- call setup times (for MO and MT)
- call setup reliability

All these above mentioned performance KPIs are heavily dependent on the LTE release and on the used network acquisition mechanism. The figures are based on Qualcomm's published material.

The acquisition of the voice network is needed when UE switches from LTE to 2G or 3G network. There are two ways how the network acquisition can be done during CSFB; handover or redirection. With the handover the target cell is prepared in advance and the UE can connect directly to the cell. With the redirection UE has dedicated target frequency and it can select any cell within the frequency. When handover mechanism is used the IRAT signal strength measurement is needed. The IRAT measurement is not needed prior the redirection. Data interruption time is shortest with the handover procedure when CSFB is performed to UMTS (3G). Total data interruption time is approximately 0.3 seconds.

In call setup times, the CSFB to UMTS (3G) is shortest with handover mechanism. MO call setup time is 4.4 seconds. Where CSFB is performed to the GSM (2G) network, the redirection based SI tunneling is the quickest option taking 4.5 seconds that is 0.6 seconds longer compared to the legacy network.

The handover procedure gives also best call setup time in MT cases when the CSFB is performed to UMTS (3G). In this case the total call setup time is 3.0 seconds, which is 0.4 seconds longer compared to the legacy network value. The call setup reliability means the percentage of successful calls established a first attempt within a timeout period. If the call setup time is too long the call is recorded as a failure.

When CSFB functionality is used the fallback is performed between 3G and LTE networks. There are two important factors that are relevant for the call setup reliability. First is the handover based CSFB IRAT measurement and related network acquisition. In this case the IRAT measurement value can change before the network acquisition is made and the connection will end in failure. This kind of issue normally comes in high mobility cases. The second is related to the mismatches on the geographic LTE and 3G signal coverage areas.

The SRVCC solution transfers the active VoLTE call when UE moves from LTE to legacy network. The following two KPI's have impact to the user experience; the voice interruption time and call retention probability. The performance information is based on Ericsson's published material. The voice interruption is meeting the required limits. For call retention probability there was no real commercial data yet available. Based on the information received from the commercial deployments it is expected that SRVCC will have retention failure less than 0.5%.

The interoperability testing (IOT) was executed by Microsoft Mobile Oy in lab environment. Test cases were different CSFB scenarios. Failover to GSM (2G) and UMTS (3G) was tested with two different UEs. The test cases had network acquisition mechanism redirect basic (release 9) in use.

IOT included the following six test cases:

- MO CSFB in idle mode with only default bearer
- MO CSFB in connected mode with data transfer ongoing on only default bearer
- MT CSFB in idle mode on default bearer
- MT CSFB in connected mode with data transfer ongoing on default bearer
- Emergency CSFB call in connected mode
- Emergency CSFB call in idle mode

The results in overall were in acceptable level. The CSFB was quicker to 3G with both UEs compared to the CSFB to 2G. The test results illustrates that 2G network has more variation. This is due to the system information scheduling. In lab environment this means for example that the location update is done prior to call setup.

With both UEs the emergency call setup times are quicker compared to normal call setup times. Reason for this is that the emergency call's priority is set higher compared to normal calls. The emergency call setup time for CSFB for LTE was found to be between 3.0 seconds and 3.5 seconds.

The following conclusions can be made from the LTE development, voice solution performance factors and tests execution. The mobile networks are developed very quickly. People who are working in this area really needs to stay up to the development speed to be able to understand in very detail level all related links and topics that may have impact for example to the IOT planning and execution. The performance figures can vary a lot depending is the network mechanism handover or redirection and what 3GPP release is chosen. The network configuration in lab environment might be different to commercial network. It is essential to understand the difference. There was only a few published measurements found. It seems that companies are treating this kind of information confidential.

The testing focus is moving from device centric more to service centric. The interoperability has to be working E2E from the service aspect to be able to fulfill the promised service quality level.

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