PERFORMANCE EVALUATION OF VOICE HANDOVER BETWEEN LTE AND UMTS

Joyce Namakoye

A Research Report submitted to the Faculty of Engineering and the Built Environment, University of the Witwatersrand, in partial fulfilment of the requirements for the degree of Master of Science in Engineering

Johannesburg, 2011

DECLARATION

I declare that this research report is my own unaided work. It is being submitted for the Degree of Master of Science to the University of the Witwatersrand, Johannesburg. It has not been submitted before for any degree or examination to any other University.

Candidate Signature : 

Name : Joyce Namakoye

Date : 10th January 2012

ABSTRACT

The main objective of seamless mobility is to enable mobile users to stay connected while roaming across heterogeneous networks. As cellular networks evolve from the third generation Universal Mobile Telecommunication System (UMTS) to the Long Term Evolution (LTE), a new Evolved Packet Core (EPC) will support heterogeneous radio access networks on the same platform. UMTS provides voice services in the circuit switched domain; while LTE operates in the packet switched domain. Cellular network operators thus face the challenge of providing voice services during initial deployment of LTE due to difficulty in mobility between the two domains. Seamless voice handover between packet switched LTE and the circuit switched UMTS network is therefore an important tool in solving this problem.

This report investigates the performance of inter-Radio Access Technology voice handover between LTE and UMTS. The schemes evaluated were Voice Call Continuity (VCC) for UMTS to LTE handover and Single Radio Voice Call Continuity (SRVCC) for LTE to UMTS handover. The performance evaluation was done using mathematical models and equations that were derived for the handover service interruption time. The resulting equations were simulated and the output was analysed and compared with the Third Generation Partnership Project (3GPP) specifications.

ACKNOWLEDGEMENT

I wish to express my sincere gratitude to my supervisor, Professor Rex Van Olst for his guidance and unwavering support. Thank you for being patient with many of my shortcomings and always being willing to help.

I also wish to convey my sincere thanks to members of the Center for Telecommunications Access and Services (CeTAS) at the University of the Witwatersrand, not only for their academic support but the financial assistance which enabled me to complete this work.

My special gratitude also goes to Mr. Kyungmin Kim of Yonsei University, South Korea as well as Mr. Ryan Van den Bergh of CeTAS and Mr. Giuseppe Piro of Telematics Labs for their advice on the subject matter of this research work.

I would like to especially thank Mr. Ryan Van den Berg, Mr. Aveer Ramnath, Mr. Ling Cheng, Mr. Jaco Versfeld, Mr. Kennedy Ifeh, Mr. Julius Popoola, Ms. Folasade Danhunsi, Mr. Mehroze Abdullah and all the members of CeTAS for the different roles they played in making this work come to fruition.

Lastly but not least I would like to thank my family and friends whose faith, friendship and love gave me the courage to take on this challenge. Thank you for believing in me.

DEDICATION

To Alex

ABBREVIATIONS

3GPP	Third Generation Partnership Project
BLER	Block Error Rate
CQI	Channel Quality Indicator
CS	Circuit Switched
DTM	Dual Transfer Mode
ENodeB (ENB)	Evolved Node B
EPC/S	Evolved Packet Core/System
E-UTRAN	Evolved Universal Terrestrial Radio Access Network
GERAN	GSM EDGE Access Network
HSPA	High Speed Packet Access
IMS	IP Multimedia System
Inter-RAT	Inter Radio Access Technology
IP	Internet Protocol
ITU	International Telecommunications Union
LTE	Long Term Evolution
MAC	Media Access Control
MIMO	Multiple Input Multiple Output
MME	Mobility Management Entity
OFDMA	Orthogonal Frequency Division Multiple Access
P-CCPCH	Primary Common Control Physical Channel
PDCP	Packet Data Control Protocol
PGW	Packet Data Network Gateway
PS	Packet Switched
RAB	Radio Access Bearer
RAN	Radio Access Network
RLC	Radio Link Controller
RRC	Radio Resource Control
RRM	Radio Resource Management
RSRP	Reference Symbols Received Power
SC-FDMA	Single Carrier Frequency Division Multiple Access
SGW	Serving Gateway
SRNC	Serving Radio Network Controller
SR-VCC	Single Radio Voice Call Continuity
UMTS	Universal Mobile Telecommunications System
UTRAN	Universal Terrestrial Radio Access Network
VoLGA	Voice over LTE via Generic Access
WCDMA	Wideband Code Division Multiple Access
Wi-Fi	Wireless Fidelity
WiMAX	Worldwide Interoperability for Microwave Access
FFS	For Further Study

LIST OF FIGURES

LIST OF TABLES

LIST OF EQUATIONS

CONTENTS

1 Introduction

1.1 Background

Recent advances in telecommunications show a general trend towards high-speed wireless networks with emphasis on an Internet Protocol (IP) based backbone and seamless mobility across heterogeneous networks. Initiatives such as the Multi Service Forum (MSF) [1], Open Mobile Alliance (OMA) [2], Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN) [3] and the IP Multimedia System (IMS) [4] have all defined a Next Generation Network (NGN) whose application and service layers are converged while maintaining heterogeneous access methods. This indicates that the future beyond third generation systems will consist of various radio access technologies, such as Global System for Mobile Communications (GSM), General Radio Packet Service (GPRS), Universal Mobile Telecommunications System (UMTS), Wireless Fidelity (Wi-Fi) and Worldwide Interoperability for Microwave Access (WiMAX). These radio access networks will be interconnected by mobile network operators to maximise spectrum, broaden the range of services and provide inter-technology mobility for multi-Radio Access Technology (RAT) mobile users.

Mobile network operators and telecommunications equipment vendors are therefore investing heavily in inter-technology mobility to take advantage of its benefits. Inter-technology mobility which is also referred to as inter-RAT handover or vertical handover or seamless mobility, involves a user terminal being able to seamlessly move from one radio access network to another without discontinuity in service.

In the next section, the meaning of inter-technology mobility is explained in detail.

1.1.1 Vertical Handover

In mobile communications, vertical handover generally refers to the ability of a network operator to provide continuous service across different radio access networks or RATs [5]. It means that the mobile user is unaware of the transfer of services during the procedure. Vertical handover thus provides a user with the ability to use services irrespective of the access network that is currently being used or that may soon be used. In technical terms, vertical handover refers to a user terminal changing the type of radio access technology it uses to access a supporting infrastructure, in order to support mobility. For example, a handset capable of both UMTS and GSM technologies will be able to handover between GSM and UMTS networks whenever necessary. A network operator may therefore deploy GSM base stations in sparsely populated or low traffic areas and UMTS in densely populated or commercial areas in order to leverage the existing GSM network. As a result, a mobile user travelling from a rural area is able to use low speed GSM access but can easily handover to UMTS radio access when it becomes available. Vertical handover therefore refers to the automatic change from GSM to UMTS in order to maintain seamless communication. This differs from 'horizontal handover' between different wireless access points that use the same technology such as two GSM base stations. The terms inter-technology mobility, vertical handover, inter-RAT handover and seamless mobility were used interchangeably throughout this report [6].

Vertical handovers between different networks serve different purposes. For example, in the recent past, vertical handovers between Wireless Local Area Networks (WLANs) and UMTS networks have attracted a great deal of attention in industry and academia due to the benefit of utilizing the higher bandwidth and lower cost of WLANs as well as better mobility support and larger coverage of UMTS. Similarly, vertical handovers between WiMAX (Wireless Microwave Access) networks and LTE have garnered a lot of appeal in the deployment of femtocells or home eNodeBs[1] given that LTE femtocells provide better indoor coverage at a lower cost. For example a WiMAX/LTE capable handset may use an LTE femtocell indoors and automatically switch to WiMAX cells when extensive outdoor coverage is needed. This enables mobile operators to inexpensively deploy the costly but efficient LTE technology as well as extended data coverage using the less expensive WiMAX technology [7].

In the case of LTE and UMTS, vertical handover particularly serves the purpose of leveraging the existing UMTS network. The advent of LTE and an all-IP Evolved Packet Core (EPC) which can accommodate multiple radio access systems such as GSM, WLAN, WiMAX and UMTS in an integrated manner while providing mobility management among these systems, brings the reality of seamless mobility closer. A fundamental requirement for LTE and EPC deployment is that users expect new exciting services as well as existing services. In addition, operators want to leverage their existing network investments before a complete move to LTE can be made. Furthermore, circuit switched services such as SMS (Short Message Service) and voice services are heavily reliant upon legacy systems such as UMTS. Unfortunately, LTE can only provide voice services over IP thus there is no guarantee of voice or SMS service continuity when a user roams between LTE and UMTS networks.

[1]A femtocell is a small cellular base station, typically designed for use in a home or small business. It connects to the service provider’s network via broadband (such as Digital Subscriber Line or cable).

There is therefore a need for a seamless inter-RAT handover scheme between LTE and UMTS networks. Inter-technology mobility is an important tool in solving this dilemma.

Vertical handover is a major challenge to mobile network operators. To begin with, providing seamless mobility to users requires considerable technical effort, depending on the kind of services and corresponding mobility management that may be required. Technical issues such as security, battery life of the mobile terminal and spectrum allocation have to be dealt with. Secondly, in order to satisfy a user's service demands, a network operator faces the challenge of choosing the best interworking network architecture and handover scheme. Thirdly, in a case where networks are owned by different operators, the operators must cooperate with each other to realize the unified authentication, authorization and accounting, as well as integrated location management [8].

1.1.2 Trends in Vertical Handover

Although international organisations such as the Third Generation Partnership Project (3GPP) and the Institute of Electrical and Electronic Engineers (IEEE) define specifications for handover between their technologies, the realization of these specifications is a complicated issue. This is due to the fact that different generations of cellular network technologies operate in different radio spectra, use different modulation schemes and are supported by different network cores. A considerable number of papers have been written on the subject of inter-technology mobility. They include handover between IEEE and 3GPP networks e.g. WLAN and UMTS, WiMAX and UMTS.

One of the earlier papers written on the subject of vertical handover and its performance was by Gregory Polloi in March 1996 [9]. More recently, vertical handover between IEEE's Wi-Fi and WiMAX technologies and 3GPP's UMTS have attracted a lot of attention. Some of the papers written on the subject vertical handover between 3GPP and IEEEE networks include [10][11][9][12]. For example, [10]proposed a handover scheme using a protocol known as the takeover protocol. This handover scheme reduced the average handover latency and enabled fast and seamless handover between any two different access technologies. It was also said to achieve better performance compared with conventional handover schemes, with respect to handover latency, packet loss, and power consumption. Papers have also been written on handover between different generations of 3GPP networks for instance[13]deals with performance analysis of Inter-RAT handover between GSM and UMTS while [14]evaluates the inter-RAT handover performance between WCDMA and CDMA-2000. It suggests parameters for optimum performance of handover between WCDMA and CDMA-2000.

Whereas a lot of research work done proposes new solutions to the inter-RAT handover problem, most of the work simply evaluates performance of existing solutions. Solutions proposed range from the introduction or removal of network entities, to creating a new protocol, to optimisation of network software. Some of these solutions have been backed and standardized by standards organisations and network operators. In this research, the task of evaluating the performance of inter-RAT handover for voice services between UMTS and LTE was undertaken. This task sought to resolve the problem of selecting the most appropriate inter-RAT handover scheme for voice services for an operator that runs both LTE and UMTS networks.

1.2 Problem Statement

This section formalises the research question that will be answered, which is:

"What is the best voice handover scheme between LTE and UMTS networks with regard to 3GPP specifications?"

1.3 Objectives of the Research

The aim of this research was to find the best handover scheme for voice services between UMTS and LTE networks. The two schemes found to be the best were Single Radio Voice Call Continuity (SRVCC) for LTE to UMTS handover and the Voice Call Continuity (VCC) scheme for UMTS to LTE handover. The research sought to answer the following questions about voice handover schemes:

- Was the handover scheme standardized by a major telecommunications standards organisation?
- Is the handover scheme more viable compared to other schemes?
- Did the service interruption time of the handover scheme conform to 3GPP specifications?

1.4 Methodology

In order to accurately evaluate the performance of a vertical handover voice scheme, one must use a critical step by step approach. The research was performed using the approach explained in detail in Chapter Five. However, a brief explanation of the approach is given in this section and was as follows.

Prior to selecting the best approach to evaluate inter-RAT voice handover schemes it was important understand the available evaluation techniques. Therefore a detailed study on methods used to evaluate handover techniques was done for intra-LTE, intra-UMTS and inter-RAT handovers. It was found that evaluation could be done using three methods:

The first approach would be to perform drive tests in operational LTE, IMS and UMTS networks. Drive tests would be done in areas with overlapping LTE and UMTS coverage so as to create several scenarios of the inter-RAT handover. Statistics of the handovers would be analysed. For example, one would analyse the success rate of handover from LTE to UMTS and then from UMTS to LTE. In addition, the service interruption time and perceived call quality would be examined. This approach was the best as it would give a real-life scenario of the handover under investigation. Unfortunately, there is currently no network operator in South Africa that has fully deployed and interworked LTE, IMS and UMTS systems. Vodacom South Africa was the closest candidate for this type of testing because they have an existing LTE network; however, their IMS network is still under construction and is not yet fully interworked with the LTE and UMTS components. Therefore a different approach had to be taken.

The second method was by modelling UMTS and LTE network nodes, their interfaces, properties, protocols and handover techniques using proprietary software tools such as OPNET, MATLAB or open source software such as NS-2, NS-3 and LTE-Sim. However, this proved to be very difficult since handover modules for UMTS and LTE within these tools are not yet well developed and some of the software tools such as OPNET were not available.

The third approach was through developing mathematical models that emulate the behaviour of the different message signal flows that occur during the voice handover. Based on these mathematical models, equations were derived for the service interruption time of the handover scheme. The resulting equations along with their mathematical models could be coded using any intuitive programming language such as MATLAB or C++. These tools were readily available in the laboratory. The software tool used for this research was therefore MATLAB.

MATLAB® simulations were run with theoretical network parameters shown in Chapter Five. The choice of network parameters was from typical values usually found in UMTS and LTE networks.

Simulations were done in two phases. The first phase was done with static network parameters. That is to say, a single user terminal, with predefined mathematical behaviour and network parameters was communicating with the UMTS and LTE network nodes that also had static behaviour.

The second simulation was with a dynamic network. That is to say, the behaviour of the network nodes was modelled in real time. For example as the user terminal accessed the UMTS and LTE network nodes, the nodes were processing messages from other user terminals in real time.

Details of how the simulations were done are further discussed in Chapter Five.

1.5 Summary of the Research Report

This research report is divided into eight chapters and a list of references. The first four chapters make up the literature survey while the last four chapters deal with the approach taken to resolve the research question. More specifically, Chapters One, Two, Three and four look into the cellular technologies whose inter-RAT voice handover is to be evaluated, their protocols and the possible handover schemes that can be used. Chapters Five to Eight deal with the approach taken in resolving the research question, the simulation set up, the results obtained, recommendations and the conclusions. The individual chapters are organised as follows:

Chapter One serves as a basic introduction to the subject of voice handover between LTE to UMTS, why it should be researched, the problem statement, the objectives of the research and the approach that was taken to tackle the problem.

Chapter Two reviews the evolution of cellular network technologies over the years. It talks about the first generation in the early 1980s to the now imminent fourth generation. It also talks about the standardization of these technologies and limits the scope of the discussion to 3GPP technologies.

Chapter Three discusses the third and fourth generation 3GPP networks under scrutiny namely; UMTS and LTE. Their architecture, physical layers, protocol models and handover mechanisms within the network and across different networks are examined. It also briefly discusses 3GPP’s networks beyond the third generation, namely, High Speed Packet Access (HSPA).

Chapter Four introduces the different voice handover techniques used for inter-RAT handover between LTE and UMTS. It discusses each technique, its network architecture and operation and finally compares the techniques to each other.

Chapter Five is an in-depth look at the key research question. In this chapter, I clearly elaborate the research question and issues that arose while trying to answer this question such as the possible approaches that could be taken to resolve it and the different outcomes of these approaches.

From the discussions in Chapter Four, the two voice handover techniques for LTE to UMTS and UMTS to LTE handover were identified as SRVCC and VCC respectively. Chapter Six therefore discusses the methodology used in evaluating the performance of the SRVCC and VCC techniques. This chapter shows the message signal flows for the SRVCC and VCC procedures and their mathematical models.

Chapter Seven presents and discusses the results obtained from the MATLAB simulation done for the SRVCC and VCC procedures. The results are explained and compared with 3GPP specifications. The different factors that contribute to the results are also stated.

Chapter Eight makes recommendations based on the obtained results, concludes and gives insight into future work that could be done on the subject of voice handover between LTE and UMTS.

1.6 Conclusion

This chapter serves as an introduction to the subject matter of inter-RAT voice handover between LTE and UMTS. It introduces the research, why it should be undertaken and formalises the research question. The methods that were used to tackle it are also clearly stated. It finally gives a summary of how the report is organised.

2 Cellular Network Technology Evolution

2.1 Introduction

The road to today's fourth generation mobile systems has been quite long. In order to understand complex 3G and 4G mobile systems, it is important to understand the evolution process. Technology development has evolved from expensive massive equipment to affordable light units. It has also changed from being standardized by national or regional bodies to a global standards organisation such as 3GPP.

3GPP's technologies are the most deployed worldwide. Cellular networks can generally be grouped into four generations, namely 1G, 2G, 3G and 4G. Each generation is an improvement on the previous generation in terms of performance and cost. The latest step in the evolution process is the Long Term Evolution (LTE) and LTE Advanced.

2.2 First Generation (1G)

First generation cellular systems began in the 1980's. These were analogue telecommunication standards introduced in the 1980s and continued until replacement by 2G digital telecommunications. The main difference between the two succeeding mobile telephone systems, 1G and 2G, is that the radio signals that 1G networks operated were analogue, while 2G networks are digital.

One example of a 1G network was the Nordic Mobile Telephone(NMT), used in Nordic countries, Switzerland, Netherlands, Eastern Europe and Russia. Others included Advanced Mobile Phone System (AMPS) used in the North America and Australia, Total Access Communications System (TACS) in the United Kingdom, C-450 in West Germany, Portugal and South Africa, Radiocom 2000 in France, and RTMI in Italy. In Japan there were multiple systems. Three standards, TZ-801, TZ-802, and TZ-803 were developed by NTT DoCoMo a Japanese network operator, while a competing system operated by DDI used the Japan Total Access Communications System (JTACS) standard. Table 2.1 shows some characteristics of the AMPS, NMT (450) and NMT (900). Note that all three networks used Frequency Modulation which has now been replaced by more sophisticated modulation schemes today [15].

First generation (1G) mobile systems suffered from many disadvantages such as lack of security e.g. there was no data encryption due to the analogue nature of the signals. In addition 1G network suffered from interference and poor voice quality hence the need to replace them with 2G technology. The table below shows the characteristics of earlier cellular networks [15].

Parameters	**AMPS**	**NMT(450)**	**NMT(900)**
Frequency of Transmission • BS (Base Station) • MS (Mobile Station)	869-894 824-849	463-467.5 453-457.5	935-960 890-915
Frequency separation - Transmit and Receive (MHz)	45	10	45
Channel Spacing (KHz)	30	25	25
Number of channels	832	180	1000
Modulation • Frequency deviation (KHz)	FM +-12	FM +-5	FM +-5
Control signal • Modulation • Frequency Deviation	FM +-8	FM +-3.5	FM +-3.5

Table 2.1: Characteristics of the AMPS and NMT cellular systems [15]

2.3 Second Generation (2G)

2G is an acronym for second-generation cellular technology. Second generation cellular networks were commercially launched on the Global System for Mobile Communications (GSM) standard in Finland by Radiolinja (now part of Elisa Oyj) in 1991. Three primary benefits of 2G networks over their predecessors were that phone conversations were digitally encrypted; 2G systems were significantly more efficient on the spectrum facilitating far greater mobile phone penetration levels; and 2G introduced data services for mobile, starting with SMS text messages [16].

After 2G was launched, the previous mobile telephone systems were retrospectively named 1G. While radio signals on 1G networks were analogue, radio signals on 2G networks are digital. Both systems use digital signalling to connect the radio towers to the rest of the telephone system.

2G has been superseded by newer technologies such as 2.5G, 2.75G, 3G, and 4G; however, 2G networks are still used in many parts of the world.

2G technologies can be divided into TDMA-based and CDMA-based standards depending on the type of multiplexing used. The main 2G standards are [16]:

- Global System for Mobile communication (GSM, TDMA-based), originally from Europe but used in almost all countries on all six continents. Today, GSM accounts for over 80% of all subscribers around the world. Over 60 GSM operators are also using CDMA2000 in the 450 MHz frequency band (CDMA-450).
- IS-95, cdmaOne (CDMA-based, commonly referred as simply CDMA in the US), used in the Americas and parts of Asia. Today it accounts for about 17% of all subscribers globally. Over a dozen CDMA operators have migrated to GSM including operators in Mexico, India, Australia and South Korea.
- PDC (TDMA-based), used exclusively in Japan
- iDEN (TDMA-based), proprietary network used by Nextel in the United States and Telus Mobility in Canada
- IS-136 aka D-AMPS (TDMA-based, commonly referred as simply 'TDMA' in the US), was once prevalent in the Americas but most operators that deployed it have migrated to GSM.

2G services are frequently referred as Personal Communications Services, or PCS, in the United States. GSM is a digital mobile telephony system that is widely used in Europe and other parts of the world. GSM uses a variation of time division multiple access (TDMA) and is the most widely used of the three digital wireless telephony technologies (TDMA, GSM, and CDMA). GSM digitizes and compresses data, then sends it down a channel with two other streams of user data, each in its own time slot. It operates at either the 900 MHz or 1800 MHz frequency band [16].

2.3.1 Global System for Mobile Communication (GSM)

Mobile services based on GSM technology were first launched in Finland in 1991. Today, more than 690 mobile networks provide GSM services across 213 countries and GSM represents 82.4% of all global mobile connections. According to GSM World, there are now more than 3 billion GSM mobile phone users worldwide. GSM World references Asia as "the largest single GSM market, with more than 500 million users, followed by Russia with 200 million, India with over 83 million and the USA with 78 million users" [17].

Since many GSM network operators have roaming agreements with foreign operators, users can often continue to use their mobile phones when they travel to other countries. SIM cards (Subscriber Identity Module) holding home network access configurations may be switched to those with metered local access, significantly reducing roaming costs while experiencing no reductions in service [13].

GSM, together with other technologies, is part of the evolution of wireless mobile telecommunications that includes High-Speed Circuit-Switched Data (HCSD), General Packet Radio System (GPRS), Enhanced Data GSM Environment (EDGE), and Universal Mobile Telecommunications Service (UMTS).

2.4 Third Generation (3G)

Cellular technologies specified by the Third Generation Partnership (3GPP) are the most widely deployed in the world, with over 2.6billion users in 2008[18]. The latest advancement in 3GPP cellular technology is the Long-Term Evolution (LTE) and an evolved packet access core network in the System Architecture Evolution (SAE) [19].

The advent of 3G and the higher bandwidth radio interface of UTRA created possibilities for a range of new services that were only ideas with 2G and 2.5G networks. The 3G radio access development today is handled by 3GPP [19].

The development of 3G was enhanced by the internalization of cellular standardization. GSM was initially a pan-European project but attracted worldwide interest. As the GSM standard gained popularity, it created economies of scale since the product market was larger. This led to a much more organised international cooperation around the standardization of 3G and beyond than earlier generations [19].

Work on 3G started in the ITU in the 1980s.The ITU-R issued a first recommendation defining Future Public Land Mobile Telecommunications Systems (FPLMTS) in 1990, later revised in 1997. The name for 3G within ITU had by then changed to IMT-2000. The World Administrative Radio Congress (WARC-92) identified 230MHz of spectrum for IMT-2000 on a worldwide basis. With this, the stage was set to specify IMT-2000 [19].

ITU-R had several task groups that defined different processes. Task Group 8/1 defined the process of evaluating IMT-2000 technologies in ITU-R recommendation M.1255. The evaluation criteria set the target data rates for the 3G circuit switched and packet switched data services as follows [20]:

- Up to 2Mbps in an indoor environment
- Up to 144kbps in a pedestrian environment
- Up to 64kbps in a vehicular environment

These figures became the benchmark data rates for 3G, however today; data rates for 3G are well beyond 2Mbps.

2.5 Fourth generation (4G)

4G is the fourth generation of cellular wireless standards. It is a successor to 3G and 2G families of standards. Speed requirements for 4G service have been set at a peak download speed of 100 Mbps for high mobility communication (fast moving vehicles) and 1 Gbps for low mobility communication (such as pedestrians and stationary users) [21].

A 4G system is expected to provide a comprehensive and secure all-IP based mobile broadband solution to laptop computer wireless modems, smart phones, and other mobile devices. Facilities such as ultra-broadband Internet access, IP telephony, gaming services, and streamed multimedia will be provided to users [22].

At present the candidate technologies for 4G are IEEE's mobile WiMAX and 3GPPs Long term evolution (LTE). These have been on the market since 2006 and 2009 respectively, and are often branded as 4GThe current versions of these technologies did not fulfil the original ITU-R requirements of data rates of up to 1 Gbps for 4G systems. However, marketing materials use 4G as a description for Mobile-WiMAX and LTE in their current forms [23].

IMT-Advanced compliant versions of the above two standards are under development and are called LTE Advanced and Wireless MAN-Advanced respectively. ITU has decided that LTE Advanced and Wireless MAN-Advanced should be accorded the official designation of IMT-Advanced. On December 6, 2010, ITU announced that current versions of LTE, WiMAX and other evolved 3G technologies that do not fulfil IMT-Advanced requirements could be considered '4G', provided they represent forerunners to IMT-Advanced and "a substantial level of improvement in performance and capabilities with respect to the initial third generation systems now deployed" [24].

In all suggestions for 4G, the CDMA spread spectrum radio technology used in 3G systems and IS-95 is abandoned and replaced by OFDMA and other frequency-domain equalization schemes. This is combined with MIMO (Multiple Input Multiple Output), e.g., multiple antennas, dynamic channel allocation and channel-dependent scheduling.

2.5.1 Drivers of 3G Evolution

For any business to be successful, it is important to take into account the driving forces of the business for the future. This is especially true for the mobile communications industry where subscriber numbers grow rapidly and global technologies have attracted many players who all want a stake in the lucrative industry. New operators and vendors try to compete with incumbents by adopting new technology and standards to provide better services at a lower cost. The existing operators and vendors in turn follow suit, creating a vicious cycle of competition which is a key driving force in technology evolution.

The development in digital technology such as digital cameras, visual displays, etc. fosters the development of better mobile communication services. In order to keep up with other areas of digital technology, the mobile communications systems need to upgrade or even be replaced by new technologies. Similarly the advancement in digital technology enables new and more powerful systems that not only provide new services but can also provide the existing services at a lower cost than the 2G and PSTN. Some of the driving forces for 3G evolution are [25];

- Staying competitive: operators must continuously improve their services in order to stay relevant in the communications market
- Services: Easier and better provisioning of services both old and new. E.g. the migration from CS1 to CS2 and now the more intelligent networks has created better services that are easy to manage and provision.
- Cost: more cost effective provisioning of new and old services. Technology advancement is necessary to provide new and more advanced services at a reasonable cost. Advancements in electronics and optics have made it possible to build devices that were not possible 20 years ago. This advancement has enabled faster computing and smaller devices at a lower cost.
- Internet and IP technology: The success of Internet and IP-based services over the internet has gone wireless. Subscribers require ubiquitous IP services and are thus putting high demands on the network operators. This has driven mobile operators to create conventional IP services over mobile networks.

2.5.2 3GPP Radio Access Network Evolution

The 3rd Generation Partnership Project (3GPP) is partnership between groups of telecommunications organisations, to make a global third-generation (3G) mobile phone system specification within the scope of the International Mobile Telecommunications-2000 project of the

International Telecommunication Union (ITU). 3GPP specifications are based on evolved GSM specifications. 3GPP standardization encompasses Radio, Core Network and Service architecture. 3GPP has several Technical Specification groups (TSGs) which deal with specific working areas [26].

In Q3 of 2004, TSG RAN[2] organised a workshop on 3GPP long-term evolution, this was the starting point of the development of the LTE radio interface. After the initial phase where targets and objectives of LTE were set, the TSG SA launched a corresponding work on the SAE, since it was important that LTE radio interface have suitable evolved system architecture. As HSPA was an evolution of WCDMA, building upon the basic WCDMA structure and with strong requirements on backwards compatibility to leverage on already existing networks, 3GPP has specified LTE as the RAN of the future. LTE targets more complex spectrum situations and has fewer restrictions on backward compatibility. 3G evolution therefore consists of two parallel tracks each with its own merits. Figure 2.1below illustrates the evolution of LTE over the different 3GPP releases over the years [27].

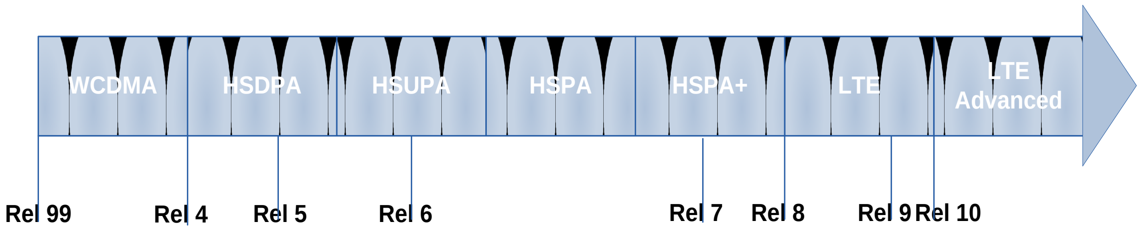

Figure 2.1: Summary of RAN evolution

- 3GPP Release 99/Release 4 was the first release of the third generation specifications was essentially a consolidation of the underlying GSM specifications and the development of a new UTRAN radio access network. A foundation was laid for future high-speed traffic transfer in both circuit switched and packet switched modes. A comprehensive review of the Release 1999 features is held in the summary of all Release 99 features document, the latest version being TR 21.101. Release 99 features can be summarized as follows [18];
  - Multiple access method: CDMA
  - Duplexing method: FDD and TDD
  - Channel Bandwidth: 5 MHz/1.6 MHz
  - Chip rate: 3.84 Mcps/ 1.28 Mcps
  - Speech Coding: AMR

[2] Technical Specification Group Radio Access Network (TSG RAN) is a group within 3GPP that is responsible for the definition of the functions, requirements and interfaces of the UTRA/E-UTRA network in its two modes, FDD & TDD. More precisely: radio performance, physical layer, layer 2 and layer 3 RR specification in UTRAN/E-UTRAN; specification of the access network interfaces (Iu, Iub, Iur, S1 and X2); definition of the O&M requirements in UTRAN/E-UTRAN and conformance testing for User Equipment and Base Stations.

    - Modulation: QPSK
    - User bitrates: 384 kbps PS, 64 kbps CS
    - Services: mobile TV, video calling
- 3GPP Releases 5 and 6 define the basis for mobile broadband access. In release 5, the following features were introduced were HSDPA: 16QAM, 14.4 Mbps on the downlink 0.4Mbps on the uplink
- 3GPP Release 6: HSUPA with 5.76 Mbps on the uplink, 14Mbps on downlink
- 3GPP Release 7: HSPA+ Downlink MIMO, 64QAM, Uplink 16QAM, Peak data rates of 28 Mbps on Downlink, 11.5Mbps on Uplink
- 3GPP Release 8 : DC-HSDPA, 42.2 Mbps on the downlink and 11Mbps on the uplink
- 3GPP Release 9: DC-HSDPA + MIMO on the uplink and 84 Mbps on the downlink bps and DC-HSUPA on the uplink
- 3GPP Release 10: 168 Mbps and 4-carrier HSDPA on the downlink

From the above release descriptions, it is apparent that a lot of work has been done from the advent of GSM to LTE. The first set of LTE release 8 specifications were finalized in the last quarter of 2007, but it was only until 2009 that the LTE commercial specifications were made. Work on LTE Advanced aims to happen with release 10 and finalization is expected in Q1 and Q2 of 2011.

2.5.3 Why move to LTE?

The LTE Radio Access Network (also called the Evolved UMTS Terrestrial Radio Access Network ,E-UTRAN), is expected to substantially improve end-user throughputs, network capacity and reduce user plane latency, bringing a significantly improved user experience with seamless mobility. With the emergence of Internet Protocol (IP) as the protocol of choice for carrying all types of traffic, LTE will provide support for IP-based traffic with end- to-end Quality of Service (QoS). Voice traffic will be supported mainly as Voice over IP (VoIP), enabling better integration with other multimedia services. Initial deployments of LTE are expected by 2011 with commercial availability on a larger scale expected 1-2 years later. A key motivation is that LTE delivers on two separate axes. Previously, new network technologies mainly focused on improved performance. LTE however, not only delivers substantial performance improvements, but also creates new business models for operators as follows [18] [25]:

- LTE enables new applications: With expected throughput in excess of 100Mbps and latency lower than 10ms, LTE will provide mobile subscribers with a user experience very comparable to what they have at home today, with xDSL and cable connections. In addition, LTE breaks the boundaries between "home and outside", meaning that applications can be shared between the home computer and outside the home. In addition because of LTE's lower cost per bit, it also makes a number of typically gigabyte hungry applications, cost effective and viable to use in a mobile environment [28].

- LTE has a future-proof data rate: Do people really need the kind of bandwidth LTE provides? Studies show different usage case scenarios in 2-3 years' time (mobile, laptop-nomadic in hotspot and household via xDSL) that the maximum bandwidth needs can be achieved with LTE whereas 2G and 3G wireless technologies would struggle to deliver enough capacity. 2G and 3G technologies are more uplink limited.
- Seamless mobility: LTE is a key technology in realizing the full promise of seamless mobility. The LTE, IP Based Evolved Packet Core, allows for connectivity and hand-over to other access technologies, including all 3GPP & 3GPP2 technologies, such as GSM, UMTS, CDMA/EV-DO, as well as Wi-Fi or even fixed line broadband services like Digital Subscriber Line (DSL) , Gigabit Passive Optical Network (GPON). And, the IMS layer or service layer will provide the user seamless continuity of services.
- Better business model: With greater spectrum efficiency, simpler architecture and the ability to re-use low frequency spectrum, LTE will boast of much improved capacity for both voice and data delivered at a significantly lower cost compared to legacy technologies. These improvements contribute to a lower cost per bit for both voice and data services. In fact, some simulations show that voice services on UMTS are several times more expensive than LTE. The relative total cost of ownership (TCO) for LTE by subscriber's GB/month also presents significant improvement opportunities over existing 3G networks [7][29].

2.6 Conclusion

A basic introduction to cellular network evolution over the past two decades has been given in this chapter. The chapter briefly talks about characteristics of different generations of cellular networks from earlier days and how they have changed along the years. Characteristics include the frequency band used, modulation and coding schemes and network elements, to mention but a few. It shows that as society changes, there is need for more sophisticated telecommunications services, which fosters network evolution. As demand grows, it ceases to be simply about providing the service but how efficient and effective the service is and how an operator can keep a competitive edge while providing the service. The need for seamless handover across networks is therefore one of the demands that come along with evolution.

3 LTE and UMTS Networks

3.1 Introduction

This chapter gives an overview of UMTS and LTE/EPC architecture, the logical network elements and interfaces, radio procedures and handovers. UMTS and LTE utilize network architecture similar to second generation networks. As in previous generations, the network is grouped into the Radio Access Network (RAN) and the Core Network (CN). UMTS radio network introduces several new protocols and designs based on the Wideband Code Division Multiple Access (WCDMA) radio access technique while the LTE network is based on the Orthogonal Frequency Division Multiple Access (OFDMA) technique. In this chapter, UMTS is first discussed then it is followed by a discussion on LTE.

3.2 Universal Mobile Telecommunications System (UMTS)

The UMTS architecture is shown in Figure 3.1. It shows the UE, the base station (known as the Node B), the Radio Network Controller (RNC) which together make up the UTRAN. Also shown are the Core Network and the associated interfaces. UMTS entities shown in this figure are briefly described as follows [30];

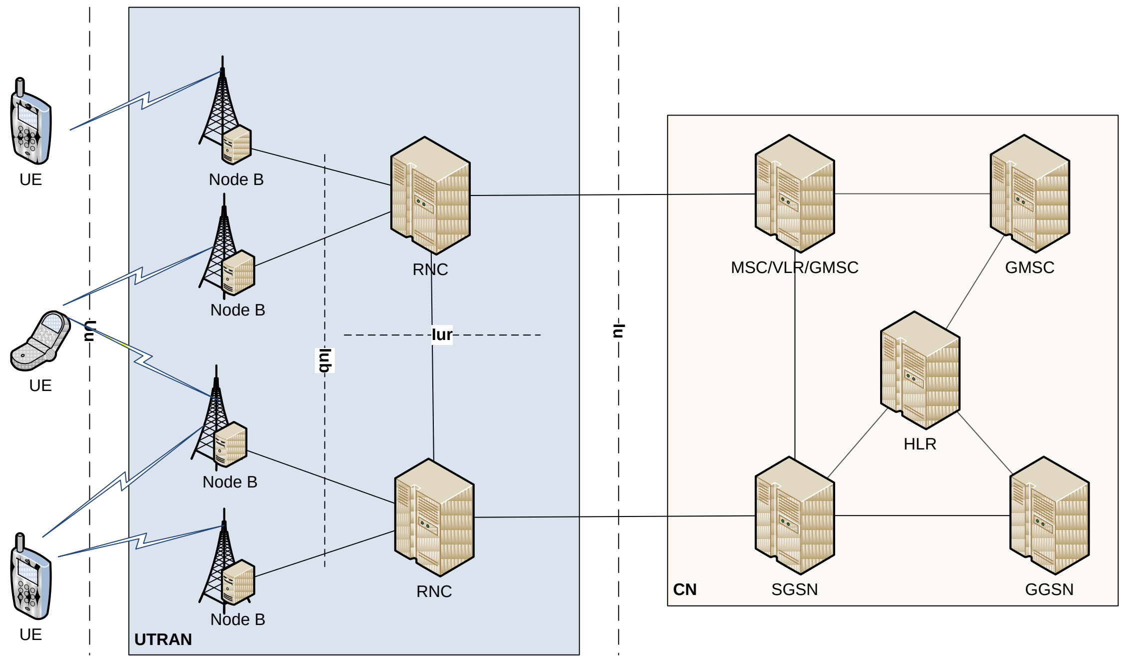

Figure 3.1: UMTS Network Architecture

- The UE: Consists of the Mobile Equipment (ME) and Universal Subscriber Identity Module (USIM).The ME is a radio terminal used for radio communication over the Uu interface. The Uu interface allows for communication between the UE and the UTRAN. The USIM is a smartcard that holds subscriber identity and performs authentication algorithms.
- Node B: Is the UMTS base station and converts the data flow between the Iub and Uu interfaces. The Iub interface is used to carry messages between the NodeB and the RNC. The Node B also performs radio resource management functions.
- The Radio Network Controller (RNC): controls radio resources (Node Bs).
- Home Location Register (HLR): is a database in the CN that stores the master copy of the user's service profile.
- Mobile Services Switching Centre/Visitor Location Register (MSC/VLR) is the switch and database that serve the UE in its current location for Circuit-Switched services. The MSC switches the CS services while the VLR holds a copy of the visitor's user profile.
- Gateway MSC (GMSC) is the switch through which the UMTS connects to other CS networks.
- Serving GPRS Support Node (SGSN) is similar to MSC/VLR except it is used for packet switched services.
- Gateway GPRS Support Node (GGSN) is similar to the GMSC but in relation to packet switched services.

3.2.1 WCDMA Air Interface

This section presents the basic principles of WCDMA air interface, particularly those features that differentiate WCDMA from GSM and LTE. Main parameters are briefly introduced in section 3.2.2 while handovers in WCDMA are discussed in section 3.2.7.

3.2.2 Main Parameters in WCDMA

In this section some system design parameters of WCDMA are presented and briefly explained. They are summarized in the Table 3.1.

WCDMA is a wideband Direct-Sequence Code Division Multiple Access (DS-CDMA) system. That is to say user information bits are spread over a wide bandwidth by multiplying the user data with quasi random bits/chips derived from CDMA spreading codes. To support very high bit rates, the use of a variable spreading factor and multi-code connections is supported [7].

Property	Description
Multiple access method	DS-CDMA
Duplexing method	Frequency division duplex/time division duplex
Base station synchronization	Asynchronous operation
Chip rate	3.84 Mcps
Frame length	10ms
Service multiplexing	Multiple services with different quality of service requirements multiplexed on one connection
Multi-rate concept	Variable spreading factor and multi-code
Detection	Coherent using pilot symbols or common pilot
Multiuser detection, smart antennas	Supported by the standard, optional in the implementation

Table 3.1: Main WCDMA Parameters [7]

The chip rate of 3.84Mcps leads to a carrier bandwidth of approximately 5MHz. DS-CDMA systems with a bandwidth of 1MHz e.g. IS-95 are referred to as narrowband CDMA systems. WCDMA supports higher user data rates and has certain performance benefits such as increased multipath diversity. WCDMA supports highly variable user data rates, i.e. Bandwidth on Demand (BoD) is well supported. The user data rate is kept constant during each 10ms frame. However, the data capacity among users can change from frame to frame. WCDMA supports both TDD and FDD modes [7].

WCDMA air interface allows for advanced CDMA receiver concepts such as multiuser detection and smart adaptive antennas. These can be used by the network operator to increase capacity and/or coverage. WCDMA is designed to be deployed in conjunction with GSM and higher generation cellular networks; therefore handovers between GSM and WCDMA are supported in order to be able to leverage the GSM coverage for the introduction of [7].

3.2.3 Generic Principles of CDMA Operation

Figure 3.2 illustrates the basic operations of spreading and dispreading for a DS-CDMA system. User data is assumed to be a BPSK-modulated bit sequence of rate R, the user data bits assuming the values of + OR - 1. The spreading operation multiplies each user data bit with a sequence of 8 code bits/chips. This is also assumed for the BPSK spreading modulation. Thus the resulting spread data is at a rate 8Xr and has a random appearance as the spreading code. The resulting code is then sent over the channel to the receiver. At the receiver, the spread sequence is multiplied by the 8 code chips used during the spreading of these bits. The original sequence is then recovered [18].

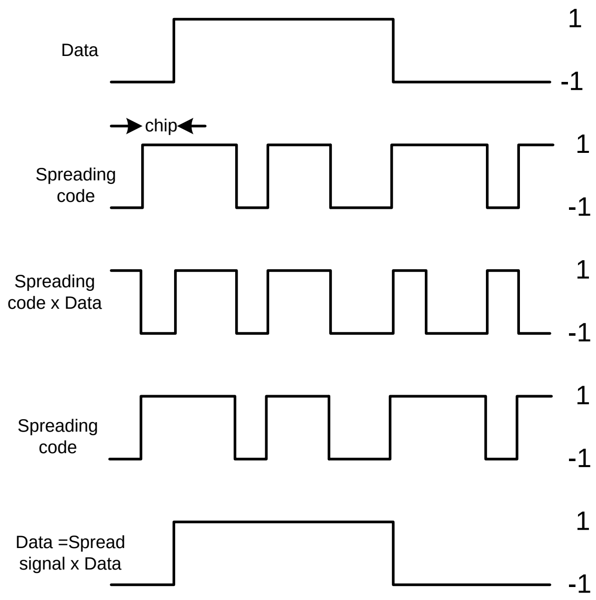

Figure 3.2: Spreading and De-spreading principal of DS-CDMA

Increasing the signalling rate by a factor of 8 results in the widening of the occupied spectrum by the same factor, hence the term; “Spread Spectrum”. Dispreading restores the bandwidth proportional to R for the signal. Due to spreading and dispreading, the carrier to interference(C/I) ratio can be lower in CDMA than GSM. Since the wideband signal can be below the thermal noise level, its detection is difficult without knowledge of the spreading sequence. Thus, spread spectrum systems originated in military applications where secrecy is paramount. Spreading/dispreading in itself does not provide any signal enhancement for wireless systems. The benefits of WCDMA are rather realized indirectly [18];

- The processing gain together with the wideband nature allows a frequency reuse of 1 between different cells of wireless systems. This feature allows increased spectral efficiency.
- Several users sharing the same wideband carrier provides interferer diversity. That is to say the multiple access interference from many system users is averaged out. This boosts capacity and reduces the need to plan for worst case interference.
- The above two benefits require the use of tight power control and soft handover to avoid one user’s signal blocking the others. This is further described in Sections 3.2.7 and 3.2.8.
- With a wideband signal, the different propagation paths of a wireless radio signal can be resolved at higher accuracy than with signals at a lower bandwidth. This results in higher diversity content against fading and thus improved radio performance.

3.2.4 General Protocol Model for UTRAN

The general radio interface protocol architecture is shown in Figure 3.3. The physical layer provides service to the MAC layer which in turn offers services to the RLC layer via logic channels. In this thesis, the chief interest lies in the Radio Link Control (RLC) protocol layer as the RLC protocol provides retransmission services for both user and control data and thus can be used to determine delay/latency over the radio link. On top of the RLC there is the Radio Resource Control (RRC). It is responsible for maintaining a reliable connection between the UE and the UTRAN and especially manages radio resources. It is also involved in managing handovers [7].

The Packet Data Control Protocol exists in the user plane and is used for services in the PS domain. It contains compression techniques required for better spectral efficiency in IP packets transmitted over the radio. The Broadcast Multicast Control (BMC) protocol is a user plane protocol designed to adapt broadcast and multicast services on the radio interface.

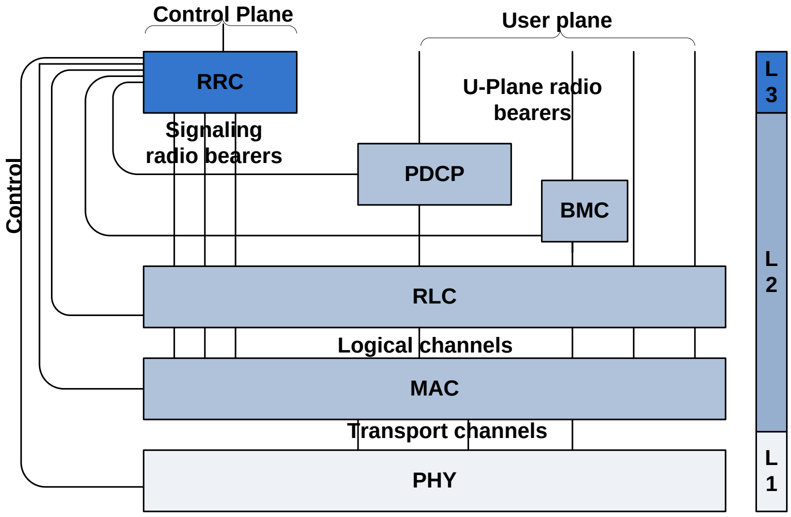

Figure 3.3: UTRA Radio Interface Protocol

3.2.5 Radio Link Control Protocol (RLC)

In a system such as UMTS, the RLC protocol is used to transfer data reliably across the network. Each RLC instance is configured by the RRC to operate in either transparent mode (Tr), unacknowledged mode (UM) or acknowledged Mode (AM). Tr and UM RLC are unidirectional while AM is bidirectional [31][32].

In this thesis, RLC-AM is the choice as it is more reliable. RLC layer lies at layer 2 of the UTRAN protocol architecture. The RLC - AM layer is important in UTRAN and performs the following functions [7];

- Transfer of user data: The protocol supports reliable data transfer. It can discard data that have not been successfully transmitted in a period of time or number of retransmissions. This is because it expects upper layer protocols to perform their own, end-to-end retransmissions as well.
- Error correction: The protocol detects erroneous frames and either asks for a retransmission or discards them altogether.
- Sequence number check, duplicate detection and in-sequence delivery of upper layer frames: The protocol detects duplicate frames by using sequence numbers and a frame window. It also delivers the frames in sequence. Out-of-sequence delivery is also supported.
- Protocol error detection and recovery: The protocol detects a number of abnormal conditions and can request retransmissions or even reset itself if all other measures fail.
- Flow control: The receiver can optionally change the sender's window size to control the flow
- Segmentation and reassembly, concatenation and padding: The super frames received from upper layers are segmented into smaller parts (if they are larger than the available space in a frame) and are concatenated to form frames of fixed size. Padding is added at the end of the frame if they are smaller than the indented size.
- Ciphering: The protocol encrypts and decrypts the data exchanged.

3.2.6 Radio Resource Control and Mobility Management

In WCDMA, RRC is a management responsibility done by UTRAN. RRC is located in the UE and RNC. RRC contains several algorithms aimed at keeping a high QoS. Figure shows the locations of the different RRC algorithms within the UTRAN [25].

RRM algorithms deliver information over the radio path, called the UTRA service. The protocol used for this service is the RRC protocol. The RRM contains several algorithms which include [25];

- Code management: Deals with the management of codes generated by the base station and assigned to users. Each user in a WCDMA network is assigned a unique code that it uses to access radio channels.
- Handover control: controls handover algorithms and procedures
- Power control: Used for Optimisation of power levels between the BS and UE in order to avoid interference among users.
- Admission Control and packet scheduling: Performs security and scheduling functions.

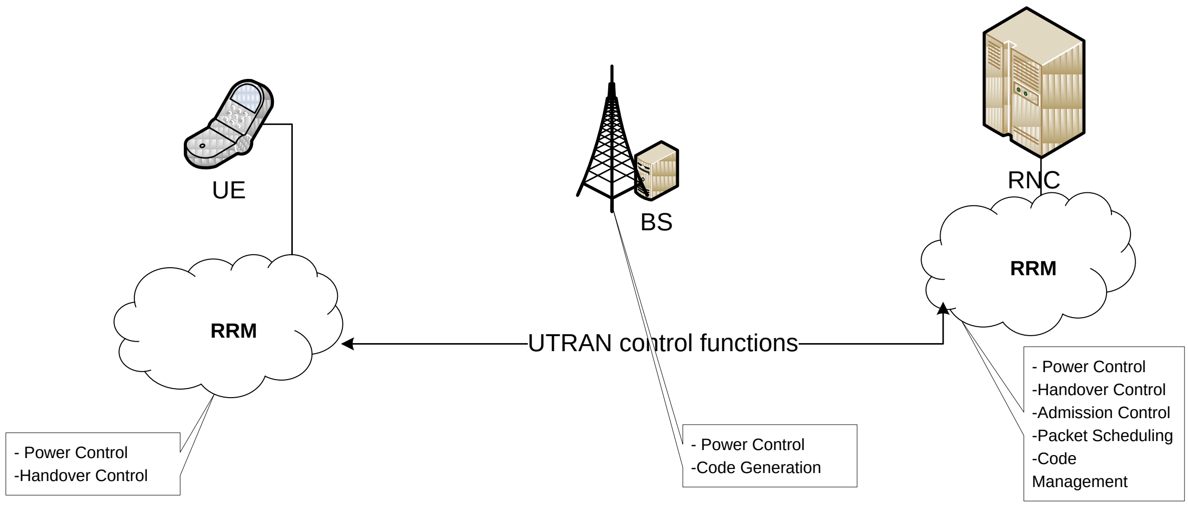

Figure 3.4: Radio Resource Management and Radio Resource Control functions

In this research report the RRM algorithms of major concern are Handover and Power Control.

UTRAN level Mobility Management refers to those functions which RNC handles in order to keep the UE in touch with the UTRAN radio cells, taking into account the user's mobility within the UTRAN and the type of traffic or Radio Access Bearer(s) it is using. The concept of UTRAN mobility is based on a cell, UTRAN Registration Area (URA), Radio Network Temporary Identifier (RNTI) and the Radio Resource Control state transition model [25].

3.2.7 Handover Control in WCDMA

Handover is one of the most important ways to guarantee user mobility in a mobile communications network. Handover is the process of maintaining an active connection with a moving subscriber. The basic concept is when a subscriber moves from one cell to another, connectivity to the old cell is released while a new connection is made with the target cell. Handover control is a complicated issue in cellular systems and especially in CDMA systems [33].

There are several reasons why handover is activated. First is that the air interface connection does not fulfil the desired criteria set for it anymore and thus the UE or the UTRAN initiates actions to improve the connection. In WCDMA, real-time handover is used in circuit switched calls. In the case of packet switched calls, handovers are mainly achieved when neither the network nor the UE has any packet transfer activity. Regardless of the type of handover, criteria that are used to decide the need for handover are performed. Execution of the handover depends on the strategy implemented in the system. However, most criteria for handover are signal quality, user mobility, traffic distribution and bandwidth utilisation [33].

- Signal Quality handover occurs when the quality of the signal strength falls below certain parameters specified in the RNC. Signal deterioration is detected by the periodic signal measurements carried out by the UE and the Node B. Handover due to poor signal quality may be done on either the uplink or the downlink.
- Handover due to Traffic occurs when the traffic capacity of a cell has reached its maximum or approaches it. In this case, if a UE is close to the edge of a cell with high load, it may be handed over to neighbouring cells with less traffic load. This allows the system load to be distributed more uniformly.

The number of handovers is dependent on how mobile a subscriber is. If the UE is moving fast in the same direction, it can be assumed that it will have more handovers to the UTRAN. To avoid unwanted handovers, a UE with high speed may be handed over from microcells to macrocells. Slow moving UEs can be handed over to microcells. The decision to perform a handover is always made by the RNC currently serving the UE, except in the case of handover due to traffic reasons. In the latter case, the MSC may also make the decision. Other reasons such as change in service may also lead to a handover [33].

3.2.7.1 Handover Process

The figure below illustrates the basic handover process consisting of three main phases namely measurement, decision and execution. The handover procedure discussed here is specifically for WCDMA although some principles apply to other cellular systems [18][25].

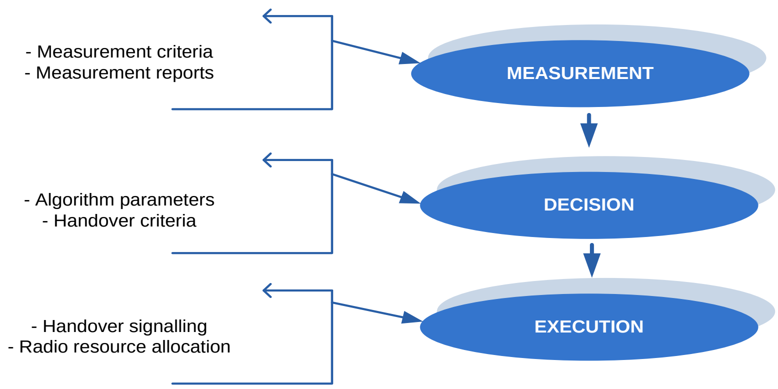

Figure 3.5: Basic Handover process

Handover measurement provisioning is an important task for good system performance for two reasons [33].

- Signal strength of a radio channel may vary drastically due to fading and path loss as a result of user mobility and cell environment.

- Excess measurement reports by the UE or handover execution by the network increases network signalling which is undesired.

The UE constantly measures signal strength of neighbouring cells and reports to the RNC. 3GPP TS 25.331 specifies the reported measurements and groups them in different categories. The different measurements are [34]:

- Intra-frequency measurements which look at strength of the downlink physical channels for signals with same frequencies.
- Inter-frequency measurements which look at the strength of the downlink physical channels for signals with different frequencies.
- Inter-system measurements which measures the strength of the downlink physical channels belonging to another radio access system other than UTRAN e.g. GSM.
- Traffic volume measurements contain measurements for uplink traffic volume.
- Quality measurements include quality parameters e.g. downlink transport block error rate.
- Internal measurements of UE transmitted power and received signal level.

The measurement events, on the other hand may be triggered based on [34];

- Change of best cell
- Change in the Primary Common Pilot Channel (CPICH) signal level
- Change in the P-CCPCH signal level
- Changes in the Signal-to-Interference (SIR) level
- Changes in the Interference Signal Code Power (ISCP) level
- Periodical reporting
- Time-to-trigger

WCDMA specifications provide various measurement criteria to support handover mechanisms in the system. It is important to choose the best measurement procedure for good system performance. Handover signalling load can be optimised by fine-tuning trade-off between handover criteria, handover measurements and the traffic model used in network planning [34].

The decision phase consists of assessing the overall QoS of the connection and comparing it with the requested QoS attributes and estimates from neighbouring cells. Depending on the outcome of the comparison, the handover procedure may or may not be triggered. The SRNC checks whether the values indicated in the measurement reports trigger criteria were set. If they trigger, then it allows handover execution. Regarding handover decision-making, there are two main types of handover [34];

- Network Evaluated Handover (NEHO)
- Mobile Evaluated Handover (MEHO)

For an NEHO procedure, the network SRNC makes the handover decision while with the MEHO approach, the UE prepares the handover decision. For joint handover, the decision is made by both UE and the SRNC. It's imperative to note that even with an MEHO; the final decision is made by the SRNC. This is because the RNC is responsible for the overall RRM of the system and is thus aware of the system's overall load and other information required for handover decision.

Handover decision-making is based on measurements from the UE and BS as well as handover algorithm criteria. Advanced handover algorithms may be utilised based on available parameters and measurement capabilities within the network. The general principles of a handover algorithm are shown in Figure 3.6 below; in here, it is assumed that the decision-making is based on the pilot signal strength reported by the UE. The following parameters are used [34];

- Upper threshold: the level at which the signal strength of the connection is at the maximum acceptable level in respect to the requested QoS.
- Lower threshold: minimum acceptable level of the signal the signal strength of the connection.
- Handover margin: pre-defined parameter, where the signal strength of the neighbouring cell (B) starts to exceed the signal strength of current cell by a certain amount of a certain time.
- Active set: list of cells through which the UE has simultaneously connection to the UTRAN.

Consider a UE camping on cell A moving towards cell B. As the UE moves towards cell B, the pilot signal (A) deteriorates, approaching the lower threshold as in Figure 3.6. This results in handover being triggered during the following steps [34]:

- Signal strength of A becomes equal to defined lower threshold. Based on measurements UE also recognises that B has better signal strength. It adds B to the active set. UE thus has simultaneous connections to the UTRAN and benefits from the summed signal from both A and B.
- When the signal of B becomes better than A, the RNC keeps this information and starts calculating the handover margin calculation.
- The strength of signal B becomes much better than the defined lower threshold. This is sufficient to satisfy the required QoS. The strength of the summed signal exceeds the upper threshold causing interference. As a result the RNC deletes signal A from the Active set.

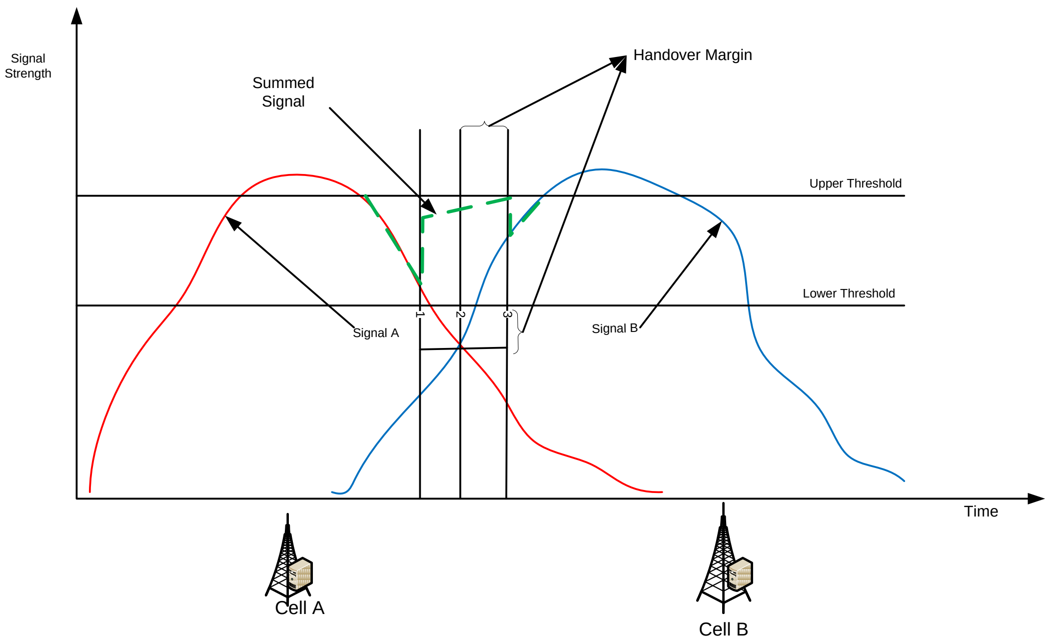

Figure 3.6: General principles of handover algorithms

The size of the active set varies but usually ranges from 1 to 3 signals. Due to the random nature of the UE, it is possible for it to return to cell A after handover. This results in the aptly named *Ping-Pong* effect which is harmful to system capacity and performance. Using a margin or hysteresis parameter is to avoid undesired handover and signalling to the UTRAN [34].

Handover mechanisms may be classified as hard handover or soft handover. Within these two categories lie other types of handover. The following sections describe the different handover types in UMTS [7].

3.2.7.2 **Handover Decision Algorithms**

Traditional handover algorithms can be classified as follows [35] [36];

- Handover based on Relative Signal Strength (RSS) in which the strongest base station (BS) is selected at all times.
- Handover based on Relative Signal Strength with threshold (RSS-T) in which a user handover is executed only if the current signal is sufficiently weak (less than a threshold) and the other is the stronger of the two.
- Handover based on Relative Signal Strength with hysteresis (RSS-H) in which a user handover is done if the new BS is sufficiently stronger by hysteresis margin than the current one. This method prevents repeated handovers (Ping-Pong effect).

- Handover based on Relative Signal Strength with Hysteresis and Threshold of serving base station (RSS-HT) in which the user handover to a new BS occurs only if the current signal level drops below a threshold and the target BS is stronger than the current one by a given hysteresis margin.
- Handover based on prediction techniques: in which the handover decision is made on the expected future value of the received signal. In cellular system wrong handover may occur, this can be reduced by delaying the occurrence of handover until the new BS signal strength gets sufficiently stronger. To achieve this, an additional criterion of absolute signal strength considered as threshold of a new BS has been involved in the signal strength based on the RSS-H algorithm. The resultant algorithm is termed as RSS-HTnew. This algorithm improves the performance as follows[36][35]:
    - With the proper setting of the new BS threshold it reduces the number of unnecessary handovers to a new base station when the signal strength of the new BS is not sufficient to serve the call.
    - With appropriate higher threshold setting, the number of handovers occurring to the neighbouring cell not intended for handover (wrong handover) can be minimized.

3.2.7.3 **Hard Handover**

In this type of handover, the old connection is released before making a new connection. This type of handover can be further divided into inter-frequency and intra-frequency hard handovers. In this type of handover, not only is there a lack of simultaneous signals but also a very short cut in the connection indistinguishable to the user.

3.2.7.3.1 Inter-frequency and Intra-frequency Handover

For inter-frequency hard handover the carrier frequency of the new radio access is different from the old carrier frequency to which the UE was connected. On the other hand, if the new carrier, to which the UE is accessed after the handover procedure is the same as the original carrier then it is termed as an intra-frequency handover. Below is an illustration of the two types of hard handover.

In Figure 3.7, the neighbouring RNC is not connected by the Iur interface (for communication between two RNCs) due to the radio network planning strategy or transmission reasons and thus an inter-RNC soft handover is not possible. Under such circumstances, intra-frequency hard handover is the only handover to support the seamless radio access connection and subscriber mobility from the old BS to the new BS. This results in an inter-RNC handover in which the MSC is also involved.

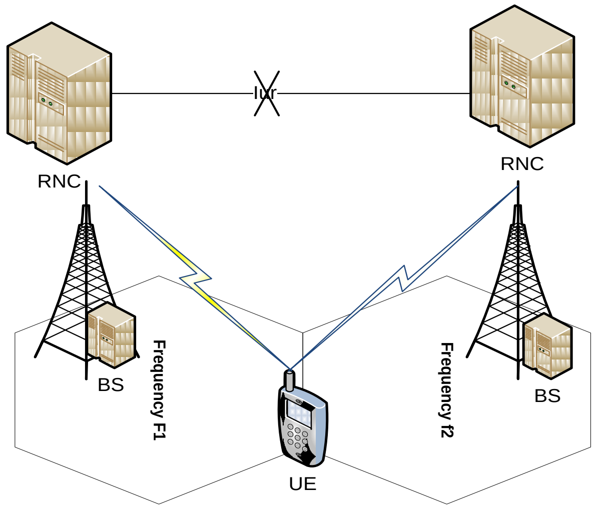

Figure 3.7: Intra-frequency hard handover

The frequency reuse factor for WCDMA is one, this implies that all BS's transmit on the same frequency and all UEs share a common frequency within the network. This does not however imply that frequency reuse is not utilised in WCDMA networks. If different carriers are allocated to cells for some reason, inter-frequency hard handover is required from one cell to another in a cell cluster [25].

Inter-frequency handover also occurs in a Hierarchical Cell Structure (HCS) network between separate cell layers, for example, between macro and micro cells which use different carrier frequencies within the same coverage area. In this case, the inter-frequency handover is used not only because the UE would otherwise lose its connection to the network but also in order to increase the system performance in terms of capacity and QoS. Inter-frequency hard handover is always an NEHO.

3.2.7.3.2 Inter-system handover

Inter-frequency handover may happen between two different RANs e.g. LTE and WCMA. In this case it can be called inter-system or inter-RAT handover as illustrated in Figure 3.8. This type of handover is the subject of this research report.

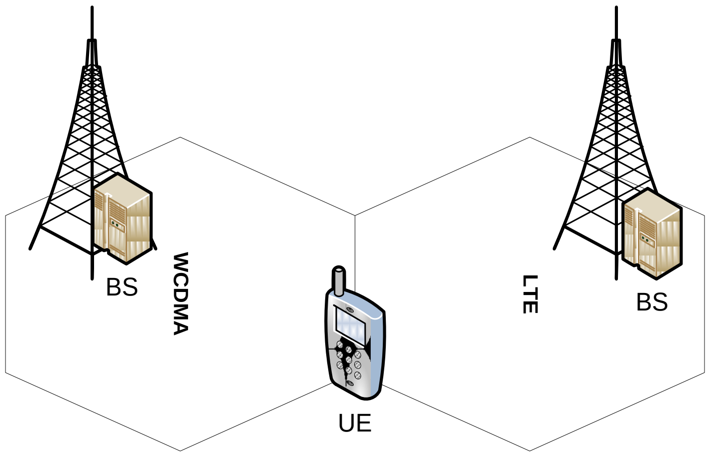

Figure 3.8: Inter-system hard handover

The ability to perform an inter-RAT handover in WCDMA is enabled by a special function mode known as, *Compressed Mode or Slotted Mode.* In WCDMA when the UE is in slotted mode, the spreading factor value of the channel can be reduced. Consequently, the radio interface connection uses only part of the space in the WCDMA frame slot. The rest of the slot can be used for other purposes by the UE; for example it may measure the surrounding GSM cells. This is the mechanism used to implement GSM/UMTS interoperability in UTRAN. Additionally, the slotted mode can be achieved by reducing the data rate using the higher layer controlling and reducing the symbol rate in association with the physical layer multiplexing. When the UE uses the Uu interface in this mode, the contents of the WCDMA frame are "compressed" in order to open a time window through which the UE is able to peek and decode the GSM BCCH information. Both WCDMA and GSM must send each other's identity information on the BCCH so that the UE is able to perform decoding properly [18][33].

Inter-system handover is best for areas where two RANs coexist. It is important to complement each other in order to ensure continuity of services. The inter-system handover can also be used to control load between the two networks. Inter-system handover is a NEHO; however the UE must have the capability to support this type of handover. The RNC recognises the possibility for an Inter-RAT handover based on the configuration of the radio network especially the neighbour definitions and other parameters [33].

No compressed mode is needed for making WCDMA measurements from GSM because GSM uses discontinuous transmission and reception. The service interruption time in the inter-system handover is 40ms maximum. The interruption time between the last received transport block on the old frequency and the time the UE starts transmission of the new uplink channel. The total service gap is slightly more than the interruption time because the UE needs to get the dedicated channel running in GSM. The service gap is typically below 80ms similar to that in intra-GSM handovers. Such a gap does not degrade voice quality and is illustrated in Figure 3.9 [7].

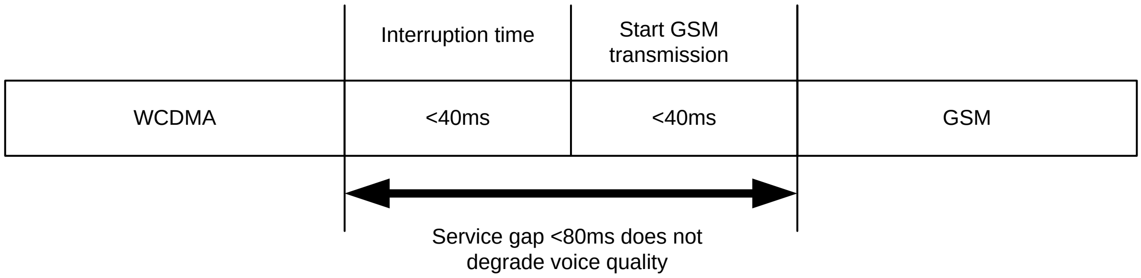

Figure 3.9: Service interruption time between GSM and WCDMA handover

3.2.7.4 **Soft and Softer Handovers**

Soft handover is when a new connection is established before the old connection is released [7]. In WCDMA, majority of handovers are intra-frequency soft handovers as illustrated in Figure 3.10.Soft handover is performed between two cells belonging to different BS's but not necessarily the same RNC. Either way, the involved RNC must coordinate the soft handover over the Iur Interface. Figure 3.10 shows soft handover. In this scenario, an MS is in the overlapping cell coverage area of two sectors belonging to different base stations [7].

Communication between MS and BS takes place concurrently via two air interface channels from each BS separately. In a soft handover event the source and target cells have the same frequency. In case of a circuit switched call, the terminal performs soft handovers almost all the time if the radio network environment has small cells. There are several variations of soft handover which include softer and soft-softer handovers. From the MS perspective, soft and softer handover are hardly different [7].

A softer handover is a handover by which a new signal is either added or deleted from the active set or replaced by a stronger signal within different sectors of the same BS. Figure 3.11 illustrates the phenomenon of a softer handover [7].

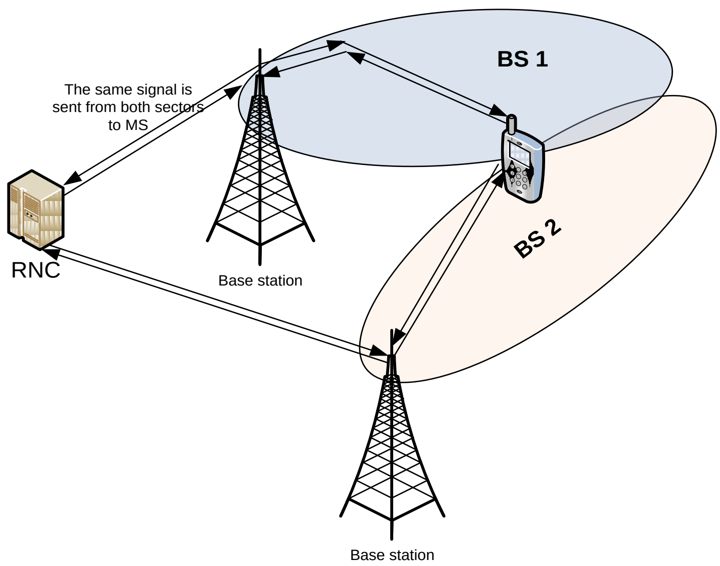

Figure 3.10: Intra-frequency soft handover

In softer handover, the BS transmits through one sector but receives from one or more sectors. That is to say the UE has active uplink radio connections with the network through more than one sector of the same BS. During softer handover only one power control loop per connection is active [7].

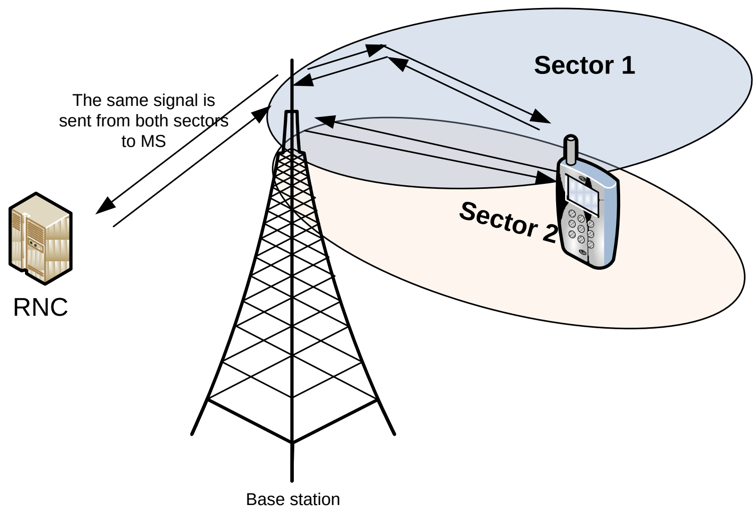

Figure 3.11: Softer handover

When soft and softer handovers occur simultaneously, the term soft-softer handover is used. A soft-softer handover may occur in association with inter-RNC handover, while an inter-sector signal is added to the UE's active set along with adding a new signal via another cell controlled by another RNC [7].

From system structural architecture standpoint, UMTS network supports the following types of handovers;

- Intra BS/inter-cell handover (soft handover)
- Inter BS handover including soft and hard handovers
- Inter RNC handover, including hard, soft and soft-softer handovers
- Inter MSC handover
- Inter SGSN handover
- Inter-RAT handover

3.2.8 Power control in WCDMA

Tight and fast power controls is one of the most important aspects in WCDMA particularly on the uplink. Without power control a single overpowered mobile could block a whole cell. In WCDMA, a transmit signal of one terminal appears as noise to another terminal. Therefore, if one terminal is transmitting at very high power, it will create an unacceptable Signal to Interference Ration (SIR) which will in turn prevent other mobiles from communicating with the base station. Power control therefore helps in maintaining a suitable SIR (The SIR of a single link is defined by a target Block Error Rate (BLER), say 3% for effective transmission). With an optimum SIR level that is set by the base station, a mobile terminal will have the correct network information before making a handover decision [7].

Figure 3.12 illustrates the problem and solution in the form of closed loop transmission power control [7].

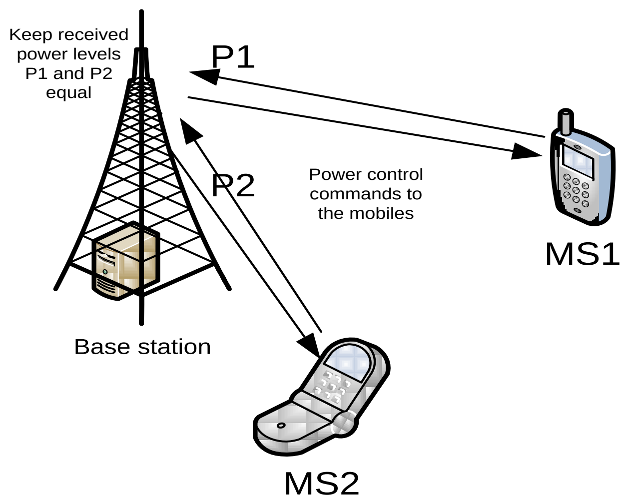

Figure 3.12: Closed loop power control in CDMA

Mobile stations MS1 and MS2 operate within the same frequency, separated at the base station by their respective spreading codes. MS1 at the cell edge suffers a path loss, e.g. 70dB above that of MS2 which is nearer to the BS. If there was no power control mechanism to bring MS1 and MS2 to the same power level, MS2 could overshoot MS1 blocking part of a cell. This phenomenon is known as the near-far problem of CDMA. The optimum strategy is thus to equalize the received power per bit of all mobile stations at all times [37]. If MS1 was mobile, it would therefore have difficulty in deciding which cell to handover as the information from BS has been blocked by MS2.

The solution to power control in WCDMA is fast closed loop power control, as shown in the figure above. On the uplink, the base station performs frequent estimates of the received Signal-to-Interference Ratio (SIR) and compares it to the target SIR. If the measured SIR is too low, it will command the MS to increase its power and if it's too high, it will command the MS to lower its power. Closed loop power control will therefore prevent any power imbalance among all uplink signals received at the base station. The same principal is applied on the downlink; however, the motivation here is different. It is desirable to provide a marginal amount of additional power to mobiles at the cell edge, as they suffer from interference from other cells [37].

3.2.8.1 **Inner loop power control**

Also called fast closed loop power control in the uplink, is the ability of the UE transmitter to adjust its output power in accordance with one or more Transmit Power Control (TPC) commands received in the downlink, in order to keep the received uplink Signal-to-Interference Ratio (SIR) at a given SIR target [37].

3.2.8.2 **Outer loop power control**

This is the ability of the UE transmitter to set its output power to a specific value. It is used for setting initial uplink and downlink transmission powers when a UE is accessing the network. In UMTS, the outer loop power control tolerance is ± 9 dB (normal conditions) or ± 12 dB (extreme conditions). The target SIR set point is within the BS according to the needs of an individual link and aims at keeping link quality consistent. The SIR is usually defined for a given bit error rate (BER) or block error rate (BLER). A network operator may define an SIR for say, BLER of 1% depending on the mobile speed and multipath profile [37].

3.3 Long Term Evolution (LTE) and the Evolved Packet System (EPS)

UMTS has evolved over the years, from pure 3G to what today is known as 3G+ or HSPA/HSPA+. At the time that LTE and HSPA evolution were started, 3GPP decided to ensure that an operator can co-exist between HSPA and LTE through an evolved core network, the Evolved Packet core/system (EPC/EPS). This was work done under the System Architecture Evolution (SAE). SAE focuses on how the 3GPP core network evolves into the core network of the next decades. The existing core was designed in the 1980s for GSM and has been extended over the years for GPRS and WCDMA. The EPC is focused on a packet-switched domain and total migration from the circuit-switched domain [18].

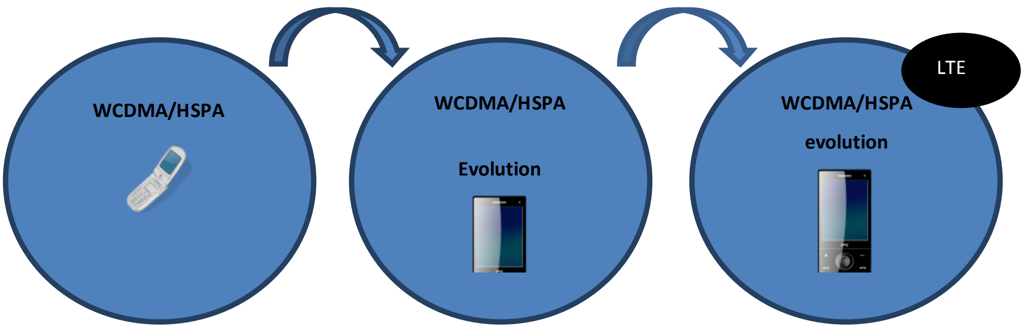

Figure 3.13: Upgrade to HSPA then deploy LTE as islands in WCDMA/HSPA Sea

Given that HSPA evolution is backward compatible and given that the EPC supports both HSPA and LTE is assurance that LTE can be deployed alongside HSPA and UMTS where it is needed as in the Figure 3.13 [18].

The techniques used for LTE radio access network architecture are Orthogonal Frequency Division Multiple Access (OFDMA) for the downlink direction and Single Carrier Frequency Division Multiple Access (SC-FDMA) for the uplink. Below are reasons why OFDMA is the preferred solution for the air interface [7] [18][25];

- Higher bandwidth and flexibility: OFDMA is a digital modulation scheme based on OFDM that allows multiple access on the same channel. OFDMA accommodates many users in the same channel at the same time. With LTE bandwidth up to 20MHz, OFDMA distributes subcarriers among users so all users can transmit and receive at the same time within a single channel on sub-channels. Subcarrier-group sub-channels can be matched to each user to provide the best performance, meaning the least problems with fading and interference based on the location and propagation characteristics of each use.

- Flat architecture: In this type of architecture more intelligence is added to the base station. With packet scheduling located in the BS, fast scheduling coupled with frequency scheduling can be used. Frequency domain scheduling can be done in OFDMA but not CDMA. Flat network architecture allows for easy scalability when data volumes increase. This enables a cost effective network rollout and capacity extension as traffic increases.
- Amplifier-friendly uplink solution with SC-FDMA: SC-FDMA technique has better power-amplifier efficiency. This therefore optimizes terminal power consumption.
- Simpler multi-antenna operation: With MIMO antenna operation, higher data rates can be achieved. MIMO is better implemented with OFDMA than CDMA.

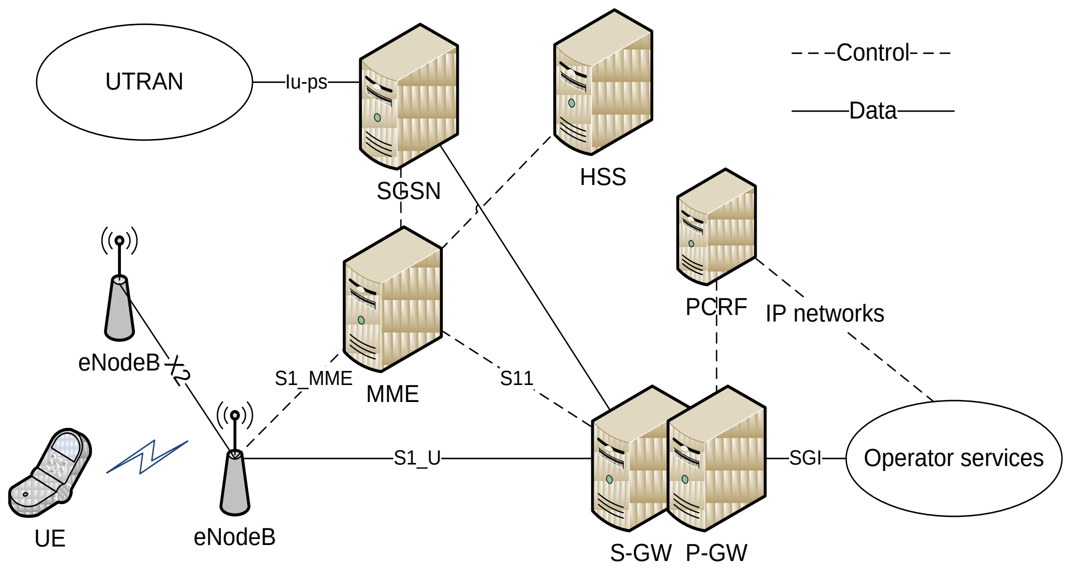

Figure 3.14: LTE/EPC Architecture

In LTE, full radio functionality lies in the base station. The base station also known as the eNodeB (eNB) now carries the following functionalities i.e. Radio Link Control (RLC) layer, Radio Resource Management (RRC) and Packet Data Convergence Protocol (PDCP). The architecture in Figure 3.15 shows the functions of the Radio Access and Core Network. The complete network architecture is shown in Figure 3.14. The LTE radio architecture is referred to as the E-UTRAN and is made up of the UE and the eNB [7].

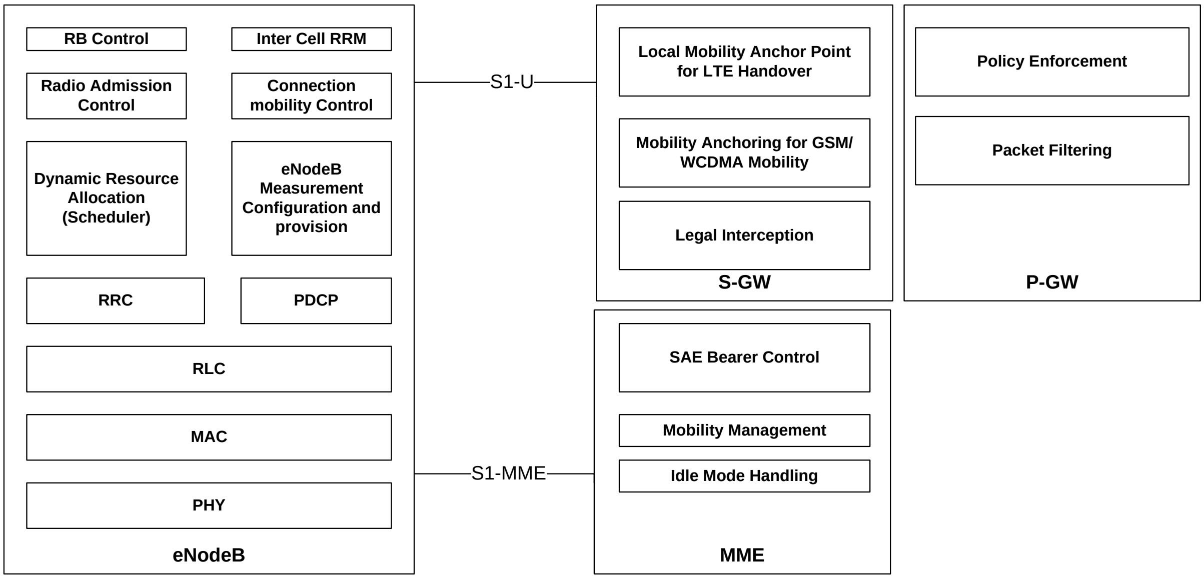

Figure 3.15: Functional split between the radio access and core network in LTE

3.3.1 LTE Network Architecture

In LTE the term EPC, Evolved Packet Core, is used to refer to the new core network. The following nodes are present in the EPC [7] [18] [19];

- Serving Gateway, SGW: gateway that terminates the interface towards E-UTRAN. For each UE associated with the EPS, at a given point of time, there is a single Serving GW. The S-GW performs several functions but mainly handles mobility management within the LTE network and other 3GPP technologies. SGSN could be connected to the S-GW to enable the smooth running with WCDMA/UMTS.
- Packet Data Network Gateway, PGW: The PDN GW is the gateway which terminates the **SGi** interface towards the PDN. The SGi interface is the reference point between the PDN gateway and the packet network. If a UE is accessing multiple PDNs, there may be more than one PDN GW for that UE; however a mix of S5/S8 connectivity and Gn/Gp connectivity is not supported for that UE simultaneously.
- Mobility Management Entity, MME deals with control plane signalling, mobility management and idle-mode. The MME is connected to the S-GW and P-GW via the S11 interface.
- Policy and Charging Rules Function, PCRF is concerned with QoS policy and charging policy.

3.3.2 The LTE Protocol Stack

The trend with LTE protocol design has been to put all radio-related functionality in the eNodeB. This follows the trend introduced in HSDPA and HSUPA where the MAC layer functionality was added to the Node B. More recently, the RLC functionality has been moved to the eNodeB as well as the PDCP functionality with ciphering and header compression [37][18][7].

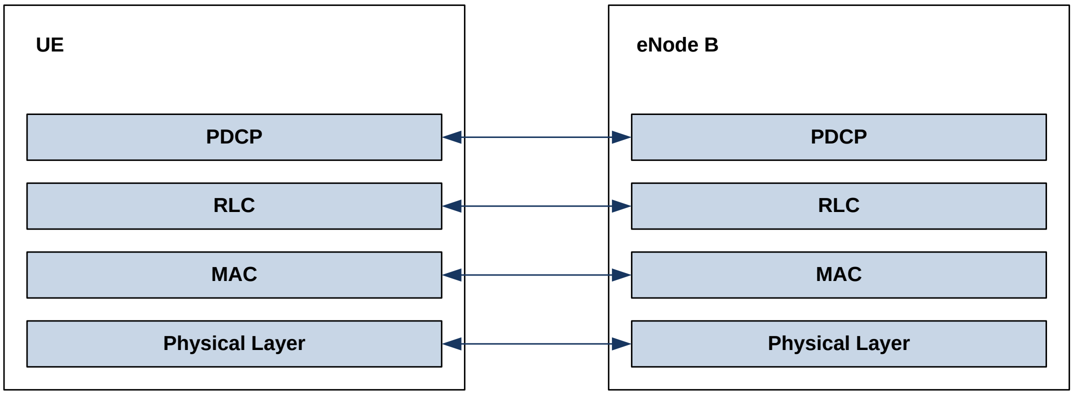

Figure 3.16: LTE user plane protocol stack

All protocols previously in UTRAN between Node B and RNC are now between the UE and eNodeB in LTE with the control plane as in Figure 3.16. In LTE the MAC and RLC layers handle similar functionalities as was with HSDPA and UMTS respectively. MAC layer deals with scheduling, priority handling, retransmissions and multiplexing of different logical channels on a single transport channel [37][18][7].

On the other hand RLC as was in WCDMA performs retransmissions in case of delivery failure, segmentation to fit the PDUs and providing logical channels to higher layers. PDCP performs ciphering and header compression [7] [18] [37].

In LTE, the RRC configures the connection parameters, controls the terminal measurement reporting, carries handover commands etc. The RRC protocol in LTE contains only two states i.e. idle and active. Figure 3.17 compares the WCDMA RRC protocol states to the LTE RRC protocol states [7] [18] [37].

RRC_IDLE state: in this LTE RRC state, the UE monitors paging messages and uses cell-reselection for mobility. In this state, no RRC context is stored in any particular eNodeB, the UE only has an ID which identifies the within the tracking area.

RRC_CONNECTED: in this state, the UE location is known at cell level and data is sent and received. An RRC connection to the eNodeB exists. Handovers controlled by the network are used for mobility. Handover to both WCDMA and GSM is supported to ensure service continuity.

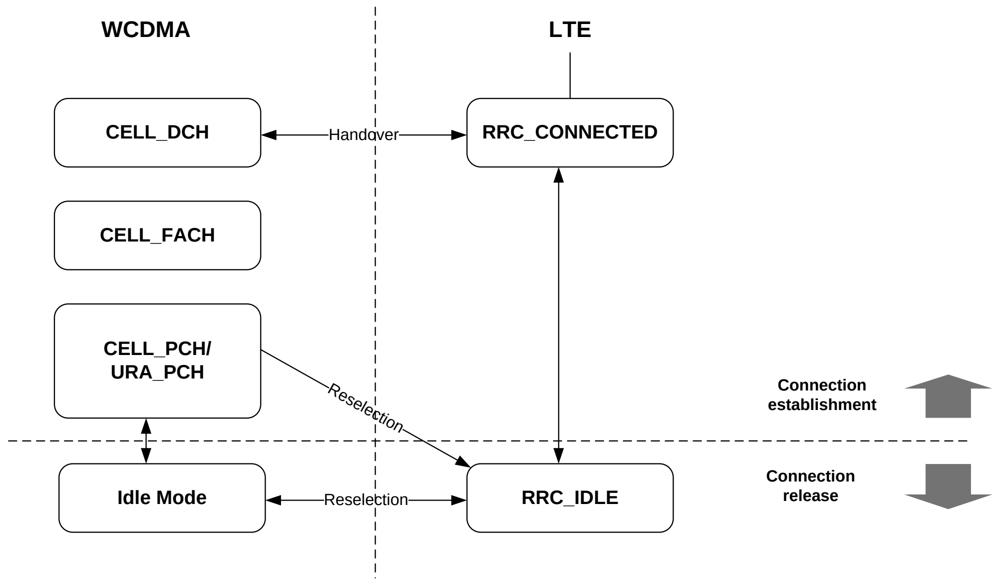

Figure 3.17: WCDMA and LTE RRC protocol states

The control plane functional split is shown in Figure 3.18. It is evident that all radio functionality lies within the eNodeB. X2 interface is used for inter eNodeB handover [7] [18] [37].

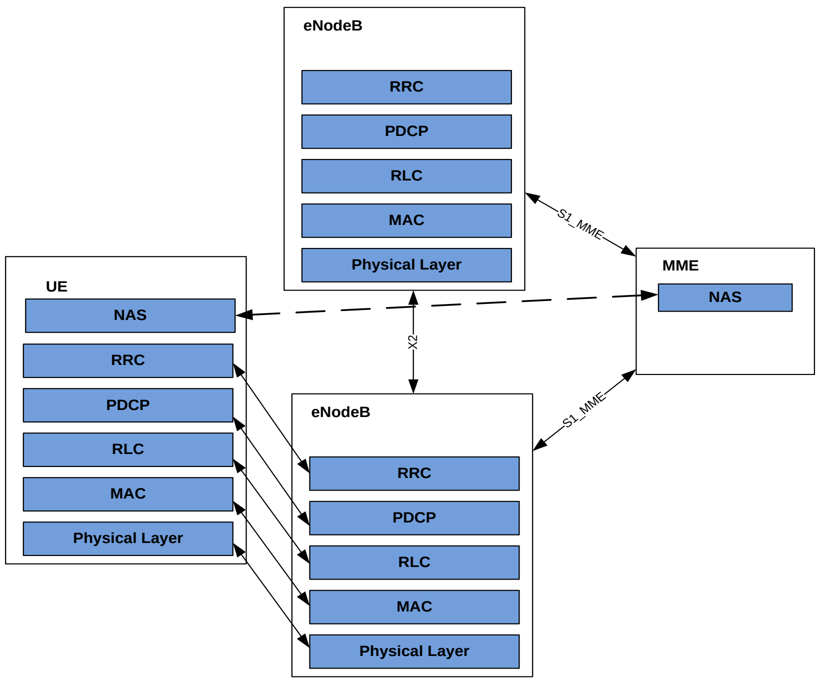

Figure 3.18: LTE Control-plane protocol stack

It is important to note that soft handover does not exist in LTE; hence there is no requirement to keep sending information over the X2 interface continuously. The interface may however be used to forward data related to the handover process. Non-Access Stratum (NAS) is signalling between the UE and the core network [7] [18] [37].

3.3.3 Comparison of LTE and WCDMA Physical layer

Although the LTE physical layer is different, it is built based on the experience of WCDMA/HSPA releases. The first LTE release covers aspects of HSDPA and HSUPA e.g. Hybrid Automatic Repeat reQuest (HARQ), base station-based scheduling and link adaptation, multi-antenna downlink transmission [20][38].

Feature	LTE	HSUPA	HSDPA
Multiple access	OFDMA SC-FDMA	WCDMA	WCDMA
Bandwidth range	1.4-20MHz	5-10MHz	5-10MHz
Fast power control	No	Yes	No (associated DCH only)
Soft handover	No	Yes	No (associated DCH only)
Adaptive modulation	Yes	Yes	Yes
BTS-based scheduling	Time/Frequency	Time/Code	Time/Code
Fast L1 HARQ	Yes	Yes	Yes

Table 3.2: LTE and WCDMA physical layer comparison

3.3.4 EPC Deployment with IMS

In this research, the handover methods chosen for evaluation (SRVCC and VCC) require a network architecture that interworks the EPC with IMS. It is therefore important to understand how the EPC will be connected to the IMS and UMTS networks in order to enable seamless mobility. This section therefore briefly deals with IMS and how it interworks with other systems.

The IP Multimedia Subsystem (IMS) is a generic platform offering IP-based multimedia services. IMS provides functions and common procedures for session control, bearer control, interworking, policy and charging [39][40][41].The IMS supports communication between heterogeneous networks such as PSTN, GSM, UMTS, WLAN and LTE. It was originally designed by the wireless standards body Third Generation Partnership Project (3GPP), as part of the vision for evolving mobile networks beyond GSM. Interworking between EPC, IMS and other networks takes on the general architecture shown in Figure 3.19 [4] [42].

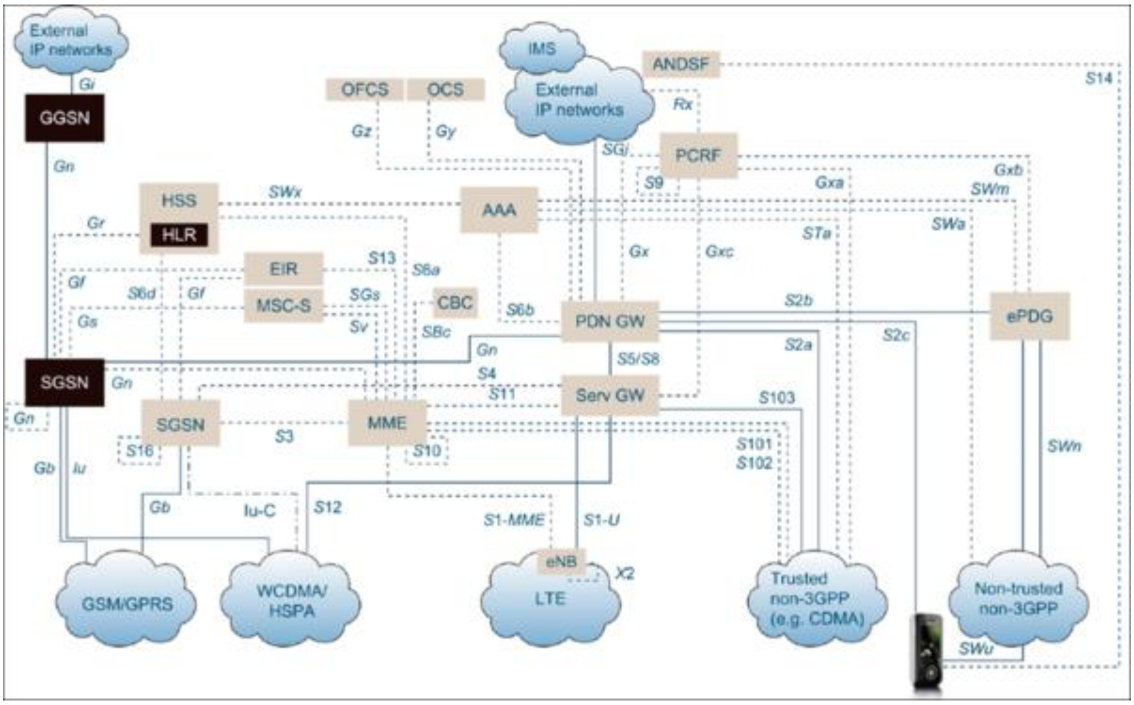

Figure 3.19: The EPC interworking IMS with other networks

The IMS network consists of several entities each with specific functionality. Important IMS entities are briefly described as follows[4];

- Proxy-CSCF: Proxy Call Session Control Function (P-CSCF) validates requests from the UE and forwards them to the selected destinations and processes. It behaves as a proxy and/or user agent.
- Interrogating CSCF: Contact between visited and home network. Is used to contact the HSS for subscriber information and to assign an S-CSF.
- Serving-CSCF: is the node for decisions on actions to be taken on receipt of session request and maintenance of the state of the session.
- Media Gateway (MGW): this terminates channels from a circuit switched network and media streams from packet switched networks.
- Media Gateway Control Function (MGCF): Performs limited call state control for connecting media streams and channels in the media gateway.
- Application Server: performs functions such as redirections, registrations, proxy etc.
- Policy charging and rules function (PCRF): is the node designated in real-time to determine policy rules in a multimedia network. As a policy tool, the PCRF plays a central role in next-generation networks. Unlike earlier policy engines that were added on to an existing network to enforce policy, the PCRF is a software component that operates at the network core and efficiently accesses subscriber databases and other specialized functions, such as a charging systems.

3.4 Quality of Service

Quality of service (QoS) refers to resource reservation control mechanisms rather than the perceived service quality. Quality of service is the ability to provide different priority to different applications, users, or data flows, or to guarantee a certain level of performance to a data flow. QoS parameters aim to share network resources more efficiently among users. For example, a required bit rate, delay, jitter, packet dropping probability and/or bit error rate may be guaranteed. During handover, the network must keep track of the available resources in order to maintain a specific quality of service to a terminal. The parameters for quality of service are usually stored within the network entities. For example, the BS is aware of the acceptable delay or bit rate and will always ensure that does not fall below or above the set level [7] .

In the EPS and UMTS, the QoS parameters and algorithms cannot be performed without the definitions of information flows to which they apply hence the following definitions are made [39];

EPS bearer: is the equivalent of PDP context in 2G/GPRS and 3G/UMTS. It is the concept of data flow between the user terminal and PDN GW. The bearer is composed of three elements, namely, the S5 bearer, the S1 bearer and the radio bearer. Each of these is implemented by the protocol of the interface after which it is named.

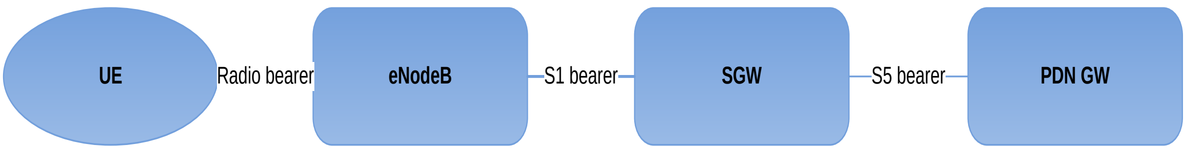

Figure 3.20: Relationship between bearer and interface

Data flows transported by the EPS bearer are known as *Service Data Flow* (SDF). Each SDF is characterized by: Source IP, Destination IP, Source Port Number, Destination port number and the protocol ID of the protocol above IP as well as the application of the service being used. In practice, an SDF may correspond to a connection to the web or video streaming server or mail server.

Quality of service of an EPS bearer is therefore characterised by the following parameters.

- Allocation Retention Priority (ARP) – This parameter is used for the allocation of bearer resources at session set up or during handover. After the bearer establishes a session, the ARP has no impact on scheduling or packet-handling.
- Guaranteed Bit Rate (GBR) –is used by bearers that require guaranteed QoS such as voice or video streaming.

- Maximum Bit Rate (MBR) – MBR parameters are used to cap the data rate for a given service. If bit rate exceeds a certain limit, the EPC network can apply traffic shaping functions.
- QoS Class Identifier (QCI) – is used as a reference to a set of Access Network-related QoS parameters between UE and eNodeB. QCI provides a representation of QoS parameters shared between core and access parts of the network. Each QoS class is associated with certain parameters. They include;
    - Bearer type: indicates whether resources associated with bearer should be allocated during whole bearer lifetime.
    - L2 Packet Delay Budget (L2PDB) – Maximum time packets shall spend when transiting RLC and MAC layers within network and the UE.
    - L2 Packet Loss Rate (L2PLR) –Maximum ratio of L2 packets which are not successfully delivered to peer.
- Aggregate Maximum Bit Rate (AMBR): This parameter associated with the terminal applied to non-GBR bearers. It is used to limit overall bit rate of all bearers associated with it for a given PDN.

Table 3.3 is an example of possible QoS class definitions for some GBR and non-GBR bearer types. Some are RT (Real Time) and others are NRT (Non Real Time). At session start up, the terminal indicates associated QoS attributes which are then checked by MME. An answer is sent to the UE with assigned attributes. For example attributes sent to E-UTRAN will be translated by the eNB into radio resource allocation, priority etc.

QCI	Bearer type	L2PDB	L2PLR	Example services
1	GBR	Low (<50ms)	Low (<10^{-6})	RT: Gaming
2	GBR	Medium(<100ms)	High (<10^{-3})	RT: Voice, live Video
3	GBR	High (<300ms)	Low (<10^{-6})	RT: Playback video
4	Non-GBR	Low (<50ms)	Low (<10^{-6})	NRT:SIP/SDP(IMS signalling)
5	Non-GBR	Medium(<100ms)	High (<10^{-3})	NRT: Web browsing RT: Interactive Gaming
6	Non-GBR	High (<300ms)	Low (<10^{-6})	NRT: bulk data transfer RT: Playback video

Table 3.3: Example of label characteristics

3.5 Comparison of QoS in UMTS and EPS systems

Table 3.4 and Table 3.5 show the QoS requirements for LTE and UMTS [7]. For example in LTE, a service such as voice, falls in the conversational class and should have a transfer delay of no more than 100 ms, it may have an allocation retention priority of 2. The Allocation/Retention Priority indicates the priority of allocation and retention of the service data flow. The Allocation/Retention Priority resolves conflicts in demand for network resources. 3GPP defines QoS parameters as a guide for equipment manufacturers/vendors. Equipment vendors in turn implement these parameters which ensure good customer experience.

Attribute	**Class**			
	Conversational	**Streaming**	**Interactive**	**Background**
Transfer delay (ms)	80	250		
GBR (kbps)	Up to 2Mbps	Up to 2Mbps		
Traffic handling priority			1,2,3	
Allocation/Retention priority (ARP)	1,2,3	1,2,3	1,2,3	1,2,3

Table 3.4: UMTS QoS classes and their parameters

Attribute	**Class**			
	Conversational voice	**Conversational video**	**Real time gaming**	**Streaming video**
Transfer delay (ms)	100	150	50	300
QCI	1	2	3	4
Allocation Retention priority (ARP)	2	4	3	5
Packet error loss rate (PELR)	10^{-2}	10^{-3}	10^{-3}	10^{-6}

Table 3.5: LTE QoS classes and their main parameters

3.6 Handover in LTE

Within the EPC, several types of handover can occur. Intra-E-UTRAN handover occurs within the LTE network. This type of handover can occur across the X2 or S1 interface. Inter-RAT handover takes place between E-UTRAN and other 3GPP networks. Both types of handover are defined in 3GPP Specification TS 123.401[19].

It is assumed that the UE is always moving from one place to another thus connecting to different eNBs. Referring to earlier LTE/EPC architecture, eNBs are connected to MME and SGW which in turn are connected to each other. When a UE moves from one area to another the following types of handover may occur [19]:

- Inter eNB handover or Intra MME handover : in this type of handover, a UE is moving from one eNB to other where both eNBs are connected to same MME.
- Inter eNB handover with MME change: Here a UE is moving from one eNB to another where each of the eNBs is connected to different MME's.
- Inter eNB handover with MME and SGW change: Here a UE is moving from one eNB to another where each eNB is connected to a different MME and SGW.

Figure 3.18 shows the eNBs connection to MME for control plane signalling. The MME has a direct link connected to the S-GW for user plane traffic. When there is inter eNB handover, there is a possibility that SGW handover has to be performed. It is also important to note the protocols used across the different network nodes. Two protocols are introduced; S1-AP and X2-AP. S1-AP is used for communication between two MME's and X2-AP is used for communication between two eNBs. GTPv2 is present in S11 and S5/S8 interfaces. An eNB may communicate with an MME using S1-AP protocol.

The three types of handover that occur in LTE can be summarized as [19]:

- X2 Based Handover: without SGW change
- X2 Based Handover: with SGW change (3GPP TS 23.401 Clause 5.5.1.1)
- S1 Based Handover: with eNB, MME and SGW change (3GPP TS 23.401 Clause 5.5.1.2).

3.6.1 X2 based Intra-E-UTRAN handover without S-GW change

In this procedure the MME is unchanged. In here, the UE is moving from source eNB to target eNB. Since the control plane (between UE and MME – no change in MME) is already established. Figure 3.21 shows this procedure [19].

1. The target eNodeB sends a Path Switch Request message to MME to inform that the UE has changed cell, including the identifying info of the target cell and the list of EPS bearers to be switched. The MME determines that the Serving GW can continue to serve the UE.

2. The MME then sends out a Modify bearer request (MBR) which includes all the bearers' information and new eNB information to SGW. All the information is sent in a single message.
3. If all information sent by MME is correct, the S-GW will accept the MBR message and return the response. The S-GW starts sending downlink packets to the target eNodeB using the newly received information. A Modify Bearer Response message is sent back to the MME.
4. In order to assist the reordering function in the target eNB, the S-GW sends one or more "end marker" packets on the old path immediately after switching the path.
5. The MME confirms the Path Switch Request message with the 'Path Switch Request Ack' message.
6. By sending Release Resource the target eNodeB informs success of the handover to source eNodeB and triggers the release of resources.

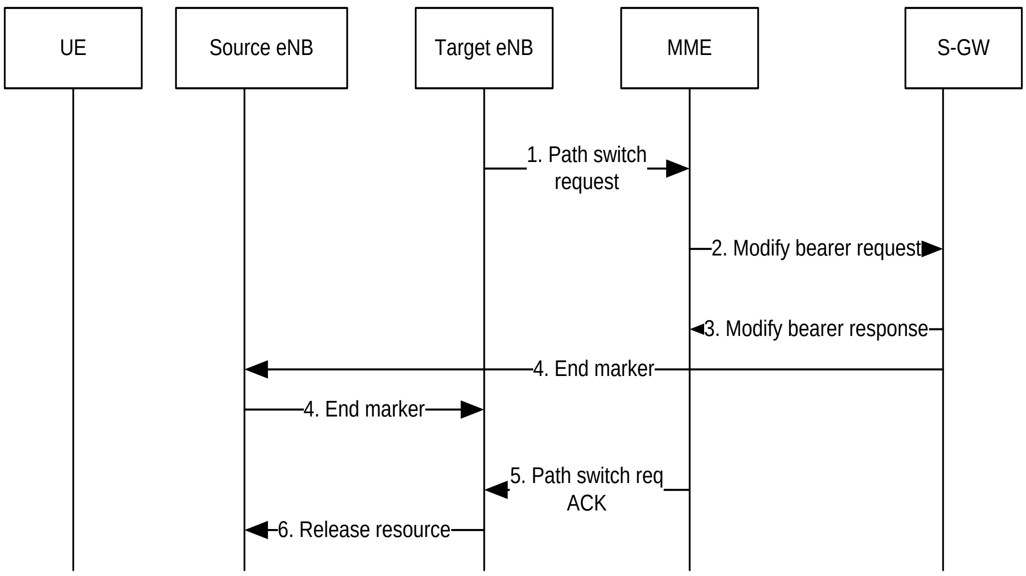

Figure 3.21: X2 based handover without S-GW relocation

3.6.2 X2 based Intra-E-UTRAN handover with S-GW change

For X2 based handover with S-GW relocation, similar procedure takes place except in this case, there is deletion of the session in the source S-GW and creation of a new session in the target S-GW. This is shown in Figure 3.22 [19].

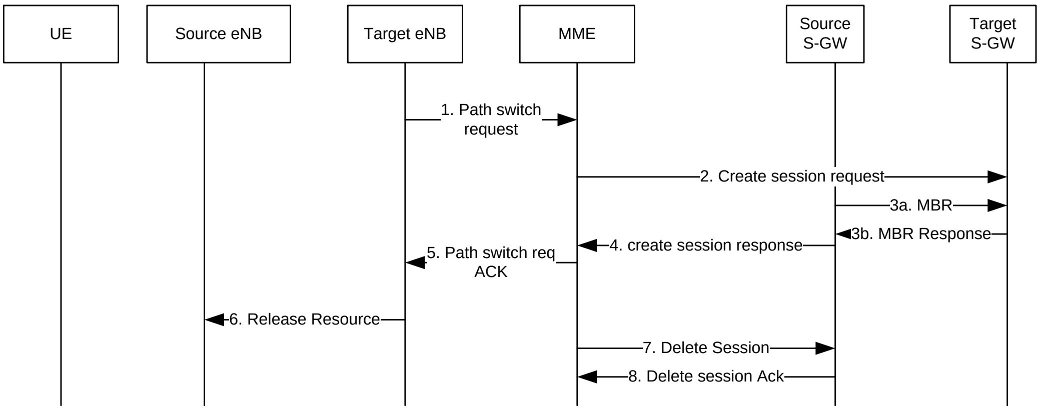

Figure 3.22: X2 based handover with change in S-GW

3.6.3 S1 based handover

This procedure takes place over the S1 interface, it is assumed there is no X2 link between source and target eNodeB. Note that there is change in both MME and S-GW. The procedure is summarized in Figure 3.23 and is explained as follows [19].

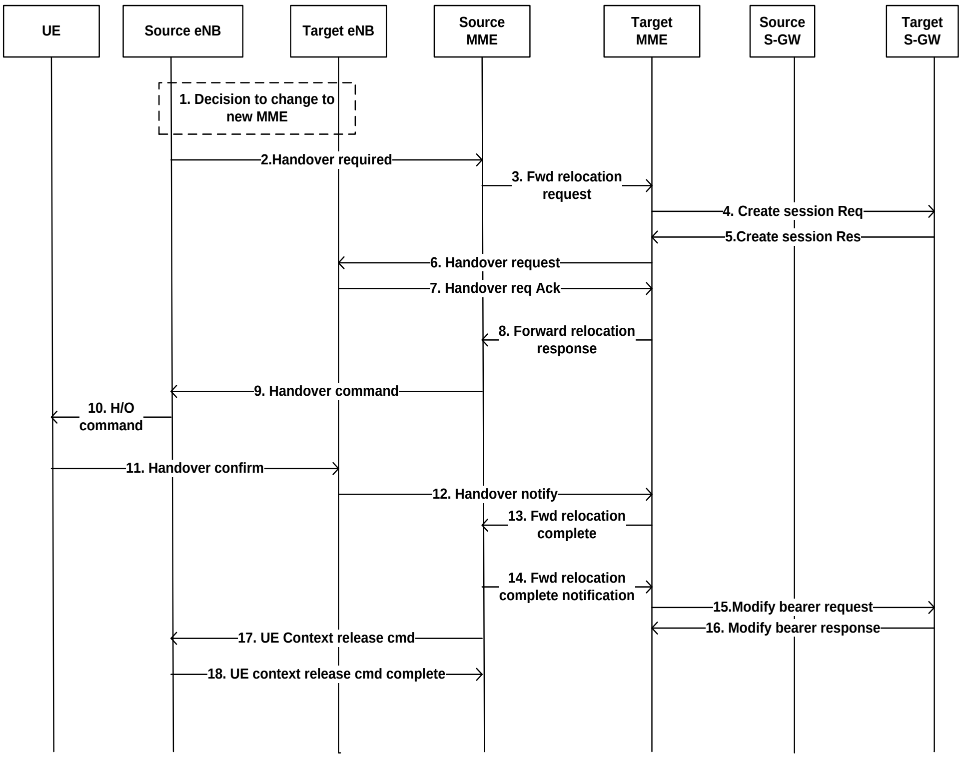

Figure 3.23: S1-based handover

1. The source eNodeB decides to initiate an S1-based handover to the target eNodeB. This can be triggered e.g. by no X2 connectivity to the target eNodeB, or by an error indication from the target eNodeB after an unsuccessful X2-based handover, or by dynamic information learnt by the source eNodeB.
2. The source eNodeB sends 'Handover Required' to the source MME.
3. The source MME selects the target MME and it sends a 'Forward Relocation Request'. The target MME determines whether S-GW relocation is needed (and, if needed, aids SGW selection). If the MME has been relocated, the target MME verifies whether the source SGW can continue to serve the UE. If not, it selects a new Serving GW.
4. If a new Serving GW is selected, the target MME sends a 'Create Session Request' message connection to the target Serving GW. The target Serving GW allocates the S-GW addresses
5. The target Serving GW sends a 'Create Session Response' message back to the target MME.
6. The Target MME sends 'Handover Request' message to the target eNodeB. This message creates the UE context in the target eNodeB.
7. The target eNodeB sends a 'Handover Request Acknowledge' message to the target MME.
8. The target MME sends a 'Forward Relocation Response' message to the source MME.
9. The source MME sends a 'Handover Command' message to the source eNodeB.
10. The 'Handover Command' is sent to the UE.
11. After the UE has successfully synchronized to the target cell, it sends a 'Handover Confirm' message to the target eNodeB. Downlink packets forwarded from the source eNodeB can be sent to the UE. Also, uplink packets can be sent from the UE, which are forwarded to the target Serving GW and on to the PDN GW.
12. The target eNodeB sends a 'Handover Notify' message to the target MME.
13. If the MME has been relocated, the target MME sends a 'Forward Relocation Complete Notification' message to the source MME.
14. The source MME in response sends a 'Forward Relocation Complete Acknowledge' message to the target MME. Regardless if MME has been relocated or not, a timer in the source MME is started to supervise when resources in Source eNodeB and if the Serving GW is relocated, also resources in the Source Serving GW are released. Upon receipt of the Forward Relocation Complete Acknowledge message the target MME starts a timer if the target MME allocated S-GW resources for indirect forwarding.
15. The MME sends a 'Modify Bearer Request' message to the target Serving GW
16. The target Serving GW sends a 'Modify Bearer Response' message to the target MME.

17. When the timer started in step 14 expires, the source MME sends a UE 'Context Release Command' message to the source eNodeB.
18. The source eNodeB releases its resources related to the UE and responds with a UE 'Context Release Complete' message.

3.6.4 Inter-RAT handover in the EPC

With the introduction of the EPC, backward compatibility to existing 3GPP technology is possible. If a UE reaches end of coverage area for LTE services, it can handover to a 2G service such as GSM or UMTS and vice versa if the UE supports multiple RAT. [19] defines Inter-RAT handover procedure between E-UTRAN and its predecessors such as UTRAN and GERAN. The following sections describe inter-RAT handover from E-UTRAN to UTRAN and vice versa within the packet switched domain. It is important to understand inter-RAT handover within the PS domain before moving onto different domains.

For handover to occur the UTRAN and E-UTRAN must be inter-connected as in Figure 3.24. The PDN Gateway replaces the GGSN in the UTRAN while the SGSN remains as before but is connected to the MME and S-GW. This means minimal changes are made to UTRAN while being able to provide fast and seamless handover to and from LTE. UTRAN to EUTRAN handover is handled by MME and SGW through the S3 interface [19].

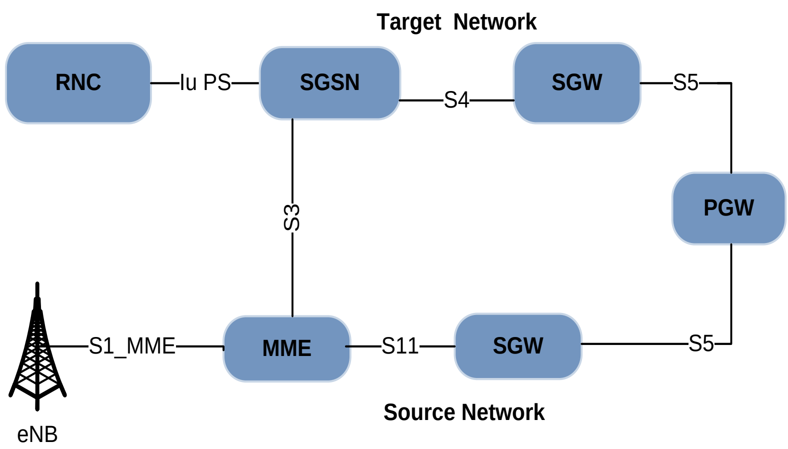

Figure 3.24: EUTRAN to UTRAN handover at physical layer

3.6.4.1 **E-UTRAN to UTRAN Inter RAT handover**

The precondition for this kind of handover to occur is that the UE is in the ECM_CONNECTED state [19]. This is the state when the signalling connection is established between a UE and MME. Figure 3.24 shows the Source network being LTE and target network as UMTS packet switched domain. The target SGSN communicates with the MME over the S3 interface. We also assume that the LTE SGW is serving the 3G network. This means that the SGSN uses the S4 interface to communicate with the SGW. Note that the GGSN is no longer required as it is replaced by the PDN-GW. Figure 3.25 shows the call flow summary. Some steps have been removed to simplify the diagram, however, a more detailed call flow can be found in [19] section 5.5.2.1. This type of handover is similar to S1 based handover in LTE except for the additional communication between RNC and SGSN.

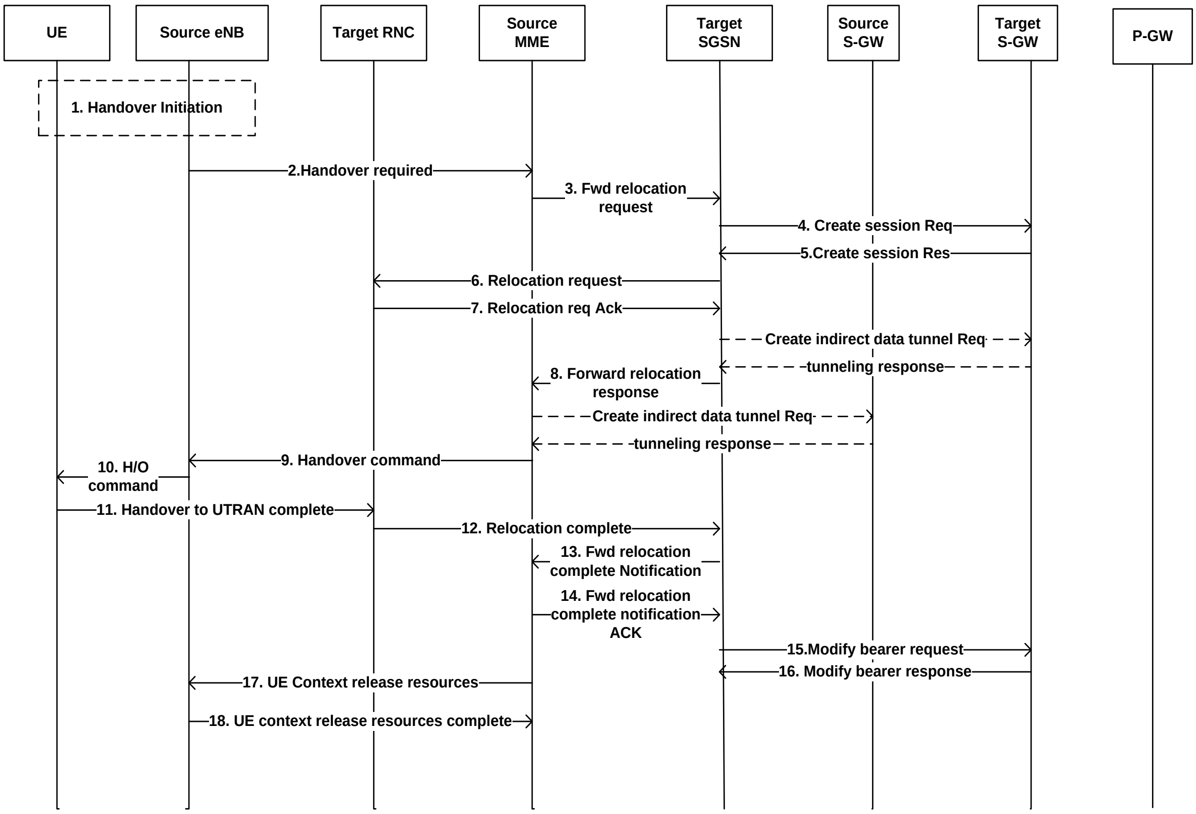

Figure 3.25: E-UTRAN to UTRAN handover call flow summary

3.7 Requirements for Interworking LTE and UMTS

3GPP specifies requirements for interworking between LTE and other 3GPP networks. In this research, we were particularly concerned with requirements for LTE to UMTS/WCDMA handover. The following requirements are applicable to inter-working between LTE and UMTS [43]:

- E-UTRAN Terminals supporting UTRAN operation should be able to support measurement of handover to and from 3GPP UTRA system with acceptable network impact on terminal complexity and network performance.
- The E-UTRAN is required to efficiently support inter-RAT measurements with acceptable impact on terminal complexity and network performance, e.g. by providing UE's with measurement opportunities through downlink and uplink scheduling.
- The interruption time during a handover of real-time services between E-UTRAN and UTRAN is less than **300ms** (maximum).
- The interruption time during a handover of non-real-time services between E-UTRAN and UTRAN should be less than 500ms (maximum).
- The interruption time during a handover between an E-UTRA broadcast stream and a UTRAN unicast stream providing the same service (e.g. same TV channel) has not yet been specified (For Further Study - FFS[3]).
- The interruption time during a handover between an E-UTRA broadcast stream and a UTRAN broadcast stream providing the same service (e.g. same TV channel) is less than FFS.

3GPP stipulates the above requirements for cases where the UTRAN networks provide support for E-UTRAN handover. The interruption times above are to be considered as maximum values.

3.8 Conclusion

This chapter introduced and explained the UMTS, LTE/EPC network architecture and the parameters that affect performance of the air interface in the two systems. It also explained the LTE and UMTS protocol stacks and how the QoS in LTE compares with that in UMTS. The chapter also discussed interworking between the EPC and UMTS which allows for Inter-RAT handover. Intra-LTE and Inter-RAT handover between LTE and UMTS-PS were discussed as well as the 3GPP requirements for Inter-RAT handover between LTE and UMTS. This chapter therefore gives insight into LTE and UMTS technologies as a whole and their relationship. Before delving into the details of UMTS and LTE inter-RAT handover, it was important to understand the handover procedures and the factors that lead to handover within the two networks.

[3] For Further Study (FFS): This is a term used by 3GPP in technical specifications. For example if studies are still being carried out between LTE and UMTS voice handover, the specified interruption time is termed as FFS.

4 Inter-RAT Voice Handover Techniques

4.1 Introduction

LTE is entirely a packet switched system and voice will be provided over IP; that is, Voice over IP (VoIP). At the beginning of LTE deployment, it may take a while before VoIP can be provided due to the size of the venture. One of the key challenges of LTE deployment is the continuity of voice services. Voice remains the “killer application” for operators because it still accounts for a large portion of their revenue. Voice will continue to remain the dominant service in the cellular network for years, and despite the technical challenges of providing the service over an all-Internet Protocol (IP) radio access network (RAN), voice is considered to be a basic service by the consumer; that is to say, it is expected. However, voice service continuity is not guaranteed when a VoIP subscriber roams between the LTE coverage area and other 3GPP networks – and it is a significant challenge to deliver voice over LTE networks [44].

Today, the telecommunications industry is exploring and evaluating different possibilities to overcome the LTE voice handover issues with each stakeholder backing a different solution. Solutions such as Circuit Switched Fall Back (CSFB), Voice over LTE Generic Access (VoLGA) and LTE VoIP-based Single Radio Voice Call Continuity (SRVCC) have been explored. These three options are discussed in the ensuing sections and compared in Table 4.1. At the time of writing this thesis, only SRVCC and CSFB had been standardised by the Third Generation Partnership Project.

4.1.1 Voice over LTE via Generic Access (VoLGA)

The VoLGA concept has been identified as one of the solutions to solve the problem of voice handover between LTE and 2G/3G networks. In VoLGA, the idea is to “raise” the existing voice network to act as a packet service delivered over the LTE access network hence the alternate name “Circuit Switched over Packet Switched” (CSoPS) [44]. With this technique, voice services can be recreated by making the existing telephony infrastructure a packet service delivered over IP via LTE [44]. The existing 3GPP Generic Access Network (GAN) standard has quickly emerged as the preferred move towards realizing the VoLGA concept. The VoLGA architecture and operation are described in the following sections.

4.1.1.1 **VoLGA Architecture**

The 3GPP GAN standard commonly referred to as the Unlicensed Mobile Access (UMA) is the technology behind VoLGA architecture. The concept of GAN is to extend existing mobile services over any generic broadband access network. GAN has been used in the extension of services by allowing seamless handover between Wireless LANs (e.g. Wi-Fi) and Wide Area Networks (e.g. GPRS or UMTS). The same concept is used by VoLGA [45].

In VoLGA the architecture of the UTRAN and E-UTRAN is not changed except a new network element known as the VoLGA Access Network Controller (VANC) is introduced as shown in Figure 4.1 [45].

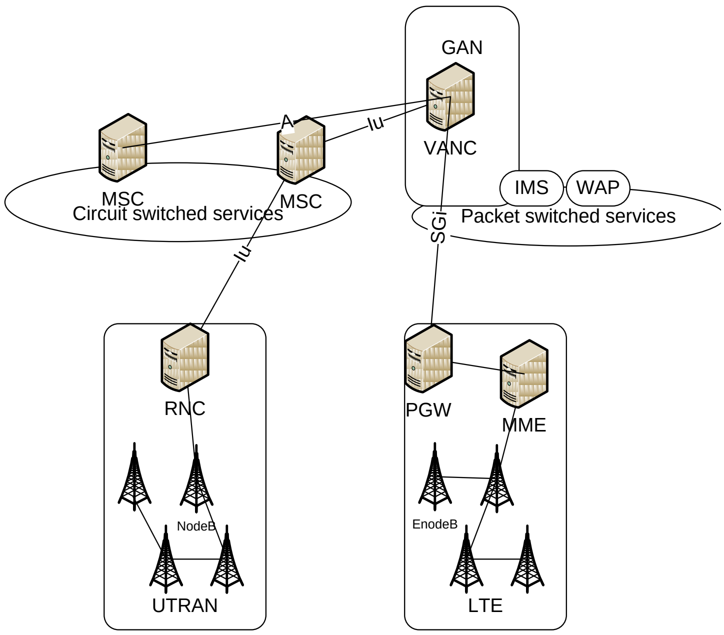

Figure 4.1: Voice over LTE via GAN

On the LTE/E-UTRAN side, the VANC connects to the Packet Data Network Gateway (P-GW) via the standard SGi interface. The SGi interface is a reference point defined by 3GPP. It is between the PDN Gateway and any packet data network [38]. The packet data network may be an operator external public or private packet data network or an intra-operator packet data network. In VoLGA, both control and user traffic is transported over this interface. From an LTE core network point of view the VANC looks like any other IP based external node and IP packets exchanged between a wireless device and the VANC are transparently forwarded through the Evolved Packet Core (EPC) network. On the circuit switched network side the Iu interface is used to connect the VANC to the UMTS MSC. The VANC thus appears like a UMTS RNC to a UMTS MSC. In practice the interface used depends on the requirements of the network operator [46].

4.1.1.2 **VoLGA Operation**

When a UE is switched on and detects an LTE network, it registers with the Mobility Management Entity (MME) over the LTE access network. After registering with the LTE network, the mobile establishes a connection to the VANC using the VoLGA configuration stored in the UE. The mobile device then registers with the MSC through a secure tunnel and the VANC. The VANC only adds information such as the service area identifier (UMTS) to the initial registration message as defined in UMTS standards. VoLGA generally operates as follows [45].

When the mobile registers with the LTE network it signals its capability to the MME. The network is thus aware that the VoLGA procedure needs to be executed when the mobile device is about to leave the LTE coverage area while a bearer is active.

When the eNodeB detects that the mobile device could be better served by a UMTS cell it can instruct the UE to measure the signal strength of such neighbouring cells. Based on these measurement results or based on pre-configured values, the eNodeB informs the MME that a handover to a UMTS cell is required.

The MME in turn informs the VANC about the imminent handover. The message includes information such as the target cell-id and the id of the subscriber for which the handover is to be made. For this purpose, the MME uses the Sv interface (between MME and MSC).

Figure 4.2 shows the signalling procedure to establish a mobile originated voice call over LTE using VoLGA. All signalling and control plane messages between the UE and the VANC are transported over an established IPsec tunnel.

In step 1, the mobile device sends a message to the VANC to change the connection from idle to dedicated state. In Step 2, the UE sends a standard UMTS Service Request message to establish a connection to the MSC. When the VANC receives the message it creates a dedicated signalling connection to the MSC over the Iu interface for the user and forwards the message to the MSC. The MSC then authenticates the user and activates ciphering (step 4 and 5).

The mobile device sends a Setup message in step 6, which contains among other things, the phone B party number. The MSC acknowledges the request with a Call Proceeding message in step 7. Since the MSC considers the VANC as a UMTS RNC, it sends an Assignment Request message to the VANC to request the establishment of a circuit switched bearer channel. The VANC translates this message into an Activate Channel message to the mobile device in step 9 to prepare it for the exchange of IP packets containing voice data.

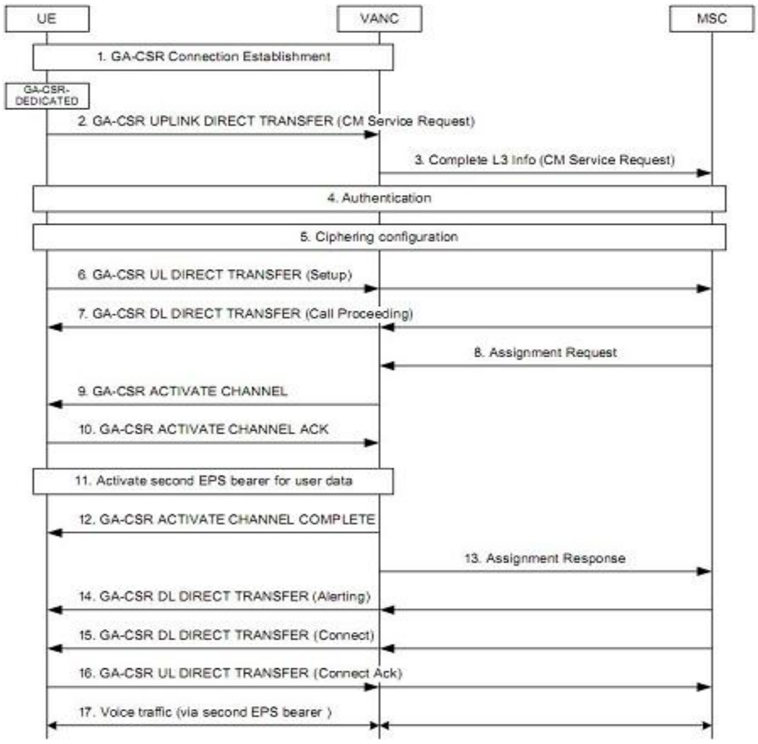

Figure 4.2: Mobile originated call flow in VoLGA

As an alternative, quality of service for the voice packets can be ensured by activating a second bearer in the LTE network (step 11). Once the mobile device is prepared for the voice data stream, an Assignment Response message is sent back to the MSC in step 13 to signal to it the successful 'pseudo' establishment of a circuit switched channel in the radio network. Once the call has been established with the other party, the MSC sends Alerting and Connect Messages (step 14 and 15) which the mobile device acknowledges. The voice path is then established and the voice conversation can begin.

The voice signal is transmitted via an ATM or IP based data flow. The VANC translates this data stream into IP packets for transmission over the LTE network and vice versa. The Real-time Transfer Protocol (RTP) is used for this purpose.

4.1.2 Circuit Switched Fall Back (CSFB)

Another proposed scheme for LTE to UMTS handover is Circuit Switched Fall Back (CSFB). As hinted by its name, a CSFB capable terminal being served by E-UTRAN falls back onto the circuit switched domain whenever it makes or receives a voice call. The CSFB procedure in the EPS therefore enables the provisioning of voice services by making use of the CS infrastructure when the UE is served by E-UTRAN. A CSFB enabled terminal, connected to E-UTRAN may use the UTRAN or GERAN to connect to the CS-domain. CSFB is only available where E-UTRAN coverage is overlapped by either GERAN coverage or UTRAN coverage. 3GPP has standardised CSFB in release 9[47]. Currently, CSFB is strongly supported by operators such as NTT DoCoMo of Japan. In January 2011, Huawei Technologies, a telecommunications equipment vendor successfully tested CSFB with China Unicom and submitted a report to 3GPP to aid with future standardization [48].

The aptly named CSFB allows users to switch voice calls from LTE to UTRAN and GERAN. In this report we limit our scope to UTRAN. The basic architectural and operational concept of CSFB is discussed in the sections that follow.

4.1.2.1 **CSFB Architecture**

The CSFB architecture is shown in Figure 4.3. The CSFB architecture makes minor alterations to the EPC and the UMTS core. The new changes made are [47]:

A reference point between the MSC Server and the MME, called the SGs interface. The SGs reference point is used for the mobility management and paging procedures between EPS and CS domain. The SGs reference point is also used for the delivery of both mobile originating and mobile terminating SMS [47].

The S3 reference point (between SGSN and MME) which is already defined in [19] is enhanced with the additional functionality to support Idle-mode Signalling Reduction (ISR)[4] [49]for CSFB over SGs [47].

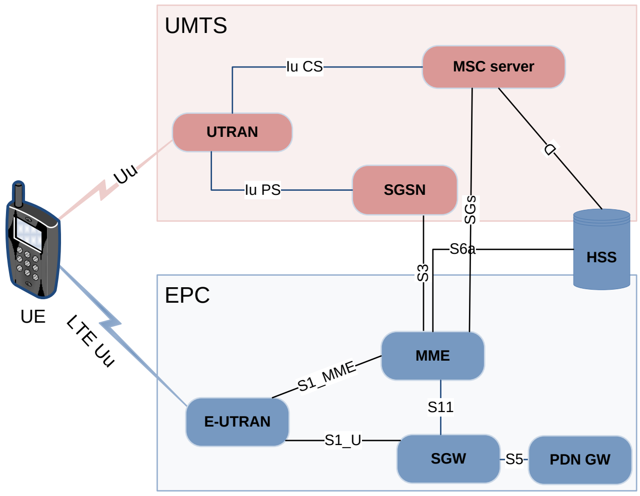

Figure 4.3: Evolved Packet System Architecture for CSFB

[4]ISR is a mechanism that allows the UE to remain simultaneously registered in an UTRAN/GERAN Routing Area (RA) and an E-UTRAN Tracking Area (TA) list. This allows the UE to make cell reselections between E-UTRAN and UTRAN/GERAN without a need to send a new request as long as it remains within the registered RA or TA list.

4.1.2.2 **CSFB Operation**

CSFB takes place whenever a mobile terminal receives or makes a voice call. Simplified CSFB procedure for a mobile terminated call (MTC) is shown in Figure 4.4. Consider a mobile terminal camping on the E-UTRAN; in step 1 a mobile terminated voice call arrives at the terminal via the SGs interface from the CS network. The UE recognises that the call is from the CS domain given the information contained (e.g. VLR/MSC address). In step 2, the EPC communicates with UMTS, UMTS prepares network resources for the new call. The EPC instructs the mobile to switch to UTRAN. In Step 3, the mobile moves to UTRAN and in step 4 the voice call proceeds as before. Throughout this process, the mobile user is unaware of the handover. This procedure is described in detail in [47].

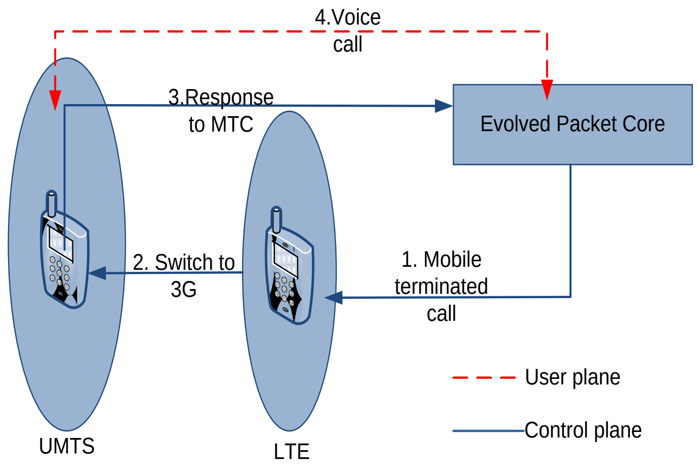

Figure 4.4: CSFB concept

Figure 4.5 is the signal flow for a Mobile Originated Call (MOC) in CSFB. A subscriber who is concurrently attached to the LTE and UMTS networks initiates a voice call, the CSFB procedure will take place as follows [47];

Procedure 1~2: The UE sends an Extended Service Request (with CSFB Indicator) to the MME via the eNodeB. The CSFB Indicator indicates to the MME to perform CSFB procedure. The UE only transmits this request if it is attached to the CS domain (with a combined EPS/IMSI Attach).

Procedure 3~4: The MME then sends a UE Context Modification Request message to the eNodeB. This message indicates to the eNodeB that the UE should be moved to UTRAN. The eNodeB replies with UE Context Modification Response message.

Procedure 5: The eNodeB may optionally solicit a measurement report from the UE to determine the target UTRAN cell to which handover will be performed.

Procedure 6~9: The eNodeB triggers handover to a UTRAN neighbour cell by sending a Context Release message to the MME. The eNodeB triggers a RRC connection release with redirection to UTRAN. The MME releases the context and the UE tunes to UTRAN. The UE then establishes the radio signalling connection with the UTRAN network.

Procedure 10~20: The UE will proceed to initiate the CS call establishment procedure with the MSC over UTRAN. After the CS voice call is terminated the UE resumes PS services over E-UTRAN.

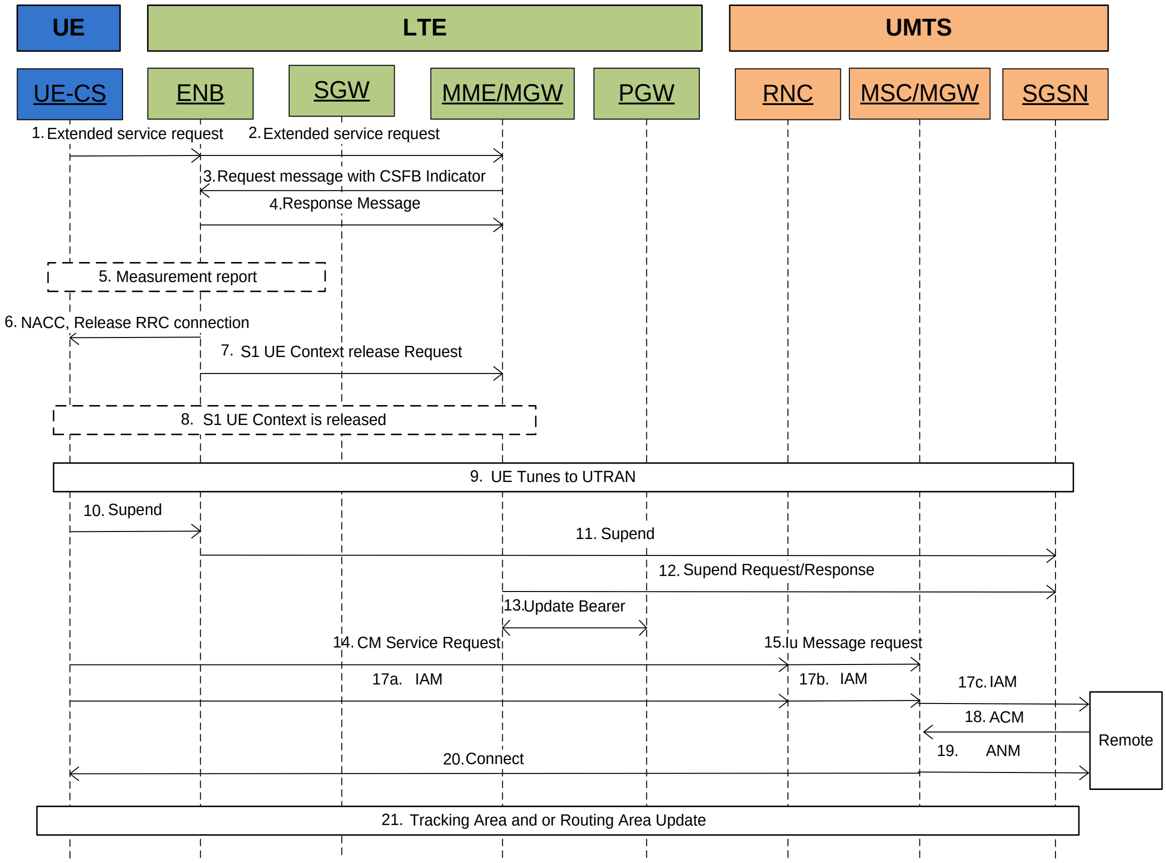

Figure 4.5: CSFB MOC procedure [47]

CSFB is a good interim solution as it does not mandate an IMS core and only requires a few modifications to be made on the MME and the MSC server. Operators can therefore move straight to providing circuit switched voice services after LTE deployment. CSFB, however, is not sustainable in the long term because it demands that user terminals must be equipped with either dual-mode/ single-standby[5] or dual-mode / dual-standby capabilities. Although dual–mode / dual-standby mobile phones require less network changes to facilitate inter-working between two networks, they

[5]Single radio mode terminal refers to the ability of a terminal to transmit or receive on only one of the given radio access networks at a given time.

drain the battery power quickly and need complex terminal customization. Experimental work performed so far for CSFB indicates a high latency [50]. This is mostly because the terminal requires extra time to start up the 3G radio, measure radio channels and setup a new call. SRVCC eliminates this problem [50].

4.1.3 Single Radio Voice Call Continuity (SRVCC)

The long-term solution for circuit switched voice handover between UMTS and LTE is SRVCC and is the main inter-RAT technique chosen for evaluation in this research. 3GPP has standardized SRVCC in Release 8 [51]. At present, 3GPP has only standardized SRVCC handover from LTE to UMTS; UMTS to LTE SRVCC procedure is not yet available. In this research, we used CS to PS domain transfer (VCC) in IMS for UMTS to LTE handover.

In general, SRVCC works by converging mobile and broadband wireless access technologies as it offers LTE-IMS based voice service within the LTE coverage area, and CS-based voice service outside the LTE coverage area. In addition, SRVCC architecture provides existing networks with a robust IMS core and both fixed and wireless component assets in order to facilitate a converged VoIP solution.

Whenever the VoIP subscriber moves out of LTE coverage, SRVCC ensures a smooth handover of voice from the LTE to the CS network. The IMS network that stores voice service link information during this time guides the target CS network to establish a link, thereby replacing the original VoIP channel with a CS channel.

4.1.3.1 **SRVCC Architecture and Operation**

SRVCC takes place when a single radio User Equipment (UE) accessing IMS-anchored voice call services switches from the LTE network to the Circuit Switched domain – while it is able to transmit or receive on only one of these access networks at a given time. This eliminates the need for a UE to have simultaneous multiple Radio Access Technology (RAT) capability.

The UE accessing the SRVCC service is assumed to have IMS Service continuity capabilities with single radio access only. For single-radio terminals, measurement gaps are needed to allow the UE to switch onto the CS network and complete radio measurements. Measurement gaps define the time periods when no uplink or downlink transmissions are scheduled so that the UE may perform the measurements. The eNodeB is responsible for configuring the measurement gap pattern and provides it to the UE using Radio Resource Control (RRC) dedicated signalling. The UE assists the eNodeB by informing the network about its gap-related capabilities; informing it that if it has a dual or single receiver. This capability is transferred along with the other UE capabilities.

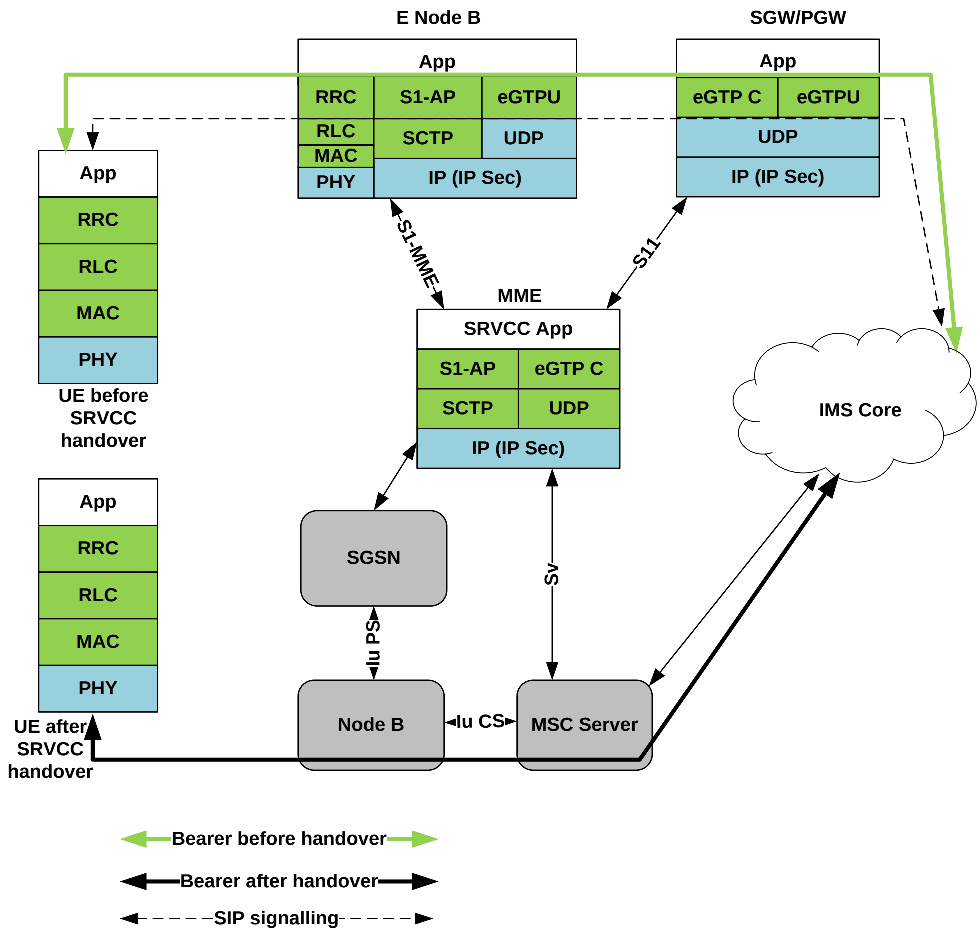

Figure 4.6: SRVCC reference architecture

Figure 4.6 illustrates the physical and protocol architecture of SRVCC between LTE and UMTS. The SRVCC architecture and basic operation can be simply described by the figure above. When the UE detects movement from E-UTRAN to UTRAN, it informs the LTE network via the scheduled measurement gaps.

The MME in conjunction with the IMS network perform handover preparation (see dashed arrow in the figure above). The old call leg is released (green arrow) and a new call is set up between UMTS and IMS network (see bold black arrow). When call setup is complete, the LTE network informs the UE which then tunes to the UMTS network. The call proceeds without discernable interruption. A detailed explanation on the SRVCC handover call flow follows in section 4.1.3.3.

4.1.3.2 Network enhancements to support SRVCC

For SRVCC procedure to take place, the MSC Server is enhanced with the following features:

- It is deployed in conjunction with the MME in the LTE network via the Sv interface. The Sv interface is used to transfer messages between the MME and MSC Server.

- The MSC comprises the call control (CC) and mobility control parts. It is responsible for the control of mobile-originated and mobile-terminated CS domain calls for channels in the CS-MGW (Media Gateway). It terminates the user-network signalling and translates it into the relevant network to network signalling, e.g. from SIP Signalling in IMS toSS7 signalling in the CS domain.

The MSC Server thus provides the following new improved features;

- Handles the Relocation Preparation procedure requested for the voice component from the MME via the Sv interface.
- Invokes the session transfer procedure from IMS to CS. This involves the access transfer at the IMS level of one or more of session signalling paths and associated media flow paths of an on-going IMS session.
- Coordinates the CS Handover and session transfer procedures.

Another network enhancement required for SRVCC procedure to take place is the Session Centralization and Continuity Application Server (SCC AS) in the IMS network. The SCC AS provides the following functionality to support SRVCC:

- Enables IMS Centralized Services. It contains the User Agent (UA) function that furnishes SIP UA behaviour on behalf of the UE for setup and control of IMS sessions using CS bearers that are established between the UE and the SCC AS.
- Executes and controls the session transfers needed by the UE for its call access legs anchored in IMS.

4.1.3.3 **Information flows for LTE to UMTS SRVCC**

A mobile terminal will set up a VoIP call in the PS domain as in Figure 4.7. Prior to handover, the call must be setup on the LTE network must be anchored in the IMS domain as in Figure 4.7 below.

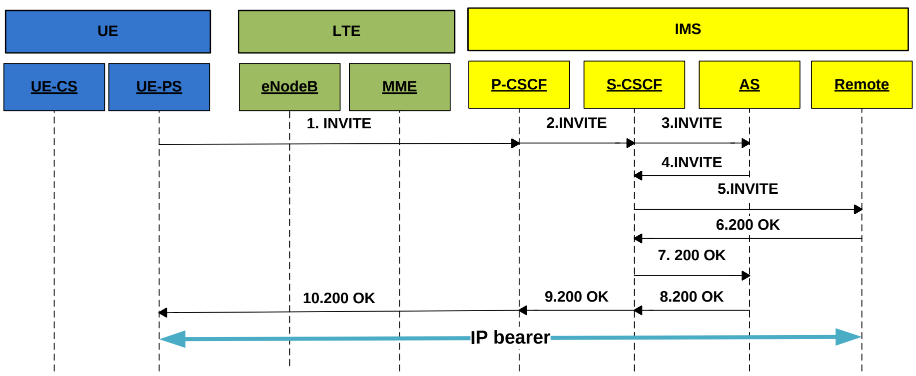

Figure 4.7: LTE originating call signal flow

Procedures for setting up a VoIP call in the PS domain can be explained as follows:

Procedure 1~5: UE sends SIP INVITE message to initiate a VoIP session and the INVITE message is transferred to IMS P-CSCF. Then the INVITE message is further transferred to S-CSCF and application server. Then the application server sends the INVITE message to a remote party. Every originating call for UE is anchored at the IMS application server which manages the call state and provides session immigration capability when handover occurs.

Procedure 6~10: A "200 OK" [6] response is transferred to UE through the network entities in reverse order of procedure 1~5.

A UE moving out of LTE coverage and into UMTS will handover to the UMTS-CS domain through the SRVCC procedure as shown in Figure 4.8 below.

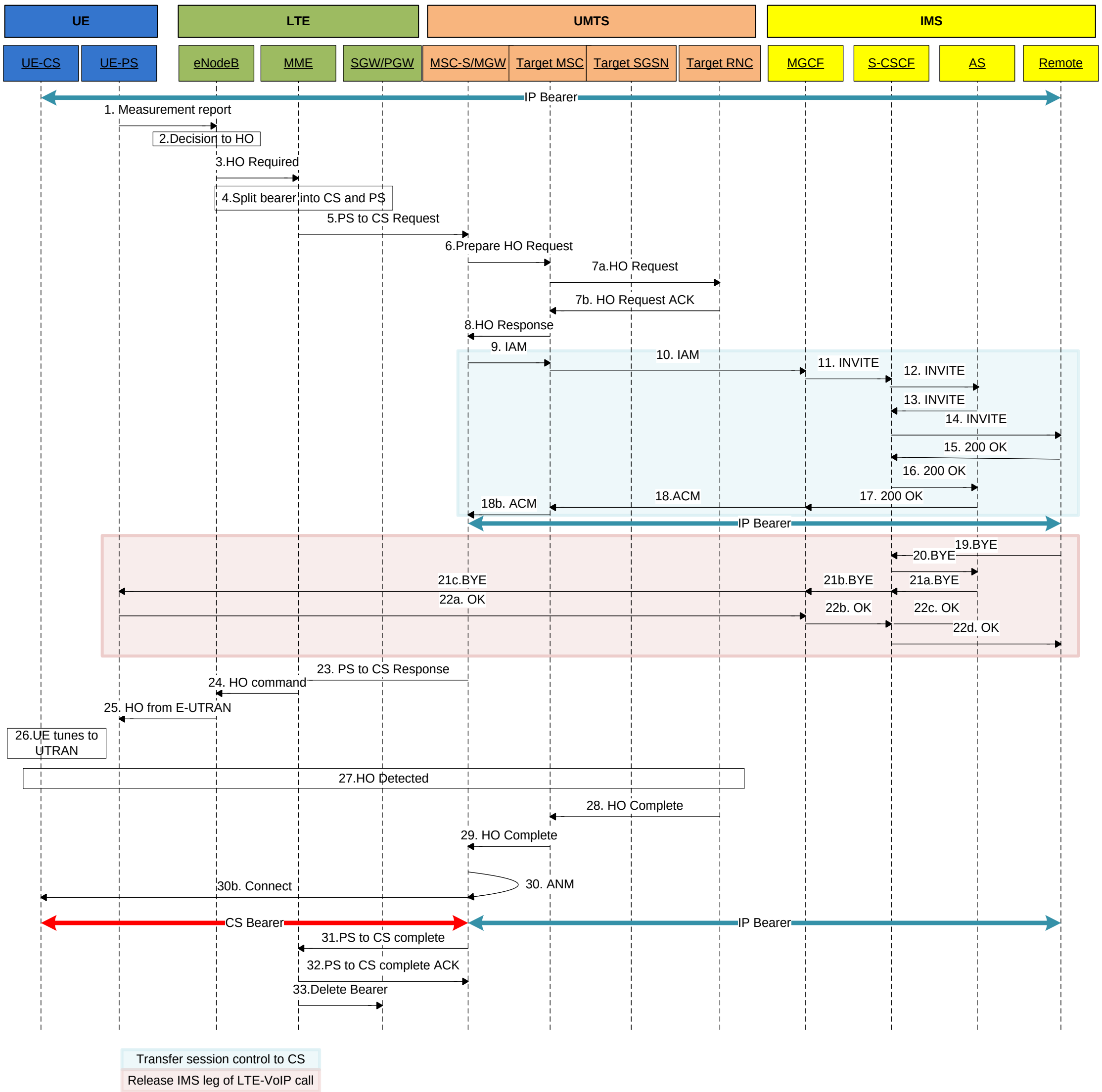

Figure 4.8: LTE to CS handover signal flow

[6] In SIP (Session initiation Protocol) signaling, a '200 OK' message is used to indicate an acknowledgment or a successfully received message.

The SRVCC procedure in Figure 4.8 is explained as follows;

Procedure 1~2: UE periodically measures the signal strength of neighbouring cells and sends a measurement report to the serving ENB. ENB decides if handover is necessary and sends a 'Handover Required' message to the MME indicating that this is an SRVCC operation.

Procedure 3~8: Based on the 'Handover Required' message, the MME classifies the request as CS+PS handover and splits the bearer into PS and CS and initiates the relocation towards SGSN and MSC Server respectively. The MME initiates PS to CS handover by sending a SRVCC 'PS to CS Request' message to the MSC Server. MSC server forwards the 'PS to CS Request' to the target MSC. The target MSC allocates resources for the call to the target RNC.

Procedure 9~18: The MSC server establishes a circuit connection between the target MSC and MGW using SS7/ISUP messaging. At the same time, the remote end is updated with details of the CS access call leg via the MGCF, thus establishing an 'understanding' between the MSC Server and the IMS remote end. The downlink flow of VoIP packets is switched towards the CS access leg.

Procedure 19~22: The remote party releases the former SIP session with the UE. Remote party sends SIP 'BYE' message to the UE and the UE responds with '200 OK' message.

Procedure 27~30: MSC server sends a SRVCC 'PS to CS Response' to the MME after the session immigration process (procedure 9~18) is completed. MME sends a 'Handover Command' to the UE. The UE tunes to UTRAN. Detecting the handover, the RNC sends 'Handover Complete' to target MSC, this message is transferred to the MSC server. A Speech circuit is connected in the MSC Server/MGW and the answer message (ANM) completes the circuit establishment procedure. The LTE to CS handover process is complete. A CS bearer now exists between the UE and target MSC/MGW while an IP/PS bearer exists between MSC/MGW and the remote end.

Procedures that occur after the 30th message do not really affect the delay of the HO process as they occur after establishment of the circuit. The mathematical analysis of the LTE to UMTS handover using the SRVCC is therefore based on Figure 4.8.

4.1.4 Voice Call Continuity (VCC)

Voice Call Continuity (VCC) is an IMS application that provides capabilities to transfer voice calls between the CS domain and the IMS network. VCC provides functions for voice call originations, voice call terminations and for domain transfers between the domain and the IMS and vice versa [52]. 3GPP has defined the Voice Call Continuity (VCC) specifications [52] in order to describe how a voice call can be maintained, as a user terminal roams between circuit-switched and packet-switched domains.

For VCC to be fully operational, enhancements must be made to the UE and to the network. In the case of the UE, the use of multiple access mode devices is mandatory. For example, a multi-mode device that is VCC capable may have LTE and UMTS modes. This means that the device has the radio and client functionality to access and register on either LTE or UMTS networks. Once registered with one of the networks, the device should be able to support all services available on the network. The UE must also be enhanced to understand and support the set of network modes which should be used [52].

Network enhancements must be made to the Core network to enable VCC. Figure 4.9 shows the VCC reference architecture with related interfaces. The architecture is made up of a UMTS Core network (shown in blue) and the IMS network (green). Note the inclusion of the VCC application (blue and green stripes) interconnecting with the IMS network and the MSC server using the Customized Application for Mobile Enhanced Logic (CAMEL) protocol [52].

In this research, the VCC procedure was used for handover from UMTS to LTE. In VCC handover, all calls, whether LTE or UMTS are anchored within the IMS network. If a subscriber requires to handover from one domain to another, the transfer will be done by the IMS network. It is important to note that the domain transfer is triggered by an Intelligent Network; hence making use of the CAMEL component shown in the architecture [52].

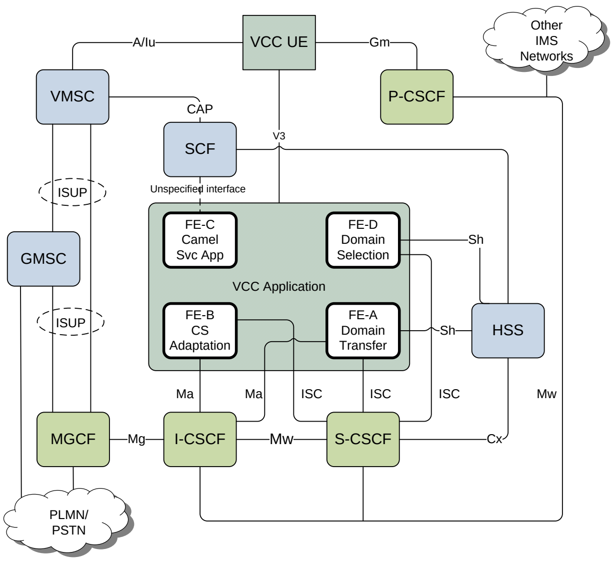

Figure 4.9: VCC Reference architecture

4.1.4.1 **Information flows for UMTS to LTE VCC handover**

In order for this type of UMTS to LTE handover to take place, the CS call must first be set up and then transferred to the IMS domain. Figure 4.10 shows the call setup using VCC in UMTS CS with subsequent IMS anchoring. Note the use of an Intelligent Network for CAMEL triggering of the VCC component.

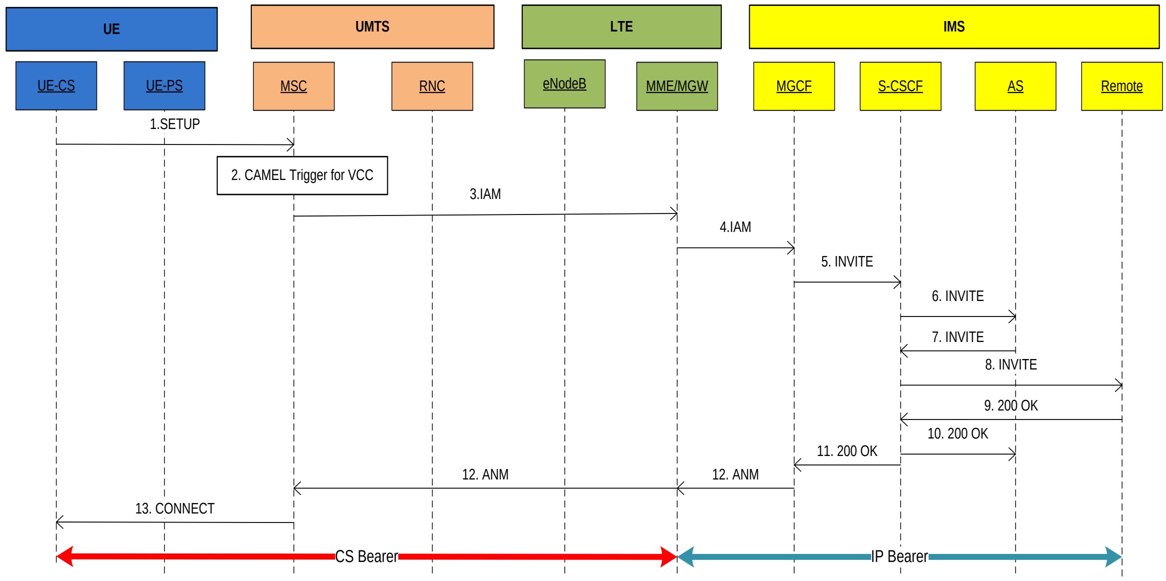

Figure 4.10: UMTS CS Call setup with IMS anchoring

When the circuit-switched call in the figure above has been set up, a VCC capable subscriber being served by UTRAN can handover to E-UTRAN. The procedure leading up to the call being served by UTRAN is shown in Figure 4.11 and is explained as follows:

Procedure 1~3: UE periodically measures the signal strength of neighbouring cells and sends a measurement report to the serving NodeB. The NodeB sends a report to RNC which decides if handover is necessary and sends Handover required message to MSC. Based on information in HO required message, MSC decides whether or not to execute handover

Procedure 4~13: MSC decides that the call is destined for the LTE network hence the need for domain transfer. It passes on the responsibility of IMS registration/Call setup to the MGW/MME with 'Prepare HO request' message. The MME/MGW sets up a bearer with E-UTRAN, performs IMS registration and informs UE to handover to LTE access.

Procedure 14~ The UE tunes to E-UTRAN and sets up a radio link with the eNodeB.

Procedure 15 ~16 The UE requests the MME to setup a bearer. The MME responds with PDP context accept/ bearer setup message which contains the IMS registration information. The UE therefore does not need to perform a new IMS registration procedure as it was already performed by the MME.

Procedure 17~18: The UE sends 'handover complete' message to MME which then forwards the same message to the source MSC.

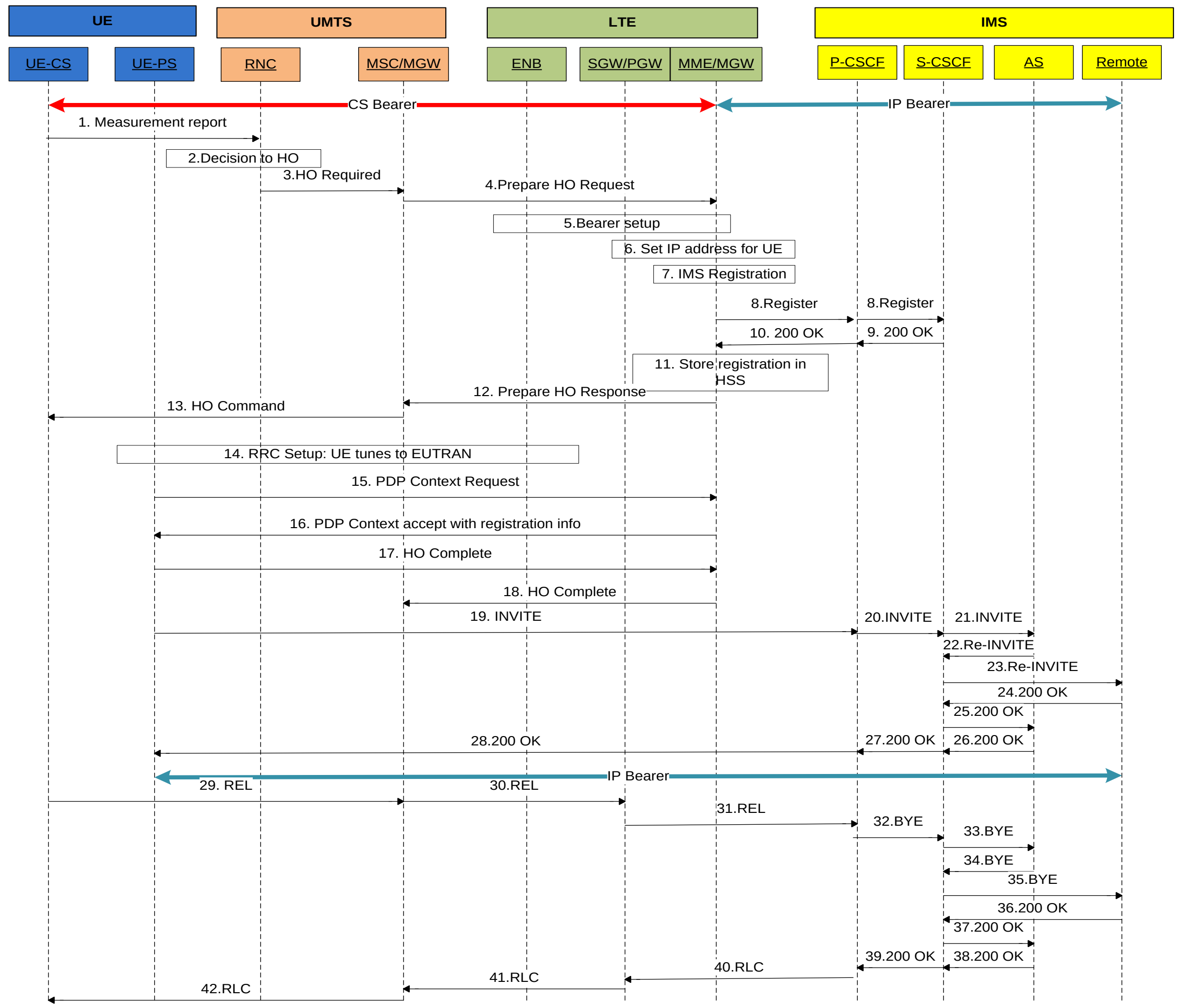

Figure 4.11: UMTS to LTE handover signal flow

Procedure 19~28: UE is now in PS mode and sends INVITE message to remote end via IMS network. When the AS receives the message, it detects that this was a domain transfer and hence changes the message to Re-INVITE and sends it to the remote network. The rest of the procedure follows standard IMS call set up with the call connected using the '200 OK' message. The call is now in progress over LTE access.

Procedure 29~42: the old access leg is ended.

It is important to note that UMTS to LTE VCC handover is more complex than LTE to UMTS SRVCC. This is because a UE has to make a location update on the E-UTRAN which is more complex than tuning to UTRAN. In addition the transfer of the call to the packet switched domain via IMS makes this procedure longer than SRVCC. This is due to the many SIP messages required to set up the new call leg.

4.2 Comparison of Inter-system handover techniques

4.2.1 SRVCC versus CSFB and VoLGA

Table 4.1 and Table 4.2 compare SRVCC, VCC, CSFB and VoLGA. Table 4.1 looks at the general business comparison while Table 4.2 compares the technical issues between SRVCC and CSFB since they are both 3GPP standards. From this comparison it is apparent that SRVCC and VCC are the long-term choice for LTE and UMTS interoperability. SRVCC has been endorsed by the ITU [53][54][55] and 3GPP as the preferred scheme for long-term LTE/UMTS/GSM interoperability. In this research report, we evaluate the performance of SRVCC and VCC using software simulations.

Parameter	SRVCC	CSFB	VoLGA	VCC
Viability	Long -term solution	Temporary solution	Temporary solution	Long term solution
Time to implement	Standard completed in Release 8; network to mature in 2011.	Standard is completed in Release 8; network to mature in 2010.	Standard in VOLGA will be completed in 2009. Network to mature in 2012.	Standard completed in Release 7. Network to mature in 2011
Operator support	Supported by most operators	NTT DoCoMo Japan	Only T-Mobile USA	Supported by most operators
Focus	VoIP controlled by IMS	No VoIP control	VoIP controlled by CS	VoIP controlled by IMS
Cost	Cost is high but it is the most feasible	Initial cost is low but performance is not the best. High cost of handsets	Network cost is acceptable but handset cost is a problem	Initial cost is high but more sustainable

Table 4.1: Comparison of feasibility voice handover techniques

4.2.2 CSFB versus SRVCC

The two 3GPP standardized voice handover techniques are SRVCC and CSFB. They are compared in Table 4.2. From this comparison it is apparent that SRVCC is the obvious choice for providing continuity of circuit-switched voice services between LTE and UMTS.

Parameter	SRVCC	CSFB
Terminal capability	Single radio mode	Dual mode/single-standby or Dual mode/ dual-standby
Terminal customization	Less complex	Complex
IMS anchoring	Mandatory	Optional
Mobility to CS network	Only when the terminal roams out of LTE coverage area	For every mobile originating and mobile terminating call
Cost	Initially expensive for operators without IMS core. Less expensive once deployed.	Cheap to deploy but expensive running cost due to high signaling load
Voice call setup time	Less as time is required only when the terminal moves out of LTE coverage area	Longer. Terminal needs to establish voice call session with CS network for every MOC and MTC.

Table 4.2: Comparison of SRVCC and CSFB

5 Key Research Question

5.1 Introduction

In trying to identify the research question, it was imperative to clearly define the problem in the form of a question. This was done in Chapter One and was stated as: "***What is the best voice handover scheme between LTE and UMTS networks with regard to 3GPP specifications".*** This was the key question that this research sought to answer. The question was answered by splitting it into two parts. The first part is, ***"What is the voice best handover scheme between LTE and UMTS networks?"*** and the second part is *"**with regard to 3GPP specifications?"***. In this chapter, we further discuss these two questions and how they were answered.

5.2 Selecting the best Handover Scheme

The choice of an optimal handover scheme between any two radio access networks is one that is difficult and that is subject to immense scrutiny by the stakeholders involved. Network operators and equipment vendors may choose a handover scheme based on several factors such as, the cost of implementation, the ease of deployment, handset capabilities, technical manpower required and 'political factors'. At present, there are four main voice handover schemes that are being investigated by the stakeholders in the telecommunications industry. These were explained in the previous chapter as Circuit Switched Fall Back (CSFB), Voice over LTE Generic Access (VoLGA), Voice Call Continuity (VCC) and Single Radio Voice Call Continuity (SRVCC).

Today, network operators are making different choices when it comes to a handover scheme between LTE and UMTS networks. For example, NTT DoCoMo and KDDI who are both big network operators in Japan have shown interest in CSFB. NTT DoCoMo has published a white paper on CSFB [56]. On the other hand operators in the USA and Canada such as T-Mobile, Rogers Wireless and Cincinnati Bell have always leaned towards UMA standards and have been said to be ready for VoLGA [57] as they already have the network architecture (UMA) to support this type of scheme. On the contrary, European operators such as Vodafone and Orange are bent on 3GPP standards such as VCC and SRVCC [53]. Each one of these operators has preference for a particular inter-RAT voice handover scheme for different reasons.

In the same vein, this research work selected SRVCC and VCC as the handover schemes of choice based on three main factors:

First, was the fact that SRVCC and VCC schemes are clearly defined by 3GPP for handover between LTE and previous 3GPP technologies namely UMTS, GSM and CDMA. 3GPP is the same standards body that defined the most widely used cellular technologies in the world such as GSM, UMTS, CDMA and more recently LTE. It is therefore expected that SRVCC and VCC uptake will match the deployment of 3GPP cellular technologies. In addition, SRVCC and VCC have so far been embraced by industry stakeholders such as ZTE [58]. ZTE tested voice over LTE using the SRVCC solution at the 2010 Mobile Asia Congress and recommended the use of SRVCC as the voice handover scheme of choice between LTE and 2G/3G networks [58].

Secondly, although CSFB is also defined by the 3GPP, SRVCC and VCC have been deemed to be more advantageous than CSFB. This is because they remove the need for a mobile terminal to camp on two radio networks simultaneously thus reducing user terminal complexity. In addition, SRVCC and VCC do not require a user terminal to fall back to the circuit switched domain every time they make or receive a voice call. This therefore reduces the amount of network signalling thus a cost saving on network resources and superior user experience.

Thirdly, at the beginning of this research work, Vodacom South Africa had just tested the first known LTE network in Africa. It had therefore been hoped that there would be a network backbone to support SRVCC and VCC research work by 2011. However, works at Vodacom are still underway and no experimental work can be done until the system is fully operational. It is still unknown when these works will be completed.

Based on the above three factors, the voice handover schemes chosen for evaluation were SRVCC for LTE to UMTS handover and VCC for UMTS to LTE handover. This therefore partially answered the first part of the research question "***What is the best voice handover scheme between LTE and UMTS...***"

5.3 Evaluating the best Handover Scheme

In an attempt to answer the second part of the problem statement, "***with regard to 3GPP specifications"***, it was necessary to either perform experimental work on live UMTS/LTE/IMS systems, or implement 'dummy' LTE/UMTS networks or carry out simulation using software and then compare the results with 3GPP specifications. Since we did not have network resources to perform experimental work for the handover schemes we made use of software tools, namely MATLAB.

SRVCC and VCC are both defined by 3GPP which provides recommendations on the handover performance. The evaluation was therefore based on 3GPP recommendations. The performance results obtained from MATLAB were assessed in comparison to 3GPP specifications. It would have been more valuable to compare the performance to other research work done on SRVCC and VCC. This would further expound on the authenticity of SRVCC and VCC as the best schemes for voice handover between LTE and UMTS. However, at present, research work on this subject is still rare or proprietary and thus not available to the general public.

5.4 Conclusion

In this chapter, we covered the research question in detail. The research question was split into two parts, that is, choosing the best scheme and, evaluating its performance based on 3GPP specifications. The available software and academic resources were used to answer the question. Although this work was purely for academic purposes, network operators wishing to deploy a handover scheme between any two wireless networks may use the approach explained above to choose the best scheme.

6 Methodology

6.1 Introduction

Chapter Four discussed various voice handover schemes for LTE and UMTS. In this chapter, we have analysed the selected handover techniques using mathematical models and MATLAB software. The techniques that were experimentally analysed were SRVCC in the case of LTE to UMTS handover and VCC for UMTS to LTE handover.

In order to evaluate the interruption experienced by a call going from UMTS to LTE and vice versa, the experienced handover delay was split into three parts namely [59]; delay on the radio link, network node queuing delay and remote/Internet network delay. Each type of delay follows a unique mathematical behaviour. Sections 6.2, 6.3 and 6.4 discuss how the three components of delay were mathematically modelled.

6.2 Radio link delay

For Inter-RAT handover to happen there must be communication between two radio technologies, in this case, UMTS and LTE. Therefore, a radio delay happens during setup on the UMTS and LTE access networks. Disturbance on the radio link is difficult to predict due to the random nature of the surrounding environment. Problems such as multipath fading, shadowing, interference, Doppler Effect all affect a radio signal. Within the UTRAN and E-UTRAN, the Radio Link Control Protocol (RLC) provides segmentation and retransmission services for user and control data to improve delay performance, responsiveness and resource utilization for reliable data transfer on the access network. Analysis of the RLC can therefore be used to measure the delay caused over the radio link as RLC detects packet loss and performs retransmissions. In this research, the RLC protocol in the UMTS and LTE accesses was assumed to have the same functions and behaviour. The functions of RLC were explained in chapter three.

The UMTS RLC protocol is defined by 3GPP in [60]. Several authors have developed RLC models, some of the papers that describe UMTS RLC models include, [61][62][63] and [64]. The model used for this research was taken from [62] and is shown in Figure 6.1. RLC delay shown in the figure is composed of the following parts;

- T_{proc}, the processing delay of an RLC frame
- T_{Iub}, latency on the Iub interface (between RNC and NodeB), assumed to be independent of the size of the block being transported
- TTI, the transmission time interval at the Node B (one radio frame for every TTI)
- T_{ack}, the time between detection of a missing or erroneous frame on the receiving side and transmission of a frame status to the sender.

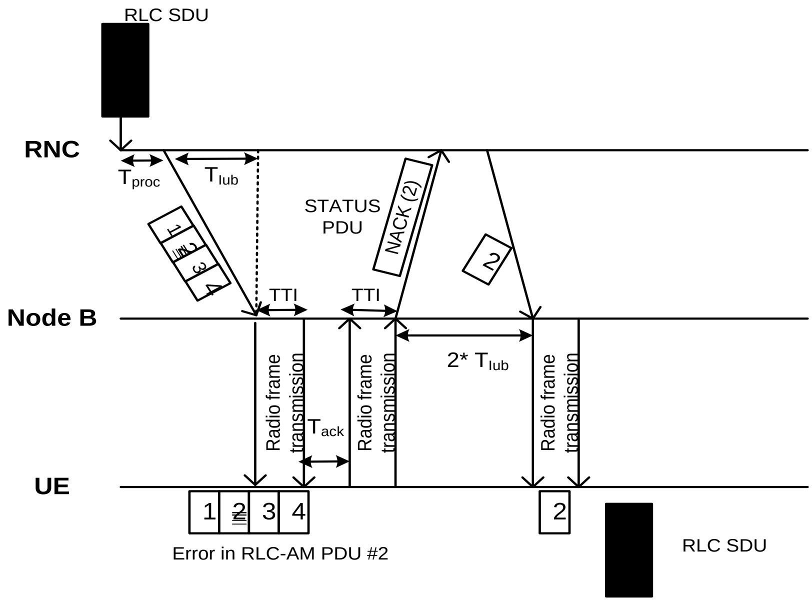

Figure 6.1: Frame transfer in UTRAN with RLC

Assuming an error free channel and all blocks transmitted once and the RLC buffer is empty, the resulting delay can be written as;

$$D_1 = T_{proc} + T_{Iub} + m * TTI \quad (6.1)$$

Equation 6.1: Delay under ideal conditions

Where, m is the required number of TTI to send a frame. If T_1 is the time between the detection of an erroneous RLC sub-frame and the reception of its retransmission and assuming all retransmissions are within one TTI, then we can write;

$$T_1 = T_{ack} + 2 * (TTI + T_{Iub}) \quad (6.2)$$

Equation 6.2: Time of detection of erroneous RLC sub-frame

In order to account for errors, we introduce a, the number of transmissions of the last correctly received sub-frame. Delay, D_a, can be expressed as a function of a.

$$D_a = D_1 + (a - 1) * T_1 \quad (6.3)$$

Equation 6.3: Delay after a retransmissions

Substituting $D_1$and T_1 into Equation 6.3 we obtain;

$$D_a = T_{proc} + T_{Iub} + mTTI + (a-1)[T_{ack} + 2(TTI + T_{Iub})] \tag{6.4}$$

Equation 6.4: Delay corresponding to a transmissions

The probability density function of delay can now be calculated. Consider the probability of receiving an error frame on the radio link (BLER) to be, p. The probability that a frame is correctly received after utmost a transmissions is $(1-p^a)$. For k frames, we can write the probability of successfully receiving a frame at the a^{th} transmission as:

$$P_{(D\leq D_a)} = (1-p^a)^k$$

$$P_{(D=D_a)} = P_{(D\leq D_a)} - P_{(D\leq D_{a-1})}$$

$$P_{(D\leq D_a)} = (1-p^a)^k - (1-p^{a-1})^k$$

$$\overline{D} = \sum_{a=1}^{a_{max}} (D_a * P_{(D=D_a)}) \tag{6.5}$$

Equation 6.5: Mean Delay for 'k' frames after 'a' transmissions

In general the probability of successfully receiving a frame on the RLC network after n retransmission trials can be written as [61];

$$P_s = 1 - p(p(2-p))^{\frac{n(n+1)}{2}} \tag{6.6}$$

Equation 6.6: Probability of success for RLC frame

From Equation 6.4 and Equation 6.6, the overall delay can be rewritten as Equation 6.7. Note the removal of T_{proc} as it is accounted for by queuing model in the next section [61].

$$T_{RLC} = T_{Iub} + (k-1)TTI + \frac{k(P_s-(1-p))}{P_s^2}\left\{\sum_j^n \sum_i^j \left[P\left(C_{ij}\right)(2jT_{Iub} + \left(\frac{j(j+1)}{2} + i\right) * TTI)\right]\right\} \tag{6.7}$$

Equation 6.7: Overall delay over RLC after n retransmissions

Where;

$\overline{D}$: is the mean delay

k: number of frames to be transmitted

n: number of RLC retransmissions (in case of an erroneous frame)

p: probability of RLC frame being received erroneously over radio link

P_s: probability of receiving RLC frame successfully after n retransmissions

T_{Iub}: the latency on the Iub interface (between RNC and NodeB)

TTI: the transmission time interval at Node B (one radio frame for every TTI)

T_{ack}: The time between detection of a missing or erroneous frame on the receiving side and transmission of a frame status to the sender.

$P(C_{ij})$: The first correctly received frame at destination (i^{th} retransmitted frame after j retransmissions)

In this research, the simulation of RLC delay was done using Equation 6.7 with the following parameters for both LTE and UMTS.

Parameter	**Value**
Block Error Rate (p)	1% : 4%: 37%
Data rate (UMTS-CS, LTE)	9.6-128kbps, 1-100Mbps
RLC frame size	7680bits
n_{max}	4
TTI(UMTS, LTE)	20ms, 10ms[65]
T_{Iub}(UMTS, LTE)	3µs, 1 µs

Table 6.1: Parameter list for RLC analysis

6.3 Network Node Queuing Delay

Each node within the UMTS core network or the EPC has a queuing behaviour which can be modelled using queuing theory. In this research, we applied the M/M/1 queuing theory to model the queuing behaviour of UMTS and LTE nodes. In queuing systems, the M/M/1 model [66]describes negative exponential (Poisson) arrivals and service times with a single server and FIFO (or not specified) queue of unlimited capacity and an infinite customer population. M/M/1 is a good approximation for a large number of queuing systems.

If we consider an MSC Server in the UMTS network, we can assume it has a traffic arrival rate, λ and service rate, µ. We then define the ratio, ρ as the measure of demand on the queue in relation to the capacity.

$$\rho = \lambda/\mu \tag{6.8}$$

Equation 6.8: Measure of demand on a queue

The average length of an M/M/1 queue can be approximated to

$$L \; = \rho/((1-\rho)\,) \qquad (6.9)$$

Equation 6.9: Average queue length

Applying Little's theorem $L = W\lambda$, where W is the mean waiting time for a message waiting to be served by the MSC Server, it follows that;

$$W = \rho/(\lambda(1-\rho)) \qquad (6.10)$$

Equation 6.10: Average waiting time in queue

In this research report, the assumed values of arrival rate and service rate of network nodes in the UMTS, LTE and IMS networks are summarized in Table 6.2 below. With these values, the average waiting time for a message waiting in a queue of any given node was calculated. The simulation of network node queuing was based on these waiting times.

UMTS Network							
	UE	NodeB	RNC	MSC	HSS		
λ	50	100	200	300	300		
µ	2500	2500	5000	5000	5000		
ρ	0.02	0.04	0.04	0.06	0.06		
W(ms)	0.4	0.4	0.2	0.2	0.2		
LTE/IMS Network							
	UE	P-CSCF	MGCF	AS	ENB	MME	S-CSCF
λ	50	500	500	500	100	900	500
µ	2500	5000	5000	5000	5000	5000	5000
ρ	0.02	0. 1	0.1	0.1	0.05	0.18	0. 1
W(ms)	0.4	0.2	0.2	0.2	0.5	0.2	0.2

Table 6.2: Theoretical Service and Arrival rates of network nodes

6.4 Internet and Remote Network Delay

Delay stemming from the Internet or remote network queuing is unstable and varies based on various conditions at the remote end. A more advanced queuing model was chosen, viz. the M/G/1 model [66][67]. In this model, the traffic arrival rate λ follows a Poisson distribution and the traffic service time is described by a mean value, while the service time distribution is not specified. The M/G/1 queue does not have a general, closed form distribution for the number of jobs in the queue in steady state. It does, however, admit a general solution for the average number of jobs in the queue, and application of Little's Theorem provides the corresponding result for the average time spent in the queue. Collectively, these results are known as the Pollaczek-Khinchine mean value formulae[68].

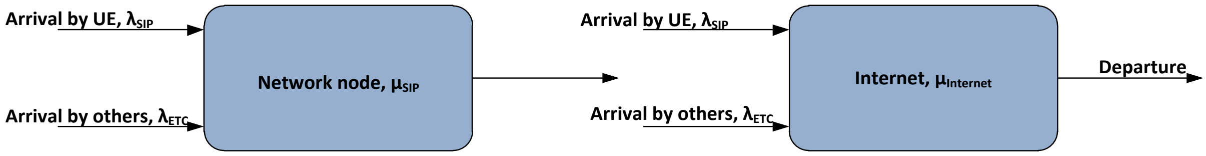

Figure 6.2: M/G/1 Model for Internet queuing

Figure 6.2 shows arrival of traffic at an IMS network node and the subsequent arrival and departure at the Internet. Traffic is divided into that which is of interest to the research i.e. λ_{IMS} and other traffic destined for the Internet λ_{ETC}. Delay at the remote network is a result of queuing at both IMS network and Internet while delay at the IMS network is only a result of IMS traffic. For the M/G/1 queue, Pollaczek-Khinchine's mean value formula is used with the following assumptions;

1. The server is busy if the queue is non-empty,
2. No job leaves the queue before completing service,
3. The order of service is independent of knowledge about job service times.

If X_{IMS} and X_{ETC} are the service time for an IMS message and other network messages respectively while $\overline{X_{ETC}^2}$ and $\overline{X_{IMS}^2}$ are their second moments. λ_{ETC} and λ_{IMS} are the respective message arrival rates while ρ_{ETC} and ρ_{IMS} are the related server utilizations. Applying Little's formula and taking into account the residual service time, the average waiting time for an IMS message waiting for a response from the Internet or Remote network can be written as in the expression below [59] [68]:

$$W_{Internet} = \frac{(\lambda_{ETC}+\lambda_{IMS})(\overline{X_{ETC}^2}+\overline{X_{IMS}^2})}{2(1-\rho_{ETC}-\rho_{IMS})} \tag{6.11}$$

Equation 6.11: Average waiting time at Remote Network

In this report, the simulations for the SIP network and the Internet were done for a period of 1000ms. The estimated values of the parameters used are shown in Table 6.3.

Parameter	**Value**
Server utilization, $\boldsymbol{\rho}$	0.02
Arrival rate at SIP network	50 (per second)
Arrival rate at Internet	1250 (per second)
Observation interval	1000ms

Table 6.3: Parameters used in M/G/1 simulation

6.5 Mathematical Expressions for Service Interruption Time

In this section, we apply the mathematical models discussed in the previous section to derive the service interruption time. To analyse the overall performance of circuit-switched handover from UMTS to LTE and vice versa, the components of the total service interruption time are summed to obtain the overall delay. Service interruption time in inter-RAT handover is the time between the last received transport block on the old frequency and the time the UE starts transmission of the new uplink channel.

6.5.1 LTE to UMTS SRVCC Analysis

A user moving from LTE to UMTS coverage will handover LTE to UMTS/CS, using the SRVCC procedure. In order to derive the expression for service interruption of LTE to CS SRVCC handover, it is important to understand the path of the message signalling flow. Figure 6.3 shows the physical path when UE initiates a call from the E-UTRAN system. A UE using the LTE radio network to make a new call has a control path through ENB, CSCF and IMS AS and the user bearer path formulated through ENB and SGW.

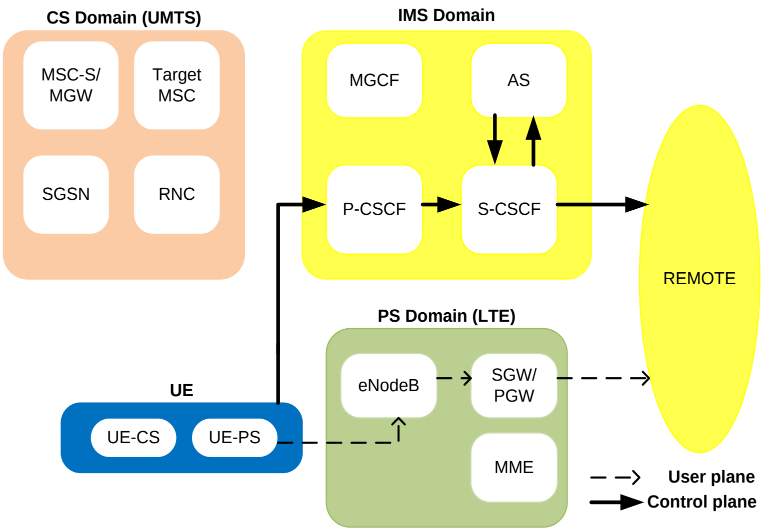

Figure 6.3: Entities involved in LTE Mobile Originated Call

The message signalling flow for LTE to UMTS SRVCC procedure is described in Chapter Four in Figure 4.8. Figure 6.4 is an extract of the SRVCC procedure of the service interruption time during this handover. Service interruption time is the time when the old call leg is released within the LTE network and setup in the UMTS network. The delay should be imperceptible by the user. We can write the service interruption time in terms of already discussed models as follows;

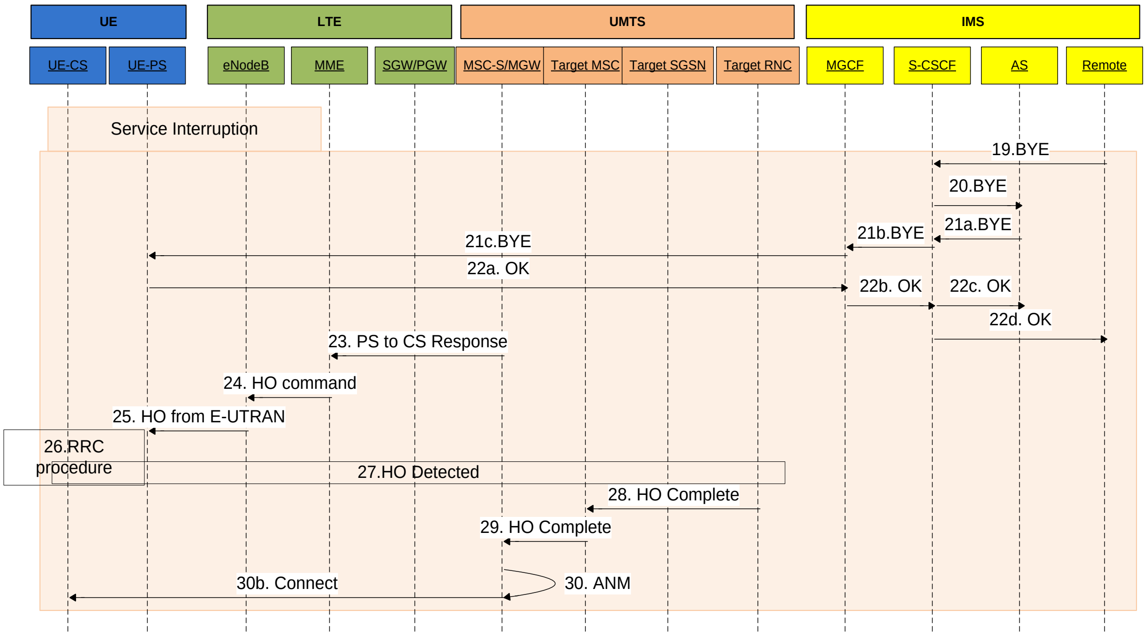

Figure 6.4: LTE to UMTS service interruption

From Figure 6.4, the expression for LTE to UMTSSRVCC service interruption can be written as

$$D_{LTEtoUMTS} = D_{Remote} + D_{Internet} + 2D_{AS} + 3D_{CSCF} + 2D_{MGCF} + D_{UE} + 3D_{MSC} + D_{MME} + D_{eNodeB} + D_{RRC} + D_{RNC} \quad (6.12)$$

Equation 6.12: LTE to UMTS service interruption

Where D_x is the delay experienced by a message any network node x or remote network x

6.5.2 UMTS to LTE VCC Analysis

Consider a circuit-switched call in progress over the UMTS network anchored in IMS [52]. A user travelling from UMTS to LTE coverage area will handover to LTE using the VCC procedure. The Call setup procedure in LTE follows the diagrams below. In this procedure, the old call is only severed after the new call leg is fully established.

From Figure 6.6, the mathematical expression representing the overall latency for handover can be written as Equation 6.13. Note that latency occurs after the MME issues the Handover command. From here on until a new call leg is created, the call is interrupted. It is this interruption time we are interested in. The expression for service interruption time for UMTS to LTE handover is therefore Equation 6.13.

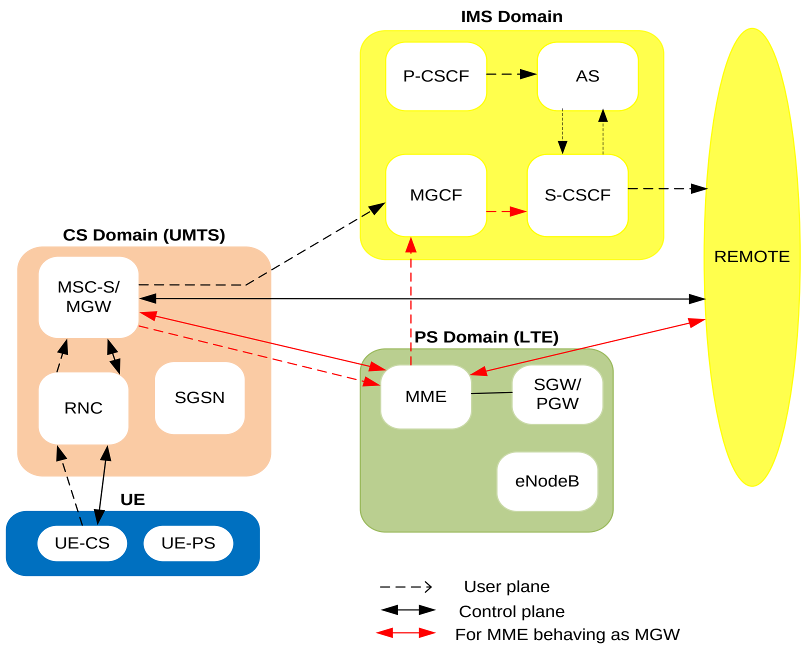

Figure 6.5: Call Setup for UMTS-CS with anchoring in IMS

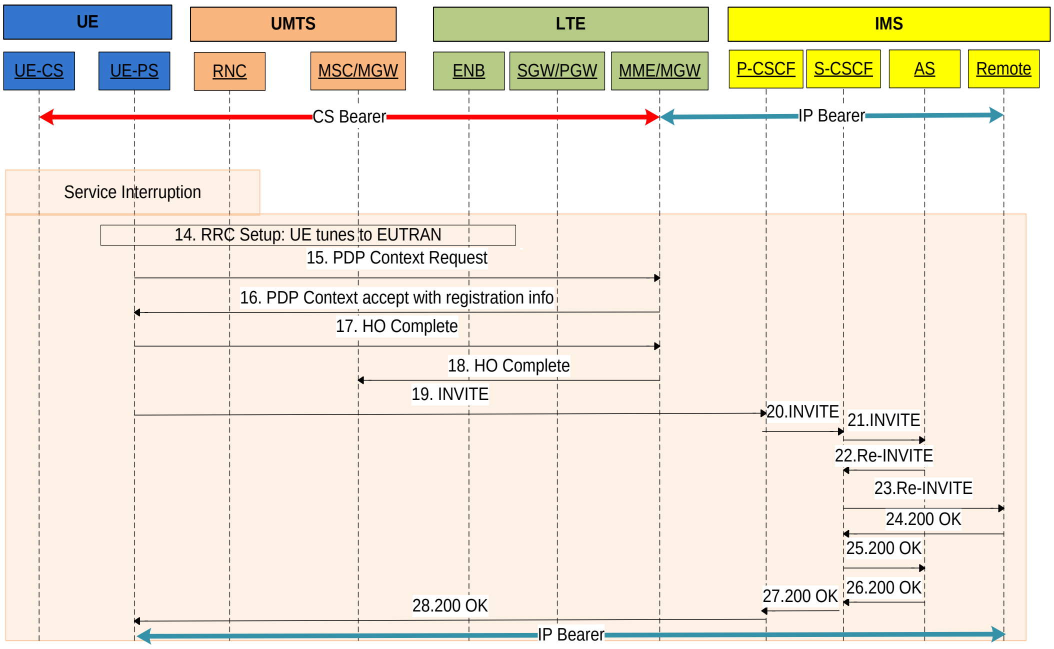

Figure 6.6: UMTS to LTE VCC service interruption time

$$D_{UMTStoLTE} = 2D_{Remote} + 2D_{Internet} + 2D_{AS} + 6D_{CSCF} + D_{UE} + 2D_{MME} + D_{RRC} \qquad (6.13)$$

Equation 6.13: Service interruption time for UMTS to LTE handover

6.6 Simulation

MATLAB® simulations for Equation 6.12 and Equation 6.13 along with the related models were run with theoretical network parameters shown in Table 6.1, Table 6.2 and Table 6.3. The network parameters were from typical values usually found in UMTS, LTE and IMS networks [59]. Simulations were done in two phases described in the ensuing sections.

6.6.1 Phase One

The first phase of the simulation was done with a static network. In here, it was assumed that a single user terminal, with a predefined M/M/1 queuing behaviour and the parameters shown in Table 6.2 was communicating with the UMTS, LTE, IMS and remote network nodes which also had static /predefined parameters shown in Table 6.3. The nodes were presumed to be independent of one another. Figure 6.7 illustrates the basic model of the first simulation.

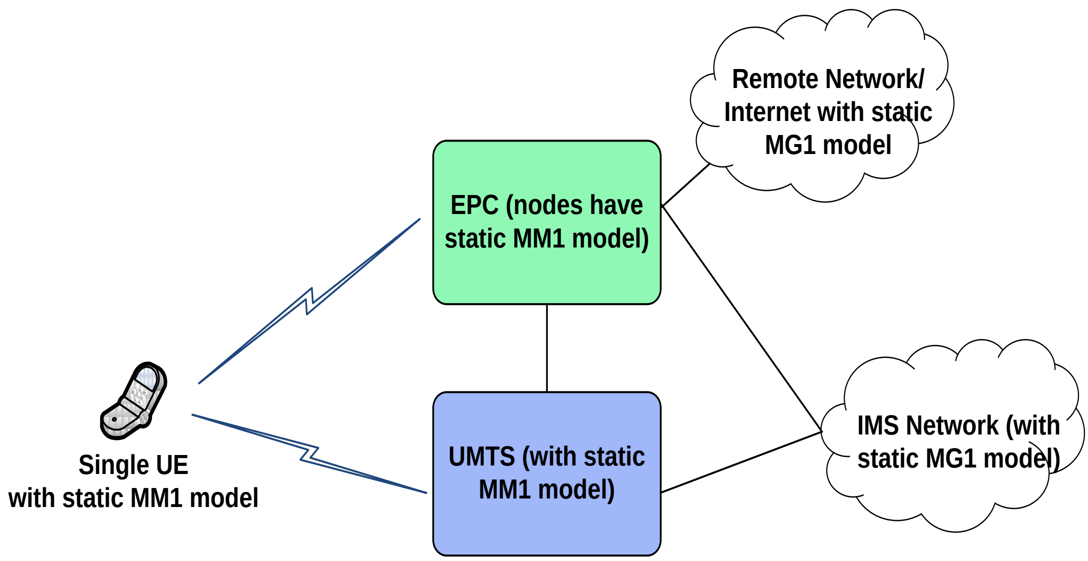

Figure 6.7: Model for Phase 1 simulation

Consider the simulation of the delay in Equation 6.13. That is to say the latency resulting from a single terminal with a voice calls being transferred from UMTS to LTE. The equation was split into four components, namely; radio delay, node delay, remote network delay and total delay. The MATLAB simulation files were therefore written into these four types of latency. Within each file, an algorithm specific to the type of delay/latency was coded. This is further explained as follows;

- Radio propagation delay MATLAB file: In this file, an algorithm for RLC behaviour (described in Equation 6.7) with the parameters listed in Table 6.1 was written and results were obtained after the file was run.

- Node network delay: in this file, an algorithm for M/M/1 queuing behaviour for a node described by Equation 6.10 was written. The input parameters in phase 1 were static, e.g. ρ.
- Remote network delay: an algorithm for M/G/1 queuing behaviour described by Equation 6.11 was written. Input parameters were static.
- Overall delay: In this file, the overall delay equation was written. This file "calls-upon" the three previous files.

6.6.2 Phase Two

The second phase of the simulation was done with real-time queuing conditions. The observation was done for a 300ms interval. Messages arriving at the network nodes and remote network were treated in real-time. For example as the user terminal under observation accessed the UMTS and LTE network nodes, the nodes were also processing messages from other user terminals. This was done so as to capture the random nature of network traffic. This is illustrated in the figure below.

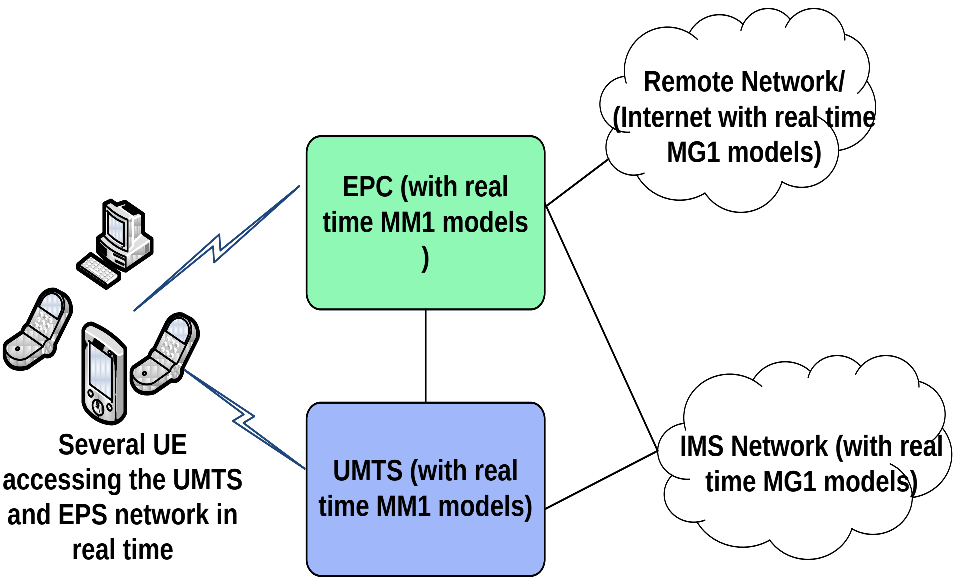

Figure 6.8: Model for Phase 2 Simulation

As in the previous section, the simulation of the delay in Equation 6.13 was done. That is to say the delay caused by a voice call being transferred from UMTS to LTE. The equation split into three components, namely, radio delay, node delay remote network delay and total delay components. The MATLAB simulation files were therefore written into four types of delay. Within each file, an algorithm specific to the type of delay/latency was coded. This is further explained as follows;

- Radio propagation delay MATLAB® file: In this file, an algorithm for RLC behaviour (described in Equation 6.7) with the parameters listed in Table 6.1 was written and results obtained. However, this algorithm was set to run for 300ms.
- Node network delay: in this file, an algorithm for M/M/1 queuing behaviour described by Equation 6.10 was written. This was also set to run with a node generating random traffic for a period of 300ms with several messages arriving at the network node randomly.
- Remote network delay: an algorithm for M/G/1 queuing behaviour described by Equation 6.11 was written. This was also done with the remote network receiving and sending random information over a 300ms period with several messages arriving at the remote network randomly.
- Overall delay / service interruption time: In this file, the equation for the total service interruption time was written. This file "calls-upon" the three previous files.

The output obtained from the two simulation phases was plotted and are discussed in Chapter Seven.

7 Results

7.1 Introduction

In Chapter Six, Equation 6.12 and Equation 6.13 along with the three mathematical models (radio, node delay, and remote delay) were simulated using MATLAB. Parameters used for the simulation were shown in Table 6.1, Table 6.2 and Table 6.3. The simulation outputs for SRVCC and VCC schemes were obtained. The ensuing sections perform a critical analysis of the results that were obtained.

7.1.1 LTE to UMTS SRVCC analysis

Figure 7.1, Figure 7.2 and Figure 7.3 show the behaviour service interruption time for LTE to UMTS SRVCC handover under static network conditions. In these figures, the service interruption time was plotted for varying data rates, propagation delay and Block Error Rate (BLER) respectively. Note that there are two relationships; that between service interruption time and the BLER, and between service interruption time and radio propagation delay.

As illustrated in Figure 7.1, it was observed that for a given BLER, the service interruption time reduced considerably with increased data rate. For example, for a constant frame size of 7680 bits, a data rate of 9.6kbps yielded a service interruption of over 400ms, which was higher than the specified 300ms for real time services. Data rate equal to 64kbps yielded a service interruption time of slightly less than 250ms which was acceptable. However, for data rate of over 64kbps, there was no significant decrease in service interruption time. This is because CS messages have a specific frame length and 64kbps is sufficient to transmit this length.

At the air interface (between the UE and the eNodeB), a high BLER (Block Error Rate) and radio propagation delay can greatly affect the fidelity of a voice call. It is therefore important to compare the effect of both propagation delay and BLER on the service interruption time.

In Figure 7.2, the service interruption time/overall latency of the call was plotted versus the BLER for various values of propagation delay. The radio propagation delay (latency over the radio link) was set from 10^{-6} up to 10^{-2}. For a given propagation delay e.g. 10^{-2}, the service interruption time was observed as the BLER was varied. Notice that all plots were well within the specified 300ms interruption time. However, for propagation delay of 10^{-5} and 10^{-6} there was no significant change in the service interruption time, therefore the graph for latency due to a propagation delay of 10^{-5} and

10^{-6} are superimposed. These are shown by the red line with green dots between the vertical range of 240– 245ms in Figure 7.2. As the propagation delay increased from 10^{-3} to 10^{-2}, there was a sharp increase in the service interruption time. This behaviour is further illustrated in Figure 7.3. In this graph, the service interruption time is the same between 10^{-6} and 10^{-5} (hence the overlapping thin red line) but is clearly seen to rise faster when the propagation delay is increased from 10^{-4} to 10^{-3} and even more sharply for 10^{-3} to 10^{-2}.

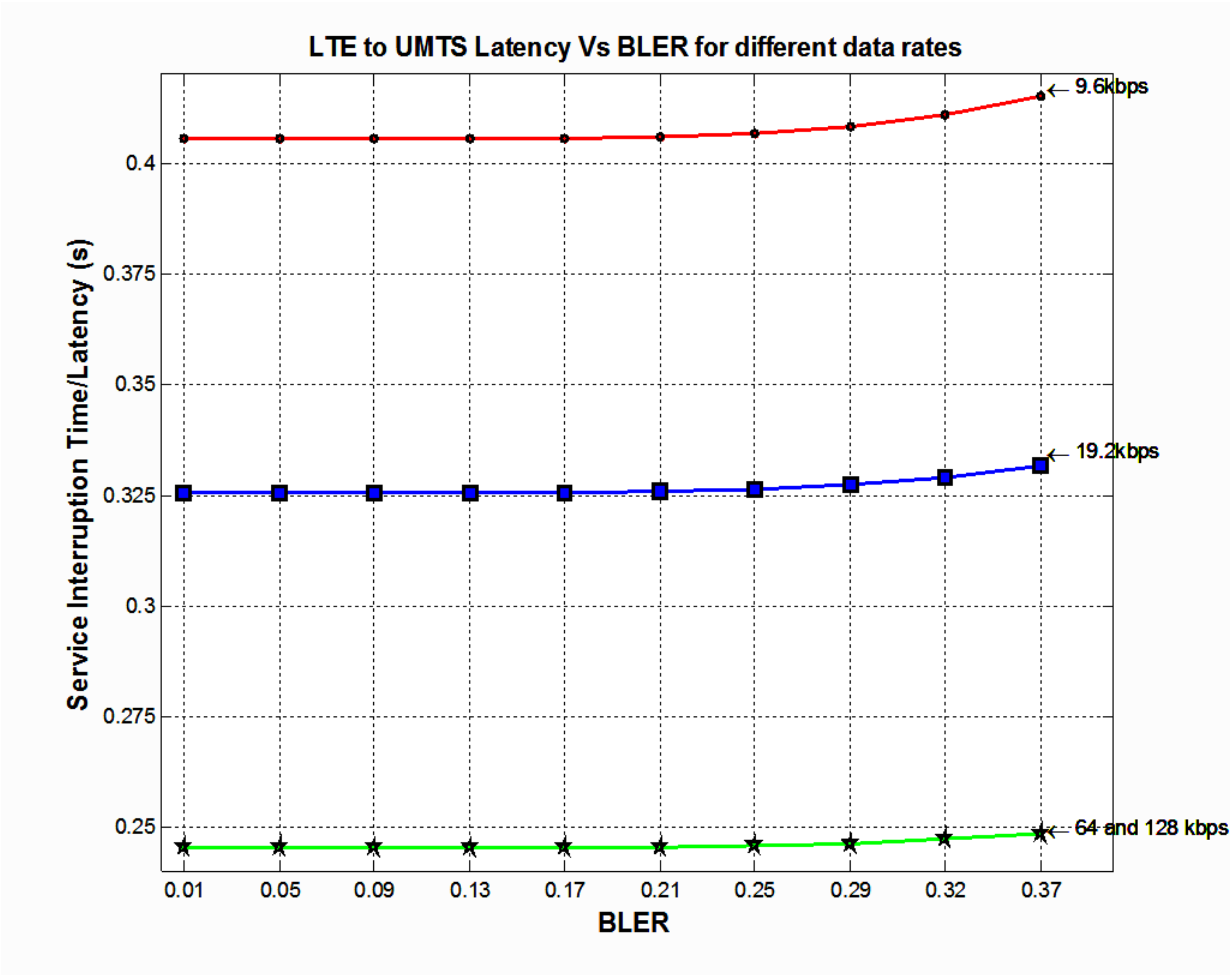

Figure 7.1: LTE to UMTS SRVCC service interruption time Vs BLER

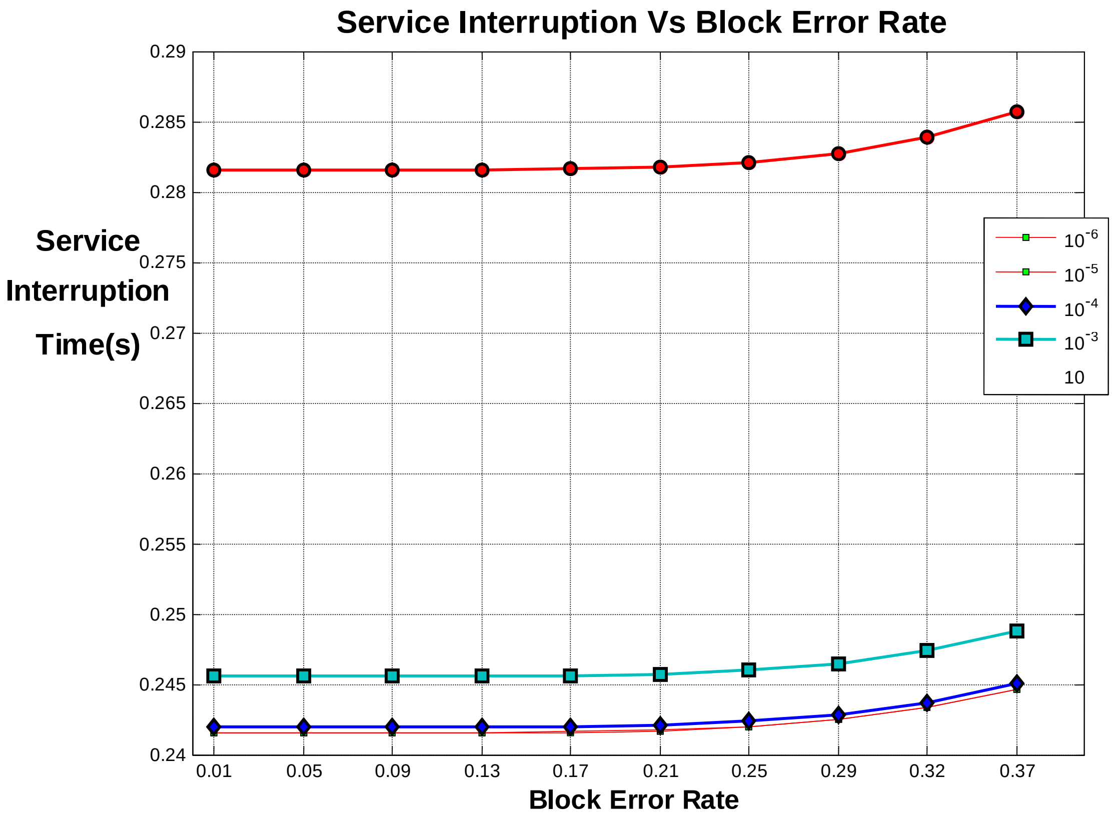

Figure 7.2: LTE to UMTS SRVCC service Interruption time Vs BLER for varying propagation delay

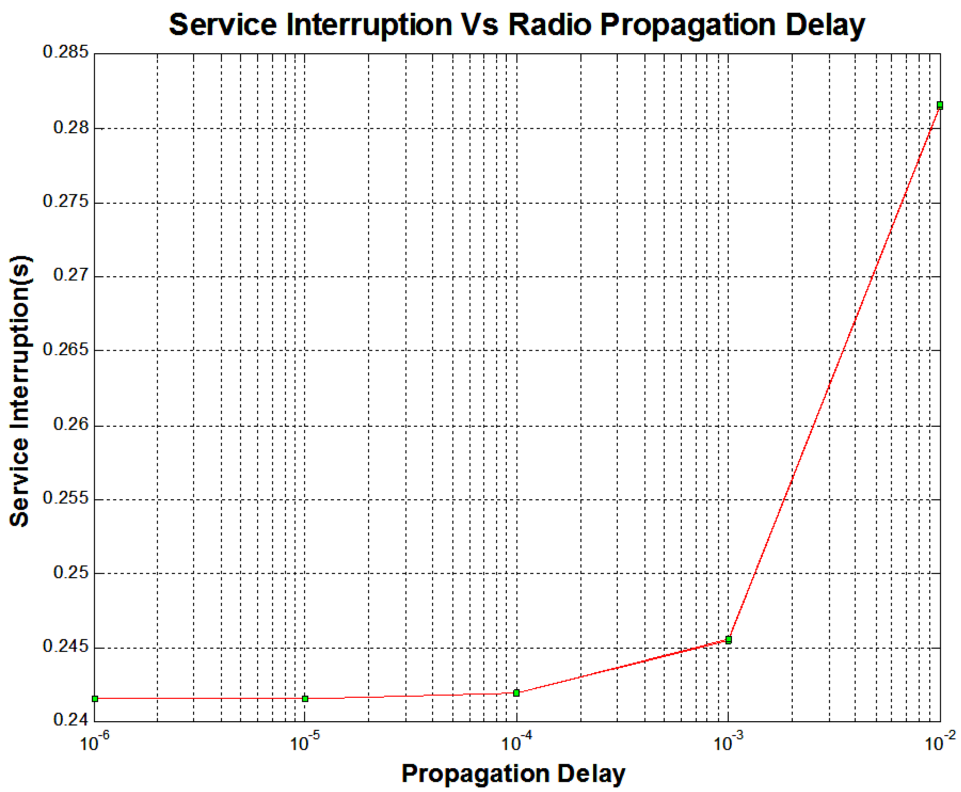

Figure 7.3: LTE to UMTS SRVCC Service Interruption versus Radio Propagation Delay

Under real-time network conditions, the relationship between service interruption time and BLER with varying radio propagation delay was found to be similar to static conditions. However, after running 500 real-time simulations, it was noticed that the service interruption was less than that under static conditions at all times. Figure 7.4 shows the interruption time versus the BLER for different orders of propagation delay under real-time conditions. Note that the interruption time is in the order of hundreds of milliseconds (100ms) unlike before where it was in the 200ms range.

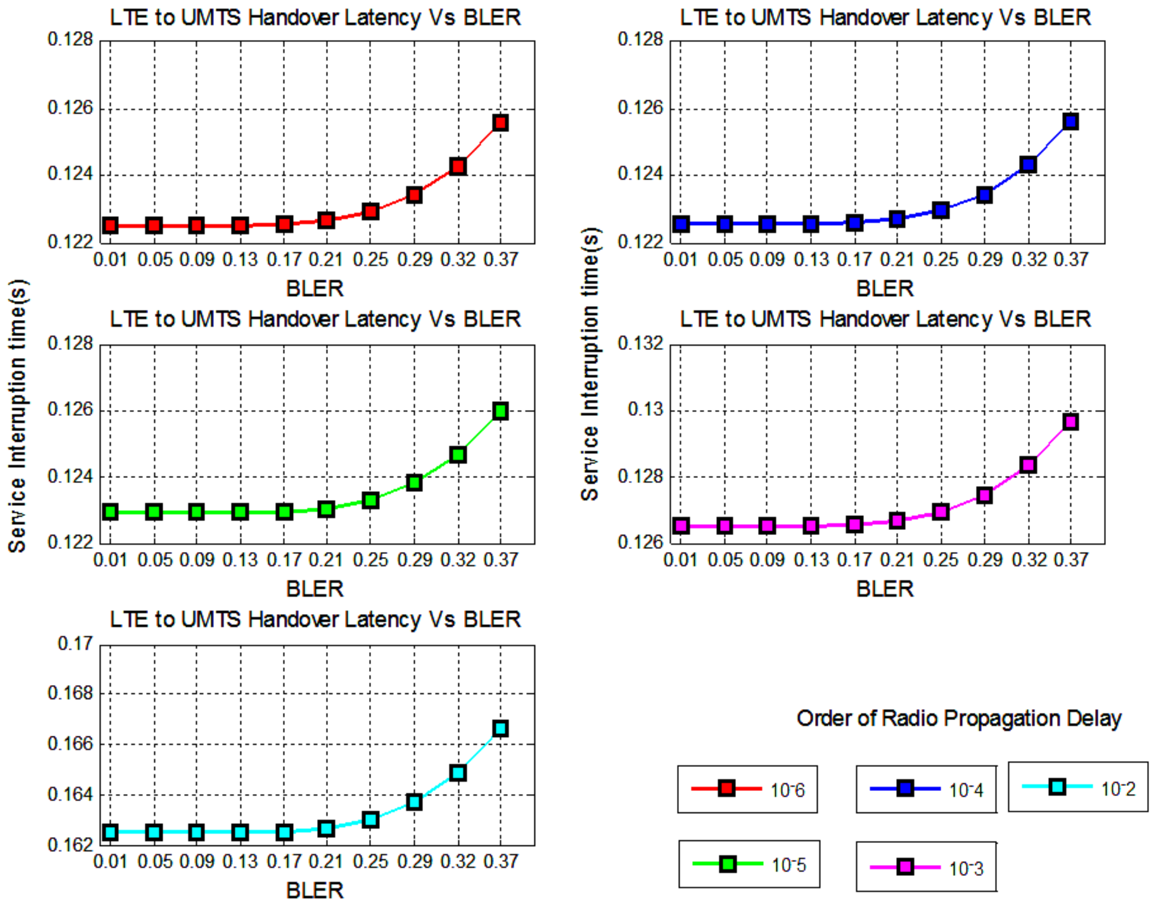

Figure 7.4: SRVCC Service interruption time under real-time network conditions

7.1.2 UMTS to LTE VCC handover analysis

Figure 7.5, Figure 7.6, Figure 7.7 show the behaviour of the service interruption time for UMTS to LTE VCC handover under static network conditions. As in the previous section, the service interruption time was plotted for varying data rates, propagation delay and Block Error Rate (BLER) respectively.

Note that for this type of handover, the service interruption time was much higher (order of 400 and 500ms) than LTE to UMTS SRVCC handover. This is because in addition to radio set up on the LTE access network, a new PDP context/EPS bearer and IMS registrations had to be established.

As in the LTE to UMTS SRVCC case, it was generally observed that for a given BLER, the service interruption time decreased considerably with increased data rate. However, for data rates higher than 1Mbps, there was no significant decrease in the service interruption time. This is due to the constant frame size. 1Mbps is sufficient to transmit a control signal in one frame; therefore increase in data rate does not improve performance.

In Figure 7.5 it can also be seen that as the BLER increases, the service interruption time gradually increases. When BLER is higher than 21%, there is an exponential increase in service interruption time. Again, it's important to note that for UMTS to LTE handover, a BLER higher than 21% adversely affects service interruption time.

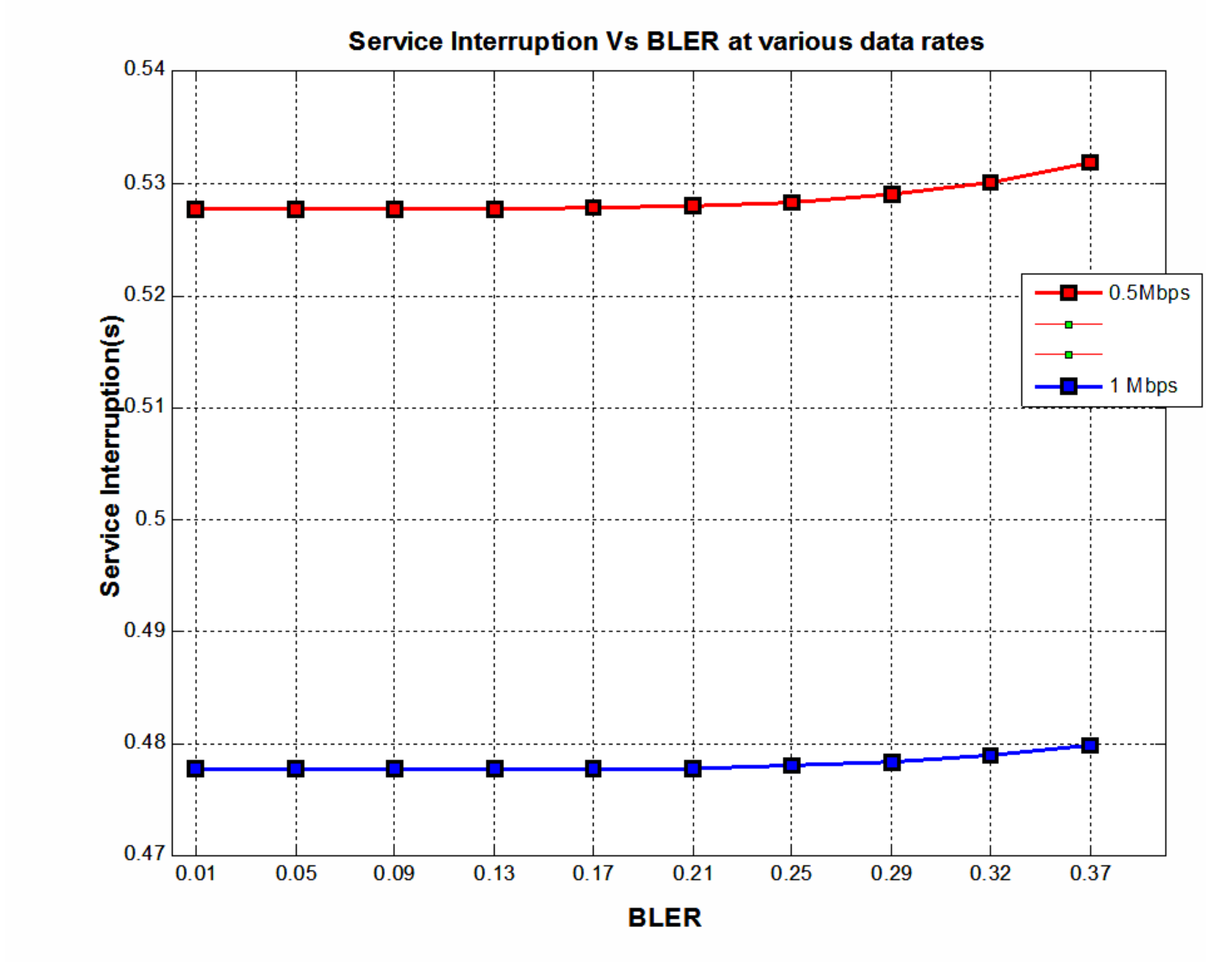

Figure 7.5: UMTS to LTE interruption time vs. BLER for various data rates

In Figure 7.6 the service interruption time was plotted versus the BLER for various orders of propagation delay. Note that all plots were higher than the specified 300ms interruption time. For propagation delay of 10^{-5} and 10^{-6} there was no significant decrease in the service interruption time hence the line plot for 10^{-5} and 10^{-6} propagation delay are superimposed upon each other. This is shown by the red line just above the 470ms mark. As the propagation delay increased from 10^{-3} to 10^{-2}, there was a sharp increase in service interruption time. This behaviour was further illustrated in Figure 7.7. This is similar to the SRVCC scenario in the previous section.

It is therefore apparent that a propagation delay over 10^{-3} adversely affects the service interruption time for both VCC and SRVCC handover.

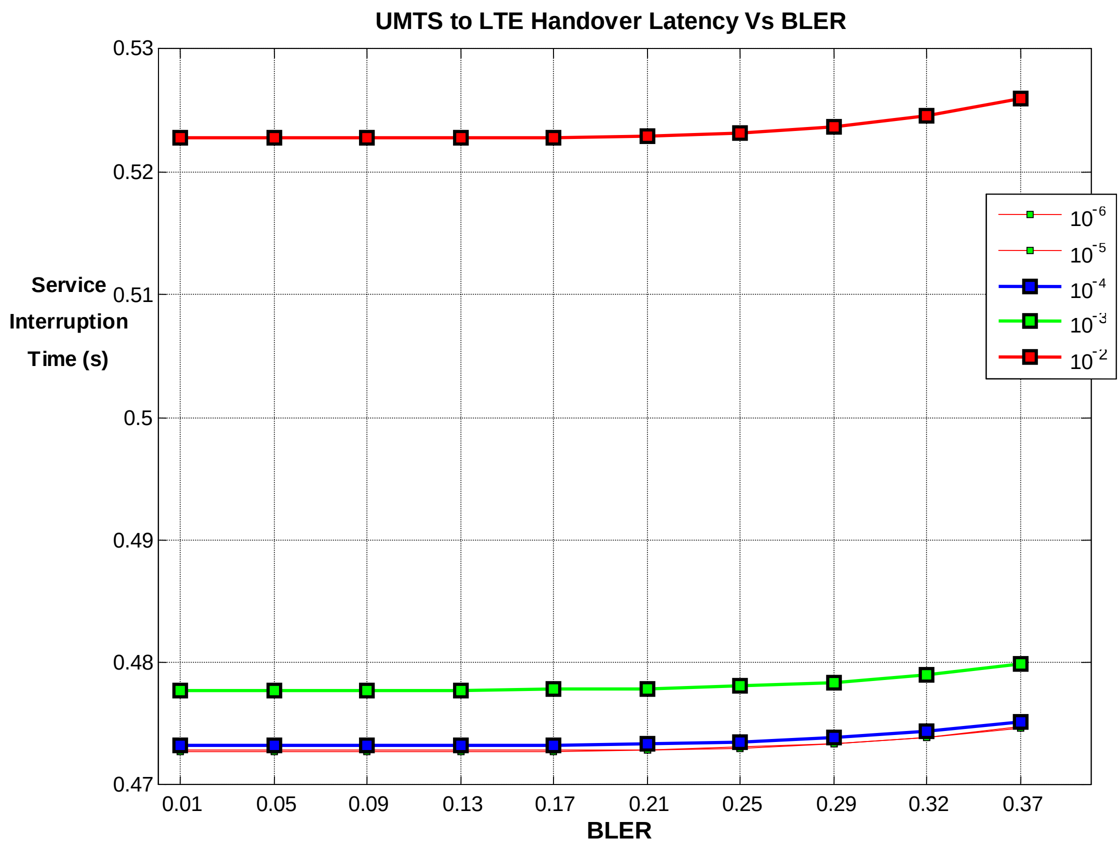

Figure 7.6 : Service Interruption time vs. BLER for UMTS to LTE handover

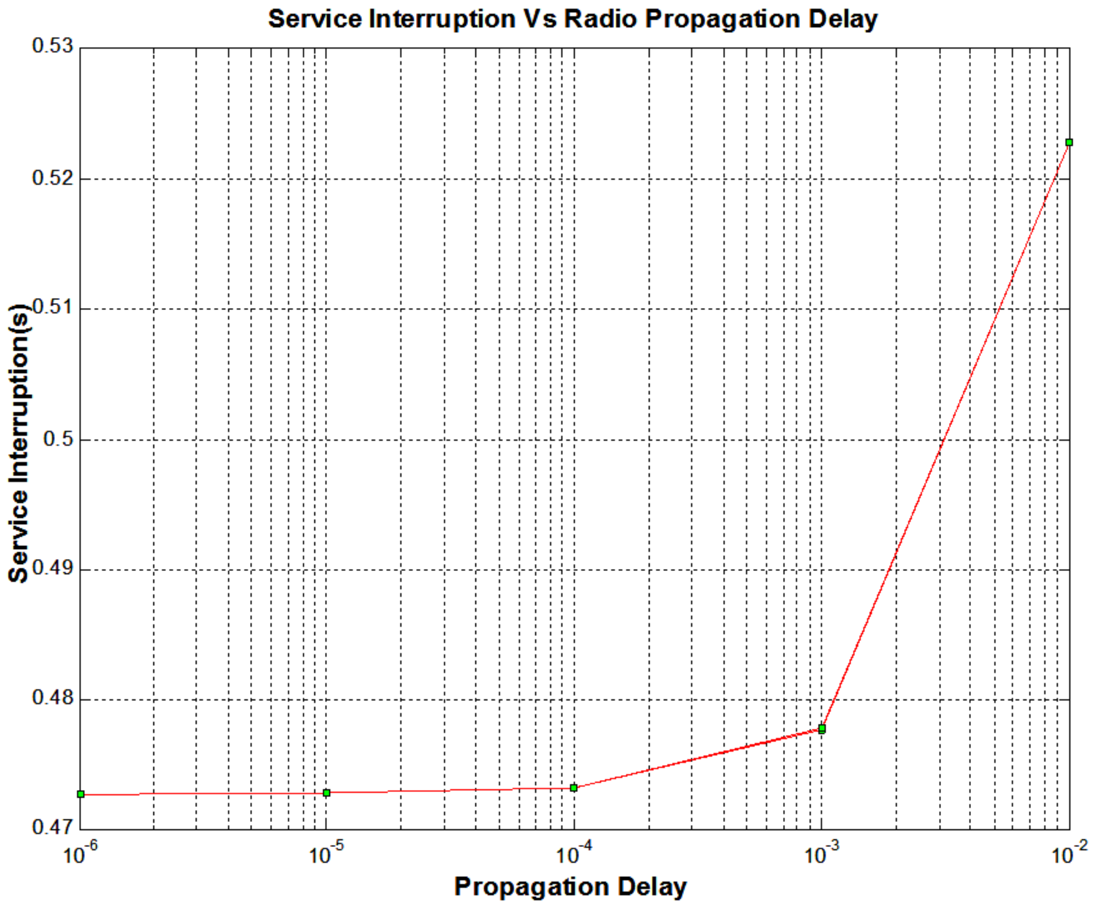

Figure 7.7: Service: Interruption Vs. Propagation delay

The performance for VCC handover from UMTS to LTE was also evaluated under real-time network conditions. Once again the trend in service interruption time versus BLER and propagation delay was found to be similar to that under static conditions. However, after 500 real-time simulations, it was noticed that the service interruption time was generally less than that under static conditions. Figure 7.4 shows the interruption time versus the BLER for different orders of propagation delay under real-time conditions. Note that the interruption time is in the order of two hundred milliseconds (200ms) unlike before where it was in the 400ms range. A real time scenario therefore generates better results than a static scenario.

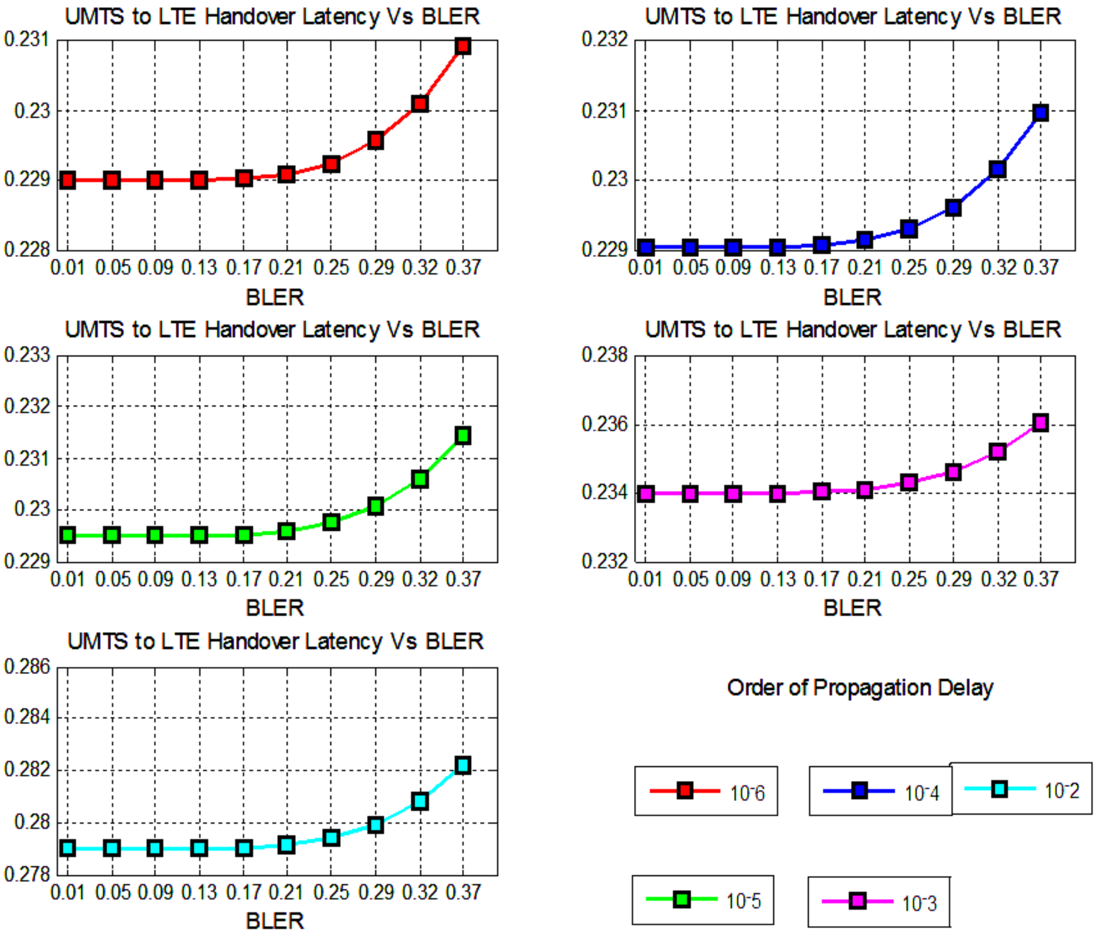

Figure 7.8: VCC Service interruption time under real-time network conditions

8 Conclusions and Recommendations

8.1 Introduction

In the past three decades, cellular networks have evolved from first generation (1G) analogue networks of the early 1980s, to GSM/GPRS in the 1990s and early 2000s to today's high-speed 3G and 4G networks. As this evolution took place, many stakeholders realised that focus should be put on network convergence. In telecommunications, network convergence refers to network architecture designs used to migrate voice and data networks into a single network [41] hence the advent of the Evolved Packet Core (EPC) which comes with the promise of seamless mobility.

With seamless mobility across mobile networks being a major trend, the problem of handover between different RANs must be tackled. This research work focused on evaluating the performance of voice handover between LTE and UMTS. Chapter One of this research report introduced the problem that was undertaken, defined it and articulated the methodology used to tackle the problem. Chapter Two discussed how cellular networks have evolved over the years, from the first generation to the current fourth generation networks.

In Chapter Three the two cellular systems under scrutiny namely UMTS and LTE were discussed. Of particular interest were their physical layer, protocol stacks and handover techniques. A detailed study of voice handover techniques between the two systems was done in Chapter Four. The study compared these handover techniques and it was found that at present, SRVCC and VCC were the most viable network handover techniques for voice handover between LTE and UMTS.

In Chapter Five, we further describe the key research question while in Chapter Six a detailed methodology that was used to carry out the research work was presented. Mathematical models that were used to evaluate SRVCC and VCC performance were discussed. In addition the simulation set up was presented.

Results obtained were presented and discussed in Chapter Seven.

8.2 Key Findings and Conclusions

In the previous chapter, it was shown that under static network conditions, the service interruption time for LTE to UMTS voice handover using the SRVCC scheme was well within the specified 300ms limit for real time services. It was also discovered that a block error rate of over 21% rapidly increased the service interruption time and that under simulated real-time conditions, the SRVCC scheme gave a much improved performance with the service interruption time decreasing by 100ms. A lower BLER translates into a lower service interruption time which directly translates into a better subscriber experience during handover. This is because a high BLER (that is the ratio of erroneous frames to total frames received) results into more retransmissions thus a higher radio propagation delay. As seen in the previous graphs, a high radio propagation delay results into higher service interruption time. In the context of this research, the service interruption time is the time it takes for an inter-RAT voice handover to be executed. That is to say, it is the duration between the issuing of the handover command to its execution. If the service interruption time is high the listener will hear the words and acknowledge the speaker later than a normal conversation and an unnatural rhythm of the conversation may occur.

For UMTS to LTE handover using the VCC scheme under static network conditions, the service interruption time in the range of 400ms was higher than the 300ms threshold. The VCC handover was found to take too long due to the extra time needed to perform IMS anchoring and PDP/bearer setup on the LTE network. These two factors lengthened the overall handover process. Under real-time network conditions, the service interruption time for VCC was seen to drop considerably and was in the range of 200ms. Therefore under real-time conditions the VCC procedure conforms to 3GPP specifications. Because of these two conflicting results, it is possible that VCC may not be able to meet 3GPP specifications. This however can be justified by the fact that LTE base stations will most likely exist as “islands” within ubiquitous UMTS coverage and handover will only be required from LTE to UMTS rather than vice versa. Therefore the VCC procedure will be required less often.

From this discussion, it can therefore be said that SRVCC and VCC are viable handover schemes for operators that intend to deploy LTE alongside UMTS and have an IMS core in existence. SRVCC and VCC are most suitable when compared to other schemes such as VoLGA and CSFB. Because of the following factors;

- They do not require complex/costly handsets. This makes uptake of new LTE services by subscribers quicker as they do not have to buy expensive handsets to enjoy them.
- Although the initial cost of implementing SRVCC and VCC is high due to the requirement of the IMS core, the long-term benefits of stability and high performance compensate for it.
- SRVCC and VCC are standardized and recommended by the 3GPP which is the same body that standardizes LTE and UMTS. They are therefore safe choices since previous inter-RAT handover schemes between 3GPP technologies have worked successfully.
- Above all SRVCC and VCC meet the recommended 3GPP service interruption time of 300ms.

This implies that the inter-RAT voice handover between LTE and UMTS can be successfully carried out without degradation in service. It can therefore be concluded that the objective of this research was met.

8.3 Recommendations

Based on the results presented in Chapter Seven, the following recommendations can be made for an operator wishing to deploy LTE alongside existing UMTS network. To begin with, it is important to note that most of the handover disturbance/delay occurs on the radio interface. This is because network surroundings are random in nature and difficult to control. Disturbances on the radio interface range from buildings, trees, to objects such as billboards and electricity poles. On the other hand, node performance and remote network performance are easier to control and mostly depend on the operator's budget. Given this background, the following recommendations can be made:

- In both LTE to UMTS and UMTS to LTE handover, it is essential to keep BLER on the radio interface at a minimum. An increase in BLER results in higher service interruption time hence degradation of voice service. It is recommended that BLER is kept below 20% in either case.
- Furthermore, regarding the radio interface, it is necessary to keep radio propagation delay at the least. Radio propagation delay is a problematic factor to control as a network operator has no control over how far the mobile terminal is from the transmitter. Regardless, a network operator should optimise the network such that a mobile terminal has a delay in the range of micro seconds (µs). This is because radio propagation delay in the range of ms translates into an increase in overall service interruption time and can adversely affect performance. A propagation delay of 10µs or less is recommended.
- In the case of UMTS to LTE VCC handover, the service interruption time was higher than the specified 300ms for real time services. This was because of the extra time required to set up on the LTE network and perform IMS procedures. Handover in this direction can be optimised by ensuring that registration on the IMS network is kept at a minimum. This may be resolved by increasing the processing speeds of network entities that perform IMS registration and IMS call set up.

8.4 Future work

In this section, suggestions for further research work that may be carried out on the subject of inter-RAT voice handover between LTE and UMTS networks are presented.

Networks that have fully operational LTE, IMS and UMTS with voice interworking between the two systems are not yet available in South Africa. When they finally become available, it will be highly advantageous to collect handover statistics from a live LTE/UMTS network and test voice quality during inter-RAT handovers over a given period of time. This will give deeper insight into the performance of SRVCC and VCC as well as the handover success rate of the handover schemes.

In view of the fact that SRVCC and VCC require the existence of an IMS core, operators without an IMS core may opt for CSFB or VoLGA. It is therefore important to carry out similar performance evaluation for CSFB and VoLGA handover schemes. Perhaps it will emerge that VoLGA and/or CSFB are more superior handover schemes compared to SRVCC and VCC.

REFERENCES

1. **Multi Service Forum.** MSF - We make Next Generations Work. *www.msforum.org.* [Online] MultiService Forum, 2010. [Cited: 15 July 2010.] http://www.msforum.org/about/who.shtml.

2. **Open Mobile Alliance.** About OMA. *www.openmobilealliance.org.* [Online] OMA, 2010. [Cited: 15 July 2010.] http://www.openmobilealliance.org/AboutOMA/Default.aspx.

3. **ETSI.** Telecommunications and Internet converged Services and Protocols for Advanced Networking. *www.etsi.org.* [Online] ETSI, 2003. [Cited: 15 July 2010.] http://www.etsi.org/tispan/.

4. **3GPP.** *UMTS;LTE;IMS.* Sophia Antapolis : ETSI, 2010. TS123338 V.9.4.0.

5. **3GPP**. *Telecommunication management; Performance Management ; Performance measurements.* Sophia Antapolis : ETSI, 2010. TS 32.426 V9.3.0 (2010-12).

6. **Enrique Stevens-Navarro, Ulises Pineda-Rico, Jesus Acosta-Elias.** *Vertical Handover in beyond Third Generation.* 1, San Luis Potosi, S.L.P., MEXICO : Science & Engineering Research Support Center (SERSC), December 2008, International Journal of Future Generation Communication and Networking (IJFGCN), Vol. 1, pp. 51-58. ISSN: 2233-7857.

7. **Harri Holma, Antti Toskla.** *WCDMA for UMTS - HSPA Evolution and LTE.* s.l. : John Wiley & Sons Ltd., 2010. Vol. 5th Edition.

8. **Golmie, N.** *Seamless mobility: are we there yet?* 4, Boulder,USA : IEEE, 2009, Vol. 6. 1536-1284.

9. **Pollhi, Gregory P.** *Trends in Handover Design.* March 1996, IEEE Communications Magazine, pp. 82-90.

10. **Hyun-Ho, Cho Dong-Ho.** *Takeover: a new vertical handover concept for next-generation heterogeneous networks.* **Cho** s.l. : Vehicular Technology Conference, 2005. VTC 2005-Spring. 2005 IEEE 61st, 5 December 2005, IEEE, Vol. 4, p. 2225 . 1550-2252.

11. **Liu Bin, Martins Philippe, Samhat Abed Ellatif, Bertin Philippe.** *A Cross-Layer Scheme for Inter-RAT Handover.* New Orleans, LO : IEEE, 8 December 2008, Global Telecommunications Conference, 2008. IEEE GLOBECOM 2008. IEEE, pp. 1-6. 1930-529X.

12. **Yang Mi-jeong, Cheon Kyung-yul, Park Ae-soon, Kim Sang-ha.** *3GE-WLAN Vertical Handover Experience using Fast Mobile IPv6.* 1738-9445, Gangwon-Do : IEEE, 28 April 2008, Advanced

Communication Technology, 2008. ICACT 2008. 10th International Conference, Vol. 2, p. 1259. 978-89-5519-136-3.

13. **Mohammed A., Kamal H., AbdelWahab, S.** *2G/3G Inter-RAT Handover Performance Analysis.* Edinburgh : s.n., 26 February 2008, Antennas and Propagation, 2007. EuCAP 2007. The Second European Conference on, p. 1. 978-0-86341-842-6.

14. **Lee Seong-Jin, Kim Jae-Hyun, Oh Seong Keun.** *Performance Evaluation of WCDMA-to-CDMA2000 Handover and Roaming.* [ed.] IEEE. Busan : s.n., 11 December 2006, Communications, 2006. APCC '06. Asia-Pacific Conference on , p. 1. 1-4244-0574-2.

15. **Tinnirello Ilenia, Bianchi Giuseppe.** Wireless Networks. *Wireless Networks.* [Online] University of Parlemo, 2005. [Cited: 31 January 2011.] http://www.tti.unipa.it/~ilenia/course/00-intro.pdf.

16. **Pashtan, Ariel.** http://www.awarenetworks.com/publications/6.108.10.CellularTelephony.pdf. *http://www.awarenetworks.com/publications/6.108.10.CellularTelephony.pdf.* [Online] 2006. [Cited: 30 September 2010.] http://www.awarenetworks.com/publications/6.108.10.CellularTelephony.pdf.

17. **GSMA.** Market Data Summary. *GSM World.* [Online] GSM Association, 2009. [Cited: 11 February 2011.] http://www.gsmworld.com/newsroom/market-data/market_data_summary.htm.

18. **Dalman Erik, Parkval Stefan, Skold Johan, Beming Per.** *3G Evolution: HSPA and LTE for Mobile Broadband.* s.l. : Academic Press, Elsevier Ltd., 2008. 978-0-12-374538-5.

19. **3GPP.** *LTE; GPRS Enhancements for EUTRAN Access.* Sophia Antapolis : ETSI, 2010. TS 23.401 version 9.6.0 Release 9.

20. **3GPP**. *Evolved Universal Terrestrial Radio Access (E-UTRA); Physical channels and modulation.* Sophia Antapolis : ETSI, 2010. TS 36.211.

21. **ITU-R.** ITU global standard for international mobile telecommunications ´IMT-Advanced´. *Radio Communication Sector ITU-R.* [Online] ITU, 7 October 2010. [Cited: 5 December 2010.] http://www.itu.int/ITU-R/index.asp?category=information&rlink=imt-advanced&lang=en.

22. **ITU/BDT.** ITU/BDT Arab Regional Workshop on 4G Wireless Systems:. *Evolution path towards 4G.* [Online] ITU/BDT, 29 January 2010. [Cited: 4 February 2011.] http://www.itu.int/ITU-D/arb/COE/2010/4G/Documents/Doc2-LTE%20Workshop_TUN_Session1_Evolution%20path%20towards%204G.pdf.

23. **Shim, Sean.** South Korea launches WiBro Service. *EE Times News and Analysis.* [Online] EE Times, 30 6 2006. [Cited: 4 October 2010.] http://www.eetimes.com/electronics-news/4062077/South-Korea-launches-WiBro-service.

24. **International Telecommunications Union.** ITU World Radiocommunication Seminar highlights future communication technologies. *ITU News Room Press Release.* [Online] ITU, 6 December 2010. [Cited: 21 January 2011.] http://www.itu.int/net/pressoffice/press_releases/2010/48.aspx.

25. **Kaaranen Heikki, Ahtiainen Ari, Laitinen Lauri, Naghian Siamak, Niemi Valtteri.** *UMTS Networks.* s.l. : John Wiley and sons, Ltd, 2001. 0471 48654 X.

26. **3GPP.** 3GPP Scope and Objectives. *www.3gpp.org.* [Online] 31 August 2007. [Cited: 4 February 2011.] http://www.3gpp.org/ftp/Inbox/2008_web_files/3GPP_Scopeando310807.pdf.

27. **3GPP**. About 3GPP: A Global Initiative. [Online] [Cited: 4 Febrauary 2011.] http://www.3gpp.org/About-3GPP.

28. **Motorola Corporation.** The Drivers of LTE. *Motorola.* [Online] Motorola Corporation, 2010. [Cited: 1 February 2011.] http://www.motorola.com/web/Business/Solutions/Industry%20Solutions/Service%20Providers/Wireless%20Operators/LTE/_Document/Static%20Files/The_Drivers_to_LTE_Solution_Paper.pdf?localeId=33.

29. **Sauter, Martin.** Relative Cost of Voice over GSM, UMTS and LTE. *WirelessMoves.* [Online] 18 August 2009. [Cited: 1st February 2011.] http://mobilesociety.typepad.com/mobile_life/2009/08/comparison-of-voice-over-gsm-umts-and-lte.html.

30. **3GPP.** *UMTS User Equipment (UE) radio transmission and reception (FDD, TDD).* Sophia Antapolis : ETSI, 2011. 25.101, 25.102, Release 9.

31. **Li Jun, Montuno Delfin Y., Wang Jianyu, Zhao Yiqiang Q.** Performance Evaluation of the Radio Link Control Protocol in 3G UMTS. [Online] August 2002. [Cited: 03 March 2011.] http://people.scs.carleton.ca/~canccom/Publications/LiPerfEvalRLC04.pdf.

32. **EventHelix.com Inc.** 3GPP LTE Radio Link Control. *www.eventhelix.com.* [Online] 2009. [Cited: 5 November 2010.] http://www.eventhelix.com/lte/presentations/3GPP-LTE-RLC.pdf.

33. **Dinan Esmael, Kurochkin Aleksey, Kettani Sam.** *UMTS Radio Interface System Planning and Optimization.* 1, s.l. : Bechtel Telecommunications Technical Journal, December 2002, Vol. 1.

34. **3GPP.** *Universal Mobile Telecommunications System (UMTS) RRC Protocol Specification.* Sophia Antipolis France : ETSI, 2010. TS 25.331 Version 9.4.0 Release.

35. **Mawjoud S.A.** Simulation of Handoff techniques in Mobile Cellular Networks. [Online] 2006. [Cited: 16 February 2011.] http://alrafidain.engineering-coll-mosul.com/files/174.pdf.

36. **Rezaei S.S.C., Khalaj B.H.** *Grey Prediction Based Handoff Algorithm.* pp. 165-167, 2005, International Journal on Information Technology, Vol. 1.

37. **3GPP.** *UMTS Power control.* Sophia Antapolis : ETSI, 2007. 3GPP TS 25.101, 25.133, 25.214, 25.215, 25.331, 25.433, 25.435, 25.841, 25.849.

38. **3GPP**. *Evolved Universal Terrestrial Radio Access (E-UTRA); Physical layer; Measurements.* Sophia Antapolis : ETSI, 2010. TS 36.214.

39. **Lucidarme, Pierre Lescuyer: Thierry.** *Evolved Packet System: The LTE and SAE Evolution of 3G UMTS.* s.l. : John Wiley and sons, Ltd, 2008. 978-0-470-05976-0.

40. **Poikselka Miika, Mayer Georg, Khartabil Hisham, Niemi Aki.** *The IMS: IP Multimedia Concepts and Services in the Mobile Domain.* s.l. : John Wiley and Sons Ltd, 2004. 0-470-87113-X.

41. **Hanrahan, Hu.** *Network Convergence: Services, Applications, Transport and Operations Support.* s.l. : John Wiley and Sons Ltd, 2007. 978-0-470-02441-6.

42. **Olsson, Magnus.** *SAE and the Evolved Packet Core: Driving the broadband revolution.* s.l. : Oxford Academic Press, 2009. 9780123748263.

43. **3GPP.** *Requirements for E-UTRA and E-UTRAN; 3GPP TR 25.913 version 9.0.0 Release 9.* Sophia Antapolis : ETSI, 2010. TR 125 913.

44. **Kineto Wireless Inc.** VoLGA: Voice over LTE Generic Access. *www.kineto.com.* [Online] March 2009. [Cited: 11 March 2011.] http://www.kineto.com/products/downloads/wp_VoLGA.pdf.

45. **VoLGA Forum.** *Voice over LTE via Generic Access;Stage 2 specification; Phase 1.* s.l. : VoLGA Forum, 2010. V1.7.0.

46. **3GPP.** *Services and Systems Aspects.* Sophia Antapolis : ETSI, 2007. TR 32.820 V8.0.1.

47. **3GPP**. *Circuit Switched Fallback in Evolved Packet System.* Sophia Antapolis : ETSI, 2009. TS 23.272 V8.4.0.

48. **LTEWorld.** CSFB. *LTEWorld.* [Online] 13 January 2011. [Cited: 15 April 2011.] http://lteworld.org/news/huawei-completes-circuit-switched-fallback-voice-call.

49. **Wikispaces, 3GPP.** What is the Idle-mode Signaling Reduction (ISR). *3GPP Wikispaces.* [Online] Wikipedia, 2011. [Cited: 1 April 2011.] http://3gpp.wikispaces.com/What+is+the+Idle-mode+Signaling+Reduction+%28ISR%29%3F.

50. **Kineto Wireless, Inc.** A critical examination of CSFB as a solution for providing Voice-on-LTE. *Disruptive Analysis Ltd.* [Online] December 2009. [Cited: 2011 20 March.] http://www.disruptive-analysis.com/Disruptive%20Analysis%20-%20Limitations%20of%20CS%20Fallback.pdf.

51. **3GPP.** *Single Radio Voice Call Continuity.* Sophia Antapolis : ETSI, 2009. 23.214 V8.6, 2009-12.

52. **3GPP.** *Digital cellular telecommunications system ,UMTS VCC Between CS and IMS systems.* Sophia Antapolis : ETSI, 2008. TS 23.206 version 7.5.0 Release 7.

53. **Huawei technologies Co., Ltd.** Mobile network technology trends. *www.itu.int.* [Online] July 2009. [Cited: 21 January 2011.] http://www.itu.int/dms_pub/itu-t/oth/21/05/T21050000030012PPTE.ppt. GSC14-GRSC-012.

54. **Vittal, Shwetha.** Single Radio Voice Call Continuity (SRVCC) with LTE. *Continuous Computing.* [Online] 2011. [Cited: 15 February 20011.] http://www.ccpu.com/articles/2010/srvcc-single-radio-voice-call-continuity-lte/. MC00267.

55. **Motorola Corporation.** LTE Inter-technology Mobility. *www.motorola.com.* [Online] [Cited: 10 December 2010.] http://www.motorola.com/web/Business/Solutions/Industry%20Solutions/Service%20Providers/Network%20Operators/LTE/_Document/Static%20Files/InterTech%20Mobility%20White%20Paper.pdf.

56. **Itsuma Tanaka, Takashi Koshimizu, Katsutoshi Nishida.** CS FallBack Function for Combined LTE and 3G Circuit Switched Services. *NTT DoCoMo Special Reports.* [Online] 2010. [Cited: 21 January 2011.] www.nttdocomo.co.jp/english/binary/pdf/.../vol11_3_013en.pdf.

57. **Sauter, Martin.** Wireless Moves. *A List of Network Operators Ready for Voicer over LTE with VoLGA.* [Online] 13 January 2011. [Cited: 25 January 2011.] http://mobilesociety.typepad.com/mobile_life/2011/01/a-list-of-network-operators-ready-for-volga.html.

58. **ZTE Technologies.** VoLTE Takes Off at Mobile Asia Congress 2010. *ZTE.* [Online] ZTE, 17 January 2011. [Cited: 15 February 2011.]

59. **Kim, Kyungmin.** *A seamless voice call handover scheme for next generation cellular network.* Shanghai : IEEE, 2009. IEEE 15th Asia-Pacific Conference on Communications, 2009. APCC 2009. . pp. 782-785. 978-1-4244-4784-8.

60. **3GPP.** *UMTS, RLC Protocol Specification;.* Sophia Antapolis : ETSI, 2010. TS 25.322 version 9.2.0 Release 9.

61. **Das Sajal K., Lee Enoch, Basu Kalyan, Sen Sanjoy K.** *Performance Optimization of VoIP Calls over Wireless Links Using H.323 Protocol.* s.l. : IEEE Computer Society, 2003. IEEE Transactions on Computers. Vol. 52, pp. 742-752. 0018-9340/03.

62. **Enderle N., Lagrange X.** *Radio link control-acknowledged mode protocol performance modeling in UMTS.* Velizy : IEEE, 2002. 4th International Workshop on Mobile and Wireless Communications Network. pp. 332 - 336. 0-7803-7605-6.

63. **Li Jun, Montuno Delfin Y., Wang Jianyu, Zhao Yiqiang Q.** Performance Evaluation of the Radio Link Control Protocol in 3G UMTS. *citeseerx.ist.psu.edu.* [Online] 2006. [Cited: 10 Febraury 2011.] http://citeseerx.ist.psu.edu/viewdoc/download?doi=10.1.1.118.2349&rep=rep1&type=pdf.

64. **Junfeng Jiang, Cao Zhigang.** *An AM-RLC scheme with adaptive acknowledgement interval.* 2, Shanghai : Higher Education Press and Springer-Verlag 2009, 2009, Frontiers of Electrical and Electronic Engineering in China, Vol. 4.

65. **3GPP.** *LTE; EUTRA; Physical channels and modulation.* Sophia Antapolis : ETSI, 2011. TS 136 211.

66. **International Telecommunications Union.** Teletraffic Engineering Handbook. *International Telecommunications Union.* [Online] 20 June 2001. [Cited: 21 February 2011.] https://www.itu.int/ITU-D/study_groups/SGP_1998-2002/SG2/StudyQuestions/Question_16/RapporteursGroupDocs/teletraffic.pdf. ITU{D SG 2/16 & ITC.

67. **Necker, Marc C.** A simple model for the IP packet service time in UMTS networks. *Institute of Communication Networks and Computer Engineering (IKR).* [Online] [Cited: 20 Febraury 2011.] http://www.ikr.uni-stuttgart.de/Content/Publications/Archive/Ne_ITC05_36418.pdf. Ne_ITC05_36418.

68. **Gregory, Elizabeth.** The M/G/1 Queue. *cnx.org.* [Online] cnx.org, 9 June 2005. [Cited: 4 April 2011.] http://cnx.org/content/m10819/latest/.

69. **ITU-R, Recommendation ITU-R M.1225.** *Guidelines for Evalution of Radio Transmission Technologies for IMT-2000.* Febuary 1997.

70. **3GPP.** *Radio Network Temporary Identities.* Sophia Antapolis : ETSI, 2010. TS 25.401.