COMPARISON OF THE OVERAL PERMORMANCE OF VOICE HANDOVER SCHEMES BETWEEN UMTS AND LTE USED BY NETWORK OPERATORS IN ZIMBABWE

BY

TATENDA AGRIPA KWAVA

R113701B

Submitted in partial fulfilment of the requirement for the degree of

BSC TELECOMMUNICATIONS HONOURS DEGREE

DEPARTMENT OF APPLIED PHYSICS AND TELCOMMUNICATIONS IN THE FACULTY OF SCIENCE AND TECHNOLOGY

Midlands State University

Gweru

MAY 2015

SUPERVISORS MR NECHIBVUTE AND MR MAZUNGA

ABSTRACT

The main objective of mobile operators 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 in Zimbabwe faced 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.

DECLARATION

I TATENDA AGRIPA KWAVA hereby declare that I am the sole author of this thesis. I authorize the Midlands State University to lend this thesis to other institutions or individuals for the purpose of scholarly research.

Signature ____________________________________ Date _______________________________

This dissertation is entitled COMPARING OF THE OVERAL PERROMANCE OF VOICE HANDOVER SCHEMES BETWEEN UMTS AND LTE BY MOBILE COMPANIES IN ZIMBABWE by Tatenda Kwava meets the regulations governing the award of the degree of BSC TELECOMMUNICATIONS HONOURS of the Midlands State University, and is approved for its contribution to knowledge and literal presentation.

Supervisor …………………………………………………...

Date ………………………………………………………….

Acknowledgement

Firstly I would like to thank the Almighty God for the gift of life and protecting me while I completed my dream of having a Telecommunications degree.

I wish to express my sincere gratitude to my supervisors Mr Nechibvute and Mr Mazunga for their guidance and support. Thank you for being patient with many of my shortcomings and always being willing to help.

I also wish to convey my sincere gratitude to Econet Wireless for their academic support and their ever increasing love to help the department.

I would also want to extend my sincere gratitude to the department of Science and Technology and members of staff for introducing this program and their love and guidance

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.

Abbreviations

3GPP	Third Generation Partnership Project
POTRAZ	Postal and Regulatory Authority of Zimbabwe
SRVCC	Single Radio Voice Call Continuity
CSFB	Circuit Switched Fall Back
IP	Internet Protocol
IMS	IP Multimedia System
SHO	Soft handover
LTE	Long Term Evolution
RAB	Radio Access Bearer
RAN	Radio Access Network
RLC	Radio Link Controller
UMTS	Universal Mobile Telecommunications System
VoLTE	Voice over LTE
WCDMA	Wide Band Code Division Multiple Access
SRNCC	Serving Radio Network Controller
PS	Packet Switched
P-CCPCH	Primary Common Control Physical Channel
1G	First Generation
2G	Second Generation
3G	Third Generation
4G	Fourth Generation

TABLE OF CONTENTS

Contents

LIST OF TABLES

LIST OF FIGURES

CHAPTER 1

INTRODUCTION

1.1 Background of the study

The voice handover mechanism is extremely important in cellular network because of the cellular architecture employed to maximise spectrum utilization. Handover in general is the procedure that transfers an ongoing call from one cell to another as the users move through the coverage area of cellular systems. One way to improve the cellular network performance is to use efficient handover prioritization schemes. There are different handover schemes used in 3G and LTE technologies of which each has its advantages and disadvantages. The research will mainly focus on the ones used by Zimbabwe mobile operators which includes Voice Call Continuity (VCC), Soft handover, Single Radio Voice Call Continuity and Circuit Switched Fall Back. The research is based on comparing voice handover schemes between Long Term Evolution (LTE) and Universal Mobile Telecommunication System (UMTS) networks in Zimbabwe with regard to 3rd Generation Partnership Project (3GPP) specification.

1.2 Research Problem

1.2.1 Problem Statement

The increase in tariffs by mobile companies in Zimbabwe is mainly due to many factors, one which include the installation of many base stations by each mobile company instead of infrastructure sharing which costs a lot of revenue. The research is about comparing the voice handover schemes used by the operators which will be suitable for UMTS and LTE. Possible results will come up with the suitable handover schemes which are not costly and will reduce the tariffs. The suitable handover scheme will have to benefit both the operator and the subscribers.

1.2.2 Advancement of the Problem Statement

Combining both data and voice services on the same LTE or UMTS data access network enables mobile operators to optimise network and service management, integrate network resources and simplify service delivery; this results in a significant reduction of operating expense. This will make their operations easier and less expensive to manage. Operators will be able to pack more information into packets that go from consumer phones to operator cell tower. The emergence of smart devices (such as smartphones and tablets), the line between who provides value to the subscriber and who they pay has blurred. Operators are at greater risk of becoming bit transporters, while content/application providers and device manufacturers capture more of the revenue from mobile subscribers. Policy management is one method operators can implement to form new business models and maximize the service monetization. The student will use Econet Wireless (Pvt) (Ltd) as his case study since it is regarded as the bench mark of the network operators in Zimbabwe. The growing demand for mobile broadband networks in emerging markets is fuelled by the fact that usually it is the only feasible means for providing first-time broadband connectivity for majority of users and to stay connected while roaming in heterogeneous networks.

1.3 Justification

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.

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. Which involves a user terminal being able to seamlessly move from one radio access network to another without discontinuity in service.

1.4 Aims

The aim of this research is to compare handover schemes for voice services between UMTS and LTE networks used by network operators in Zimbabwe will. With Voice over LTE not yet a realistic solution yet to be used by Econet Wireless due to the slow uptake of Internet Protocol Multimedia Network Subsystem (IMS-this is an architectural framework delivering IP multimedia services), since voice is still a major source of their revenue, they need to maintain and offer same quality of service that their subscribers have become accustomed to. By comparing the schemes, one has to know the parameters to measure for network performance. One have to know the four categories in which the network is measured: accessibility, retainability, integrity and mobility.

Table 1.1 below shows the key performance indicators that will be used to measure the network performance

Table 1.1 Key Performance Indicators Parameters

CATEGORY	DESCRIPTION
Accessibility	How easy it is for the user to obtain a service within specified tolerances and other given conditions. Session Setup Success Rate is a common KPI in this category.
Retainability	The capability of a service, once obtained, to continue to be provided under given conditions for a requested period. Examples of KPIs in this category include Session Abnormal Release Rate (dropped calls) and Minutes Per Abnormal Session Release.
Integrity	The degree to which a service, once obtained, is provided without excessive impairments. Examples include downlink (DL) and uplink (UL) throughput, latency and packet loss.
Mobility	Performance of all handover types. Examples include LTE Handover Success Rate and Inter-Radio Access Technology (IRAT) Handover Success Rate.

1.5 Research Objectives

The research is to answer the following questions about voice handover schemes:

1) For the schemes employed by the operator, is the operator meeting the targeted Key Performance Indicators (KPI). (A set of quantifiable measures that a company or industry uses to gauge or compare performance in terms of meeting their strategic and operational goals.) [5]
2) How long have been these schemes been in use?
3) Is the operator satisfied by the current handover performance statistics?
4) Is the handover scheme standardized by a major telecommunications standards organisation?
5) Is the handover scheme more viable compared to other schemes?
6) To research on the potential benefits and challenges that can be realised with LTE handover techniques in Zimbabwe

1.6 Dissertation Outline

Chapter 2: The literature review

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. Also to look more into the physical structure of these generations.

Chapter 3: METHODOLOGY

A clear description of the data capturing method shall be provided in this section. This section will highlight how the researcher will perform the drive drive tests, also how they will capture the data. Other. It will describe in detail the procedures involved in data collection. It introduces the different voice handover techniques used for inter-RAT handover between LTE and UMTS used by Econet Wireless. It discusses each technique, its network architecture and operation and finally compares the techniques to each

Chapter 4: Results and Discussion.

This section will seek to give the answers to the questions or hypotheses brought forward in chapter 3. Illustrative factual analysis will be indicated in this area. This will incorporate diagrams and tables where suitable. An outline of the outcomes might be given and talked about in point of interest. Hence an integration of the results with the hypotheses and literature review.

Chapter 5: Summary and Conclusions.

The conclusion will report the outcomes in short and unambiguously examine the impacts of the outcomes' recommendations. Proposals for future examination will be drawn up and why the proposed exploration is required and in what direction should the research follow.

1.7 Assumptions

The tests will be performed for both busy hours during the day and in low network load periods during the night

CHAPTER 2

Literature Review

2.1 Introduction

This chapter gives a brief discussion of the evolution process of network technology, discussions of the types of handover schemes that will be compared in this research and optimisation process. These handovers are Voice Call Continuity, Single Radio Voice Call Continuity, Circuit Switched Fall-Back, and soft handover which are used by Econet Wireless. 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 3rdGeneration Partnership Project (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. Also this chapter will

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 [1].

Use of cellular system in 1G or First generation of wireless telecommunication technology resulted in great spectrum usage. The First generation of wireless telecommunication technology used analog transmission techniques which were basically used for transmitting voice signals. 1G or first generation of wireless telecommunication technology also consist of various standards among which most popular were Advance Mobile Phone Service (AMPS), Nordic Mobile Telephone (NMT), Total Access Communication System (TACS). All of the standards in 1G use frequency modulation techniques for voice signals and all the handover decisions were taken at the Base Stations (BS). The spectrum within cell was divided into number of channels and every call is allotted a dedicated pair of channels. Data transmission between the wire part of connection and PSTN (Packet Switched Telephone Network) was done using packet-switched network.[1]. Analog Signals does not allow advance encryption methods hence there is no security of data i.e. anybody could listen to the conversion easily by simple techniques. The user identification number could be stolen easily and which could be used to make any call and the user whose identification number was stolen had to pay the call charges. 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.[2]

2.3 Second Generation (2G)

Second Generation (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. 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 Short Messaging Service (SMS) text messages.[3]. 2G phone systems were run through digital circuit switched transmission. The 2G digital cellular networks expanded on the voice-only services of 1G networks, enabling a variety of new features such as push-to-talk, short messaging service (SMS), conference calling, caller ID, voicemail and simple data applications like email messaging and Web browsing. These networks are still in existence today, providing voice service to the majority of today’s cell phone users [4]. The most dominant technology in 2G system is Global System for Mobile Communication (GSM). The next sector revises the principle of GSM, the principle of GSM, its architecture and also its operation.

2.3.1 Principle of Operation of Global System for Mobile Communication (GSM)

Global System for Mobile communication (GSM) is a digital mobile telephony system that is widely used or a globally accepted standard for digital cellular communication .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 [5]. 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[6].GSM was devised as a cellular system specific to the 900 MHz band, called The Primary Band. The primary band includes two sub bands of 25 MHz each, 890 to 915 MHz and 935 MHz to 960 MHz GSM-PLMN has allocated 124 duplex carrier frequencies over the following bands of operation [7].

2.3.2 GSM Architecture

The GSM network can be divided into three broad parts, the subscriber that carries the mobile station, the base station subsystem which controls the radio link with the mobile station, the network subsystem which performs the switching of calls between the mobile users and other mobile and fixed network users. Fig 2.1 show the GSM architecture

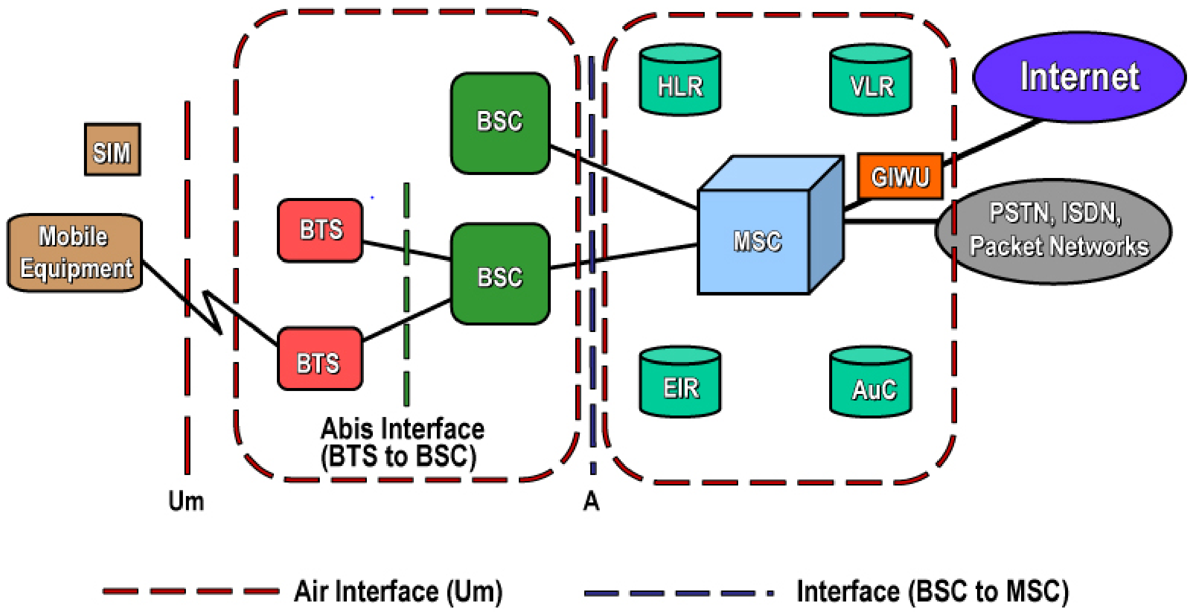

Fig 2.1 Simplified GSM Network overview [7]

(a) Mobile Station

The mobile station consists of the mobile equipment, i.e. the handset, and a smart card called the Subscriber Identity Module (SIM). The SIM provides personal mobility, so that the user can have access to subscribed services irrespective of a specific terminal. By inserting the SIM card into another GSM terminal, the user is able to receive and make calls from that terminal, and receive other subscribed services [8]. The mobile equipment is uniquely identified by the International Mobile Equipment Identity (IMEI). The SIM card contains the International Mobile Subscriber Identity (IMSI) used to identify the subscriber to the system, a secret key for authentication and other information. The IMEI and the IMSI are independent, thereby allowing personal mobility. The SIM card may be protected against unauthorised use by a password or personal identity number

(b) Base Station Subsystem

The base station subsystem is composed of two parts, the base transceiver station and the base station controller. These communicate across a standardised "Abis" interface, allowing operation between components made by different suppliers. The base transceiver station houses the radio transceivers that define a cell and handles the radio-link protocols with the mobile station. In a large urban area, there will potentially be a large number of base transceiver stations deployed, thus the requirements for a base transceiver station are ruggedness, reliability, portability and minimum cost. The base station controller manages the radio resources for one or more base transceiver stations. It is the connection between the mobile station and the mobile services switching centre [8].

(c) Network Subsystem

The central component of the network subsystem is the mobile services switching centre. This acts like a normal switching node of the Public Switched Telephone Network (PSTN) Integrated Services Digital Network (ISDN) and connects the mobile signal to these fixed networks. It additionally provides all the functionality needed to handle a mobile subscriber, such as registration, authentication, location updating, handovers and call routing to a roaming subscriber [8].

2.4 Third Generation (3G)

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 [9]. Technically speaking 3G is a network protocol which refers to the generations of mobile phones and telecommunication equipment which are compatible with the International Mobile Telecommunications-2000 (IMT-2000) standards stated by International Telecommunication Union (ITU). The basic requirement for compiling to IMT-2000 standards is that the technology should provide peak data rates of at least 200 kbit/s. It’s worth mentioning that speed isn’t the only criteria for deciding whether the network protocol is 3G or not. 3G isn’t just any high speed network but a protocol which has its own standards defined under IMT-2000 by ITU [10].

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)[11].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. The reason behind the 4G service offering is to deliver a comprehensive IP based solution where multimedia applications and services can be delivered to the user anytime and anywhere with a high data rate, premium quality of service and high security. Seamless mobility and interoperability with existing wireless standards is crucial to the functionality of 4G communications. Implementations will involve new technologies such as femtocell and picocell, which will address the needs of mobile users wherever they are and will free up network resources for roaming users or those in more remote service areas[12].

2.5.1 Summary of evolution of 3G (UMTS)-4G (LTE)

Table 2.1 Evolution of 3G-4G

	WCDMA (UMTS)	HSPA HSDPA / HSUPA	HSPA+	LTE
Max downlink speed bps	384 k	14 M	28 M	100M
Max uplink speed bps	128 k	5.7 M	11 M	50 M
Latency round trip time approx	150 ms	100 ms	50ms (max)	~10 ms
3GPP releases	Rel 99/4	Rel 5 / 6	Rel 7	Rel 8
Approx years of initial roll out	2003 / 4	2005 / 6 HSDPA 2007 / 8 HSUPA	2008 / 9	2009 / 10
Access methodology	CDMA	CDMA	CDMA	OFDMA / SC-FDMA

2.5.2 Summary of evolution from 1G-4G

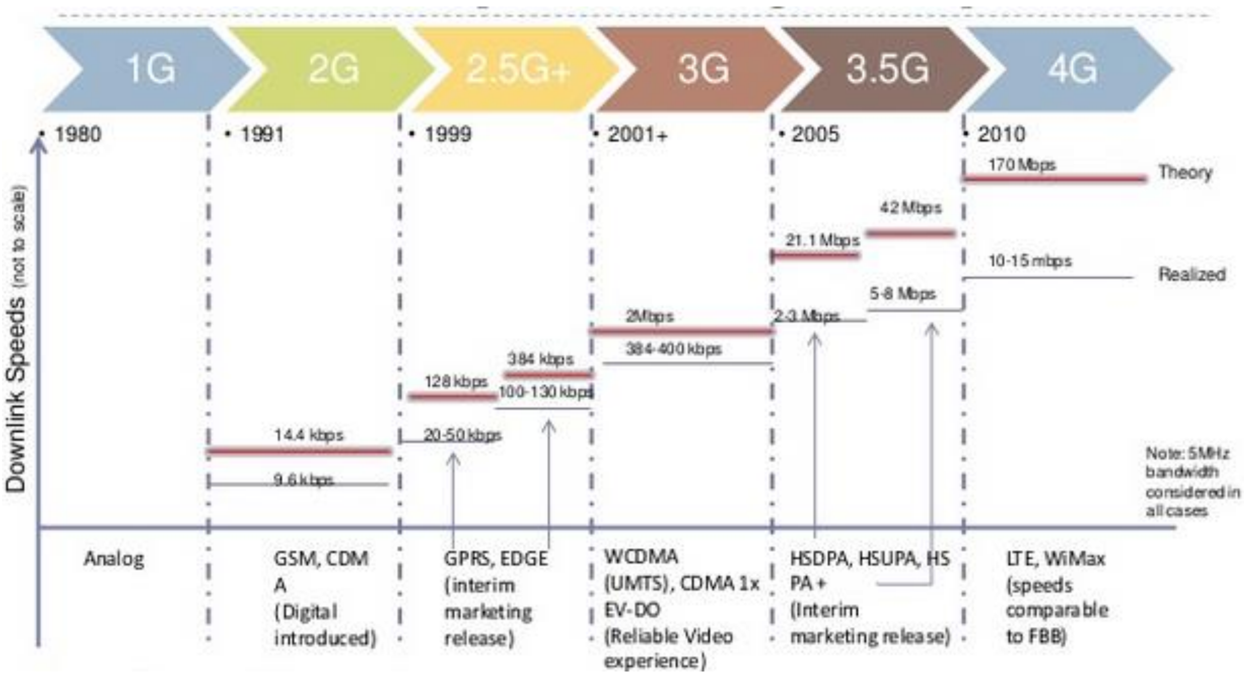

Fig 2.2 Summary of evolution from 1G-4G [8]

2.6 Universal Mobile Telecommunications System (UMTS)

UMTS network architecture consists of three domains which are Core Network (CN), UMTS Terrestrial Radio Access Network (UTRAN) and User Equipment (UE) which are shown in fig 2.2

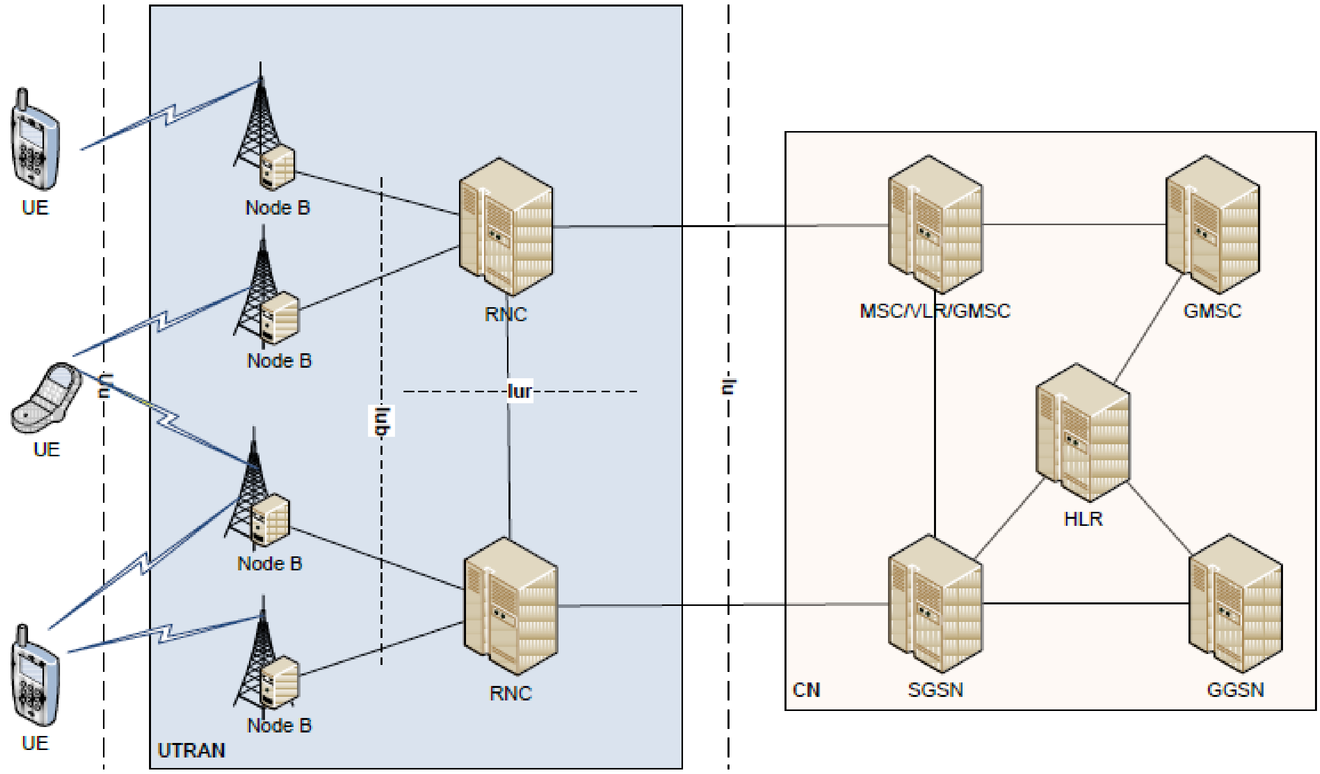

Fig 2.3 UMTS ARCHITECTURE [6]

The UE: Consists of the Mobile Equipment (ME) and Universal Subscriber Identity Module (USIM).The ME is a radio terminal used for radio communication .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.

Gateway GPRS Support Node (GGSN) is similar to the GMSC but in relation to packet switched services.

Gateway MSC (GMSC) is the switch through which the UMTS connects to other CS networks.

2.7 Long Term Evolution

Long Term Evolution is the Generation of mobile broadband technology with promised data transfer rates of 100 Mbps. Also based on UMTS (3G) technology. LTE is entirely a packet switched system and voice will be provided over IP which is Voice over IP (VoIP).

2.7.1 LTE Handover events

Below is a list of LTE handover events which are standardised by the 3rd Generation Partnership Project (3GPP) which are used internationally as the standards

Table 2.2 LTE Handover Events [14]

Event Type	Description
Event A1	Serving becomes better than threshold
Event A2	Serving becomes worse than threshold
Event A3	Neighbour becomes offset than serving
Event A4	Neighbour becomes better than threshold
Event A5	Serving becomes worse than threshold 1 and neighbour becomes better than threshold 2
Event B1	Inter RAT neighbour becomes better than threshold
Event B2	Serving becomes worse than threshold 1 and inter RAT neighbour becomes better than threshold 2

2.7.2 Advantages of LTE

- For network operators

- It lowers the total cost of ownership of the network
- It reduces power consumption and footprint, delivering a greener sustainable solution
- It offers an order of magnitude increase in capacity and flexibility to manage growth
- High network throughput and low latency
- LTE is all IP which is based on IPv6 which supports massive numbers of additional IP addresses and provides other improvements over IPv4

For the Users

- Low mobile wireless latency
- Improved performance
- Highly reliable

2.7.3 LTE Architecture

Mobility Management Entity (MME) this deals with control plane signalling, mobility management and idle-mode. The Serving Gateway (S-GW) mainly handles mobility management within the LTE network and other 3GPP technologies. Fig 2.4 shows the LTE architecture.

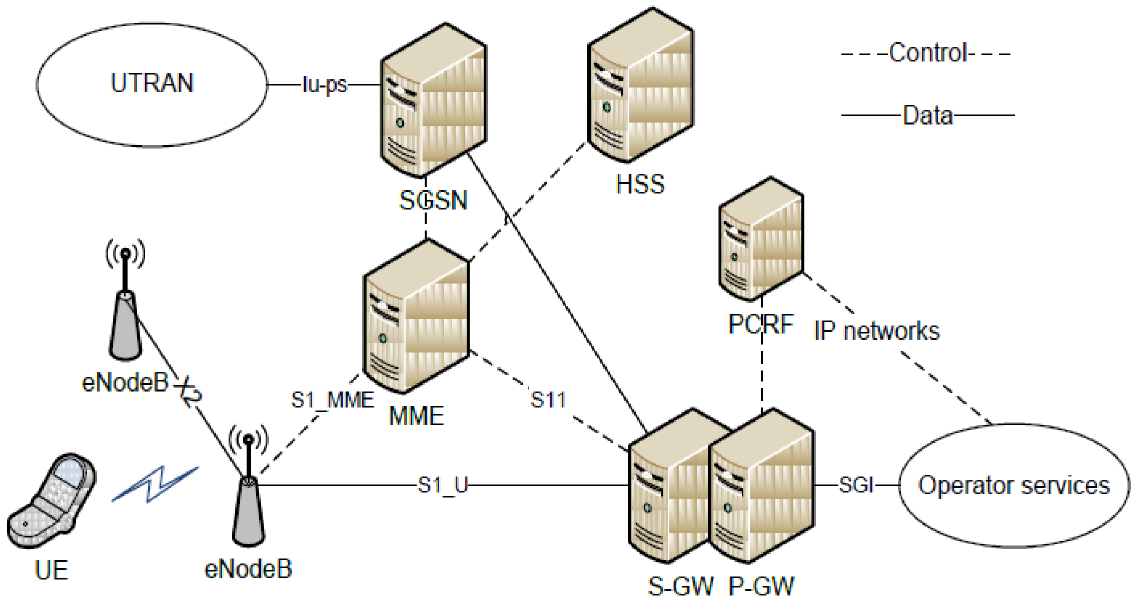

Fig 2.4 LTE Architecture [9]

2.8 Handover process

Handover measurement is mainly good for two reasons which are 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 [13]. Fig 2.5 shows the handover process

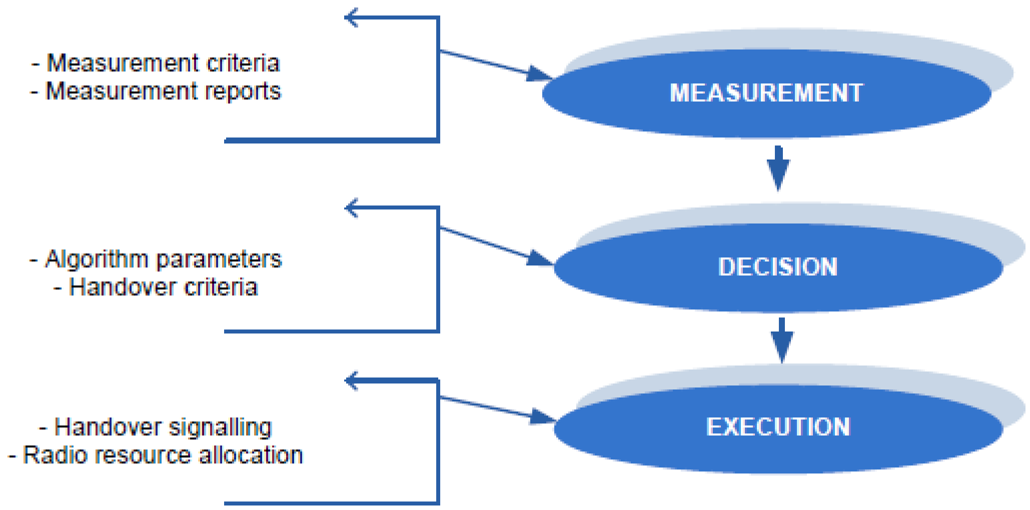

Fig 2.5 Handover process [13]

The measurement events may be triggered based on the following criteria

- 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

For a handover decision making, there are two types of them which are Network Evaluated handover (NEHO) and Mobile Evaluated handover (MOHO). For a NEHO decision the network Serving Radio Network Controller (SRNC) makes the handover decision while with the MEHO approach, the UE prepares the handover decision. These handover decision makings depends on the measurement from the UE and BS, also the handover algorithm criteria [14].

2.8.1 Handover Algorithm

The User Equipment (UE) constantly measures signal strength of neighbouring cells and reports to the RNC based on the strength of the downlink physical channels for signals with same frequencies, signals with different frequencies and belonging to another radio access system other than UTRAN. It also considers traffic volume measurements which contain measurements for uplink traffic volume, quality measurements include quality parameters e.g. downlink transport block error rate and internal measurements of UE transmitted power and received signal level [15].Fig 2.6 below shows the handover algorithm

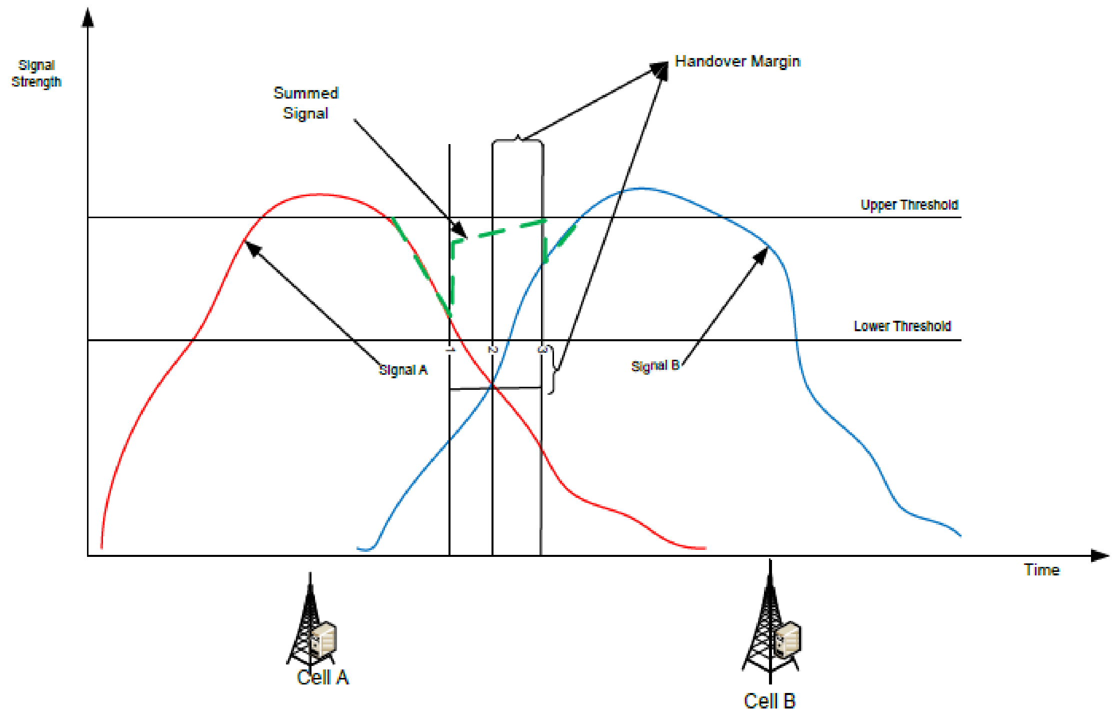

Fig 2.6 Handover Algorithm [15]

If UE camping on cell A moving towards cell B, as UE moves towards cell B, the pilot signal (A) deteriorates, approaching the lower threshold thereby handing over to cell B. If the signal of B becomes better than A, the RNC starts calculating the handover margin calculation.

Fig 2.7 shows the overview of the handover process.

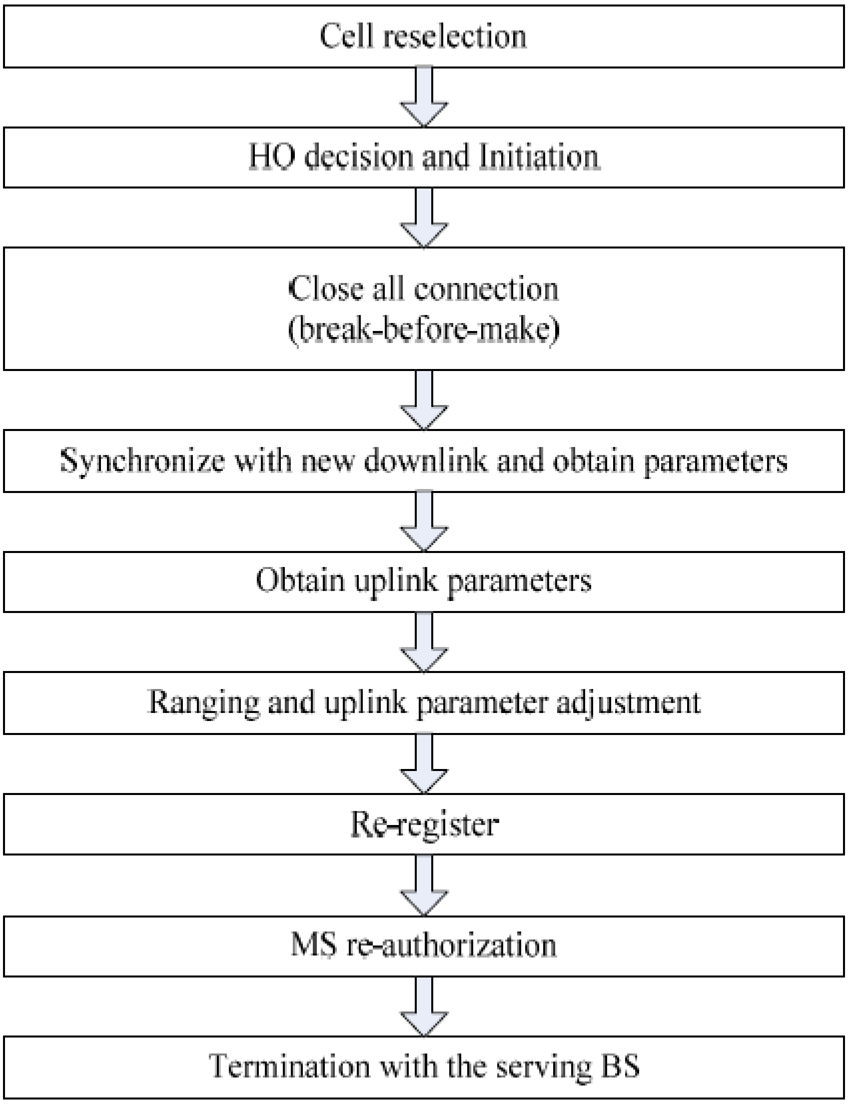

Fig 2.7 Overview of Handover Process

2.9 Circuit Switched Fall Back (CSFB)

This is a type of handover that enables circuit-switched voice and SMS services to be delivered to LTE devices. When an LTE handset makes or receives voice calls, the device "falls back" to the 3G (UMTS) network to complete the call. This type of handover is needed because LTE is an all-IP, packet-based network that cannot support circuit-switched calls [16]. CSFB has got a capable terminal being served by E-UTRAN which falls back onto the circuit switched domain whenever it makes or receives a voice call.

2.9.1 CSFB Architecture

From fig 2.7 the SG interface is the reference point between the MME and MSC server and the interface is used for the mobility management and paging procedures between EPS and CS domain. It is also based on the Gs interface procedures. The SGs reference point is also used for the delivery of both mobile originating and mobile terminating SMS. The architecture of CSFB is shown in Fig 2.8 below.

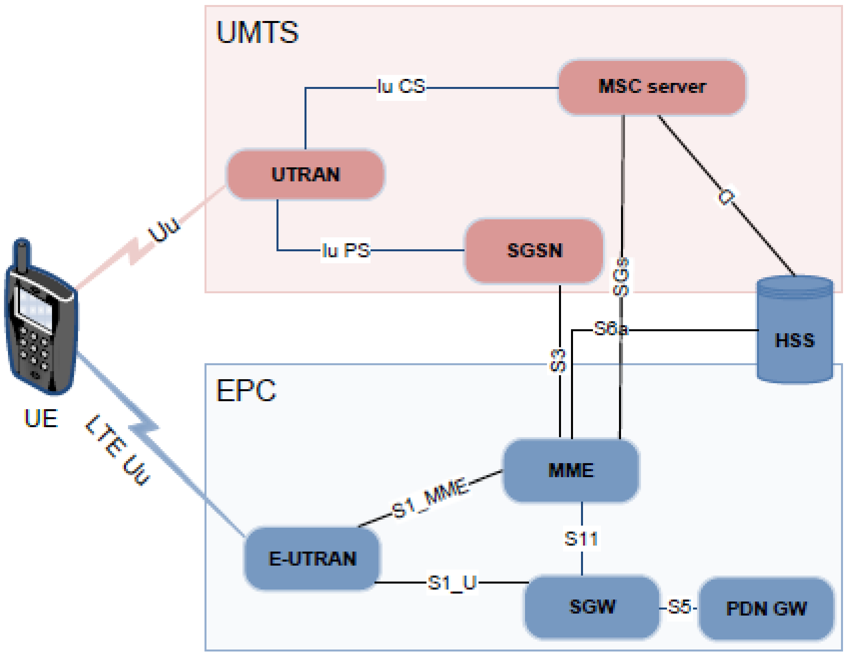

Fig 2.8 CSFB Architecture [17]

.

a) Mobile Management Entity (MME)

This is the key control-node for the LTE access-network which is responsible for idle mode UE (User Equipment) paging and tagging procedure including retransmissions. ALSO involved in the bearer activation/deactivation process and is responsible for choosing the SGW for a UE at the initial attach and at time of intra-LTE handover involving Core Network (CN) node relocation. MME is also responsible for responsible for authenticating the user by interacting with the HSS

b) Mobile Switching Centre (MSC)

It is used for maintaining SGs association towards MME for EPS/IMSI attached UE and support of SMS procedures as provided in 3GPP specification

c) Evolved UMTS Terrestrial Radio Access (E-UTRAN)

IT is used for forwarding paging request and SMS to the UE and directing the UE to the target CS capable cell.

2.9.2 Circuit Switched Fall-Back Operation

CSFB takes place whenever a mobile terminal receives or makes a voice call in LTE network. Considering a mobile terminal camping on the E-UTRAN as illustrated step 1, a mobile terminated voice call arrives at the terminal via the SGs interface from the CS network. Then the UE recognises that the call is from the CS domain given the information contained in VLR/MSC address. Below is the diagram of the operation of CSFB. The overview of the operation of CSFB is shown in Fig 2.9

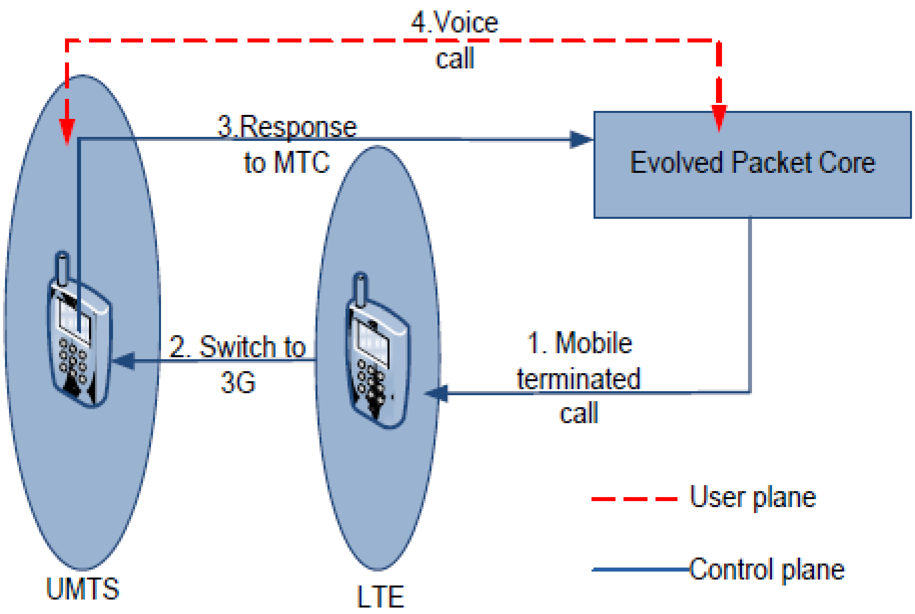

Fig 2.9 CSFB Operation [17]

The EPC then communicates with UMTS, UMTS prepares network resources for the new call and the EPC instructs the mobile to switch to UTRAN, the mobile moves to UTRAN and the voice call proceeds [17]

2.9.3 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 and initiates a call. The signal flow of a subscriber moving from LTE-UMTS is shown 2.5

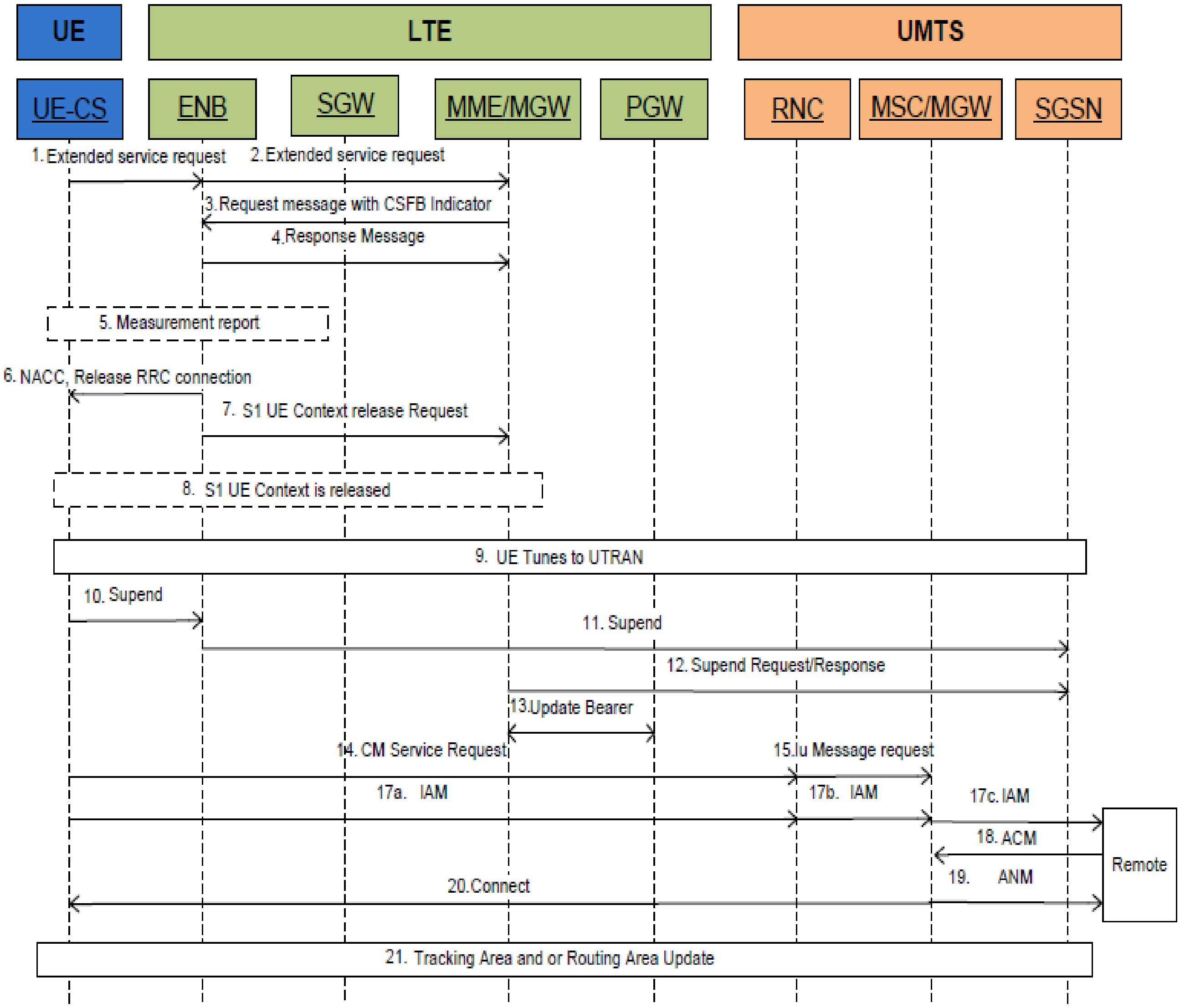

Fig 2.10 Signal flow for a Mobile Originated Call in CSFB.

2.8 Single Radio Voice Call Continuity (SRVCC)

SRVCC, Single radio Voice Call Continuity, is a technique that enables Inter Radio Access Technology, Inter RAT handover as well as a handover from packet data to circuit switched data voice calls [18].It is also a solution for circuit switched voice handover between UMTS and LTE. Single Radio Voice Call Continuity (SRVCC) in Release 8 specifications is to provide seamless continuity when an UE handovers from LTE coverage (E-UTRAN) to UMTS coverage (UTRAN). With the introduction of SRVCC, calls are anchored in IP Multimedia Subsystem (IMS) network while UE is capable of transmitting/receiving on only one of those access networks at a given time. Also with SRVCC, a call anchored in IMS core will continue in to be served by UMTS and GSM networks outside of LTE coverage area[19].The SRVCC works by centring mobile and broadband wireless access technologies as it offers LTE-IMS based voice service within the LTE coverage area, and also CS-based voice service outside the LTE coverage area.

2.10.1 SRVCC Architecture

The Sv is an interface between the Mobility Management Entity (MME) or Serving GPRS Support Node (SGSN) and 3GPP MSC server enhanced for SRVCC which is shown in fig 2.11. It is used to support Inter-RAT handover from VoIP/IMS over EPS to CS domain over 3GPP UTRAN access and to transfer messages between the MME and MSC Server.

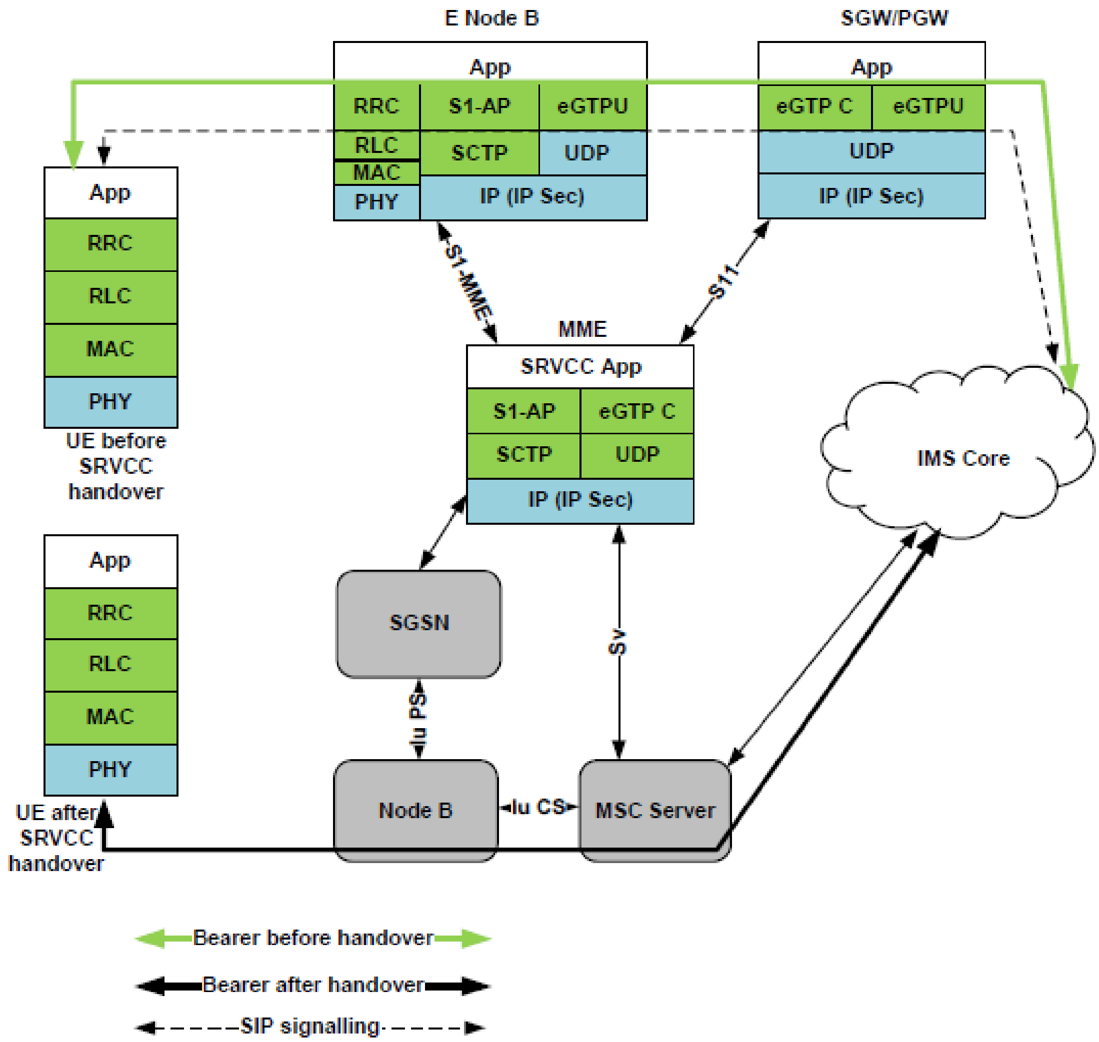

Fig 2.11 SRVCC Architecture [20]

The MSC Server handles the Relocation Preparation procedure requested for the voice component from the MME via the Sv interface, initiates the CS Handover and session transfer procedures.

2.10.2 Operation of SRVCC

When the UE moves away from the LTE coverage area, LTE Reference Signal Transmit Power (RSTP) starts reducing. The UE then notifies eNodeB about the change in the signal strength and SRVCC handover is initiated. The LTE network determines that the active voice call needs to be moved from the packet to the circuit domain. The SRVCC handover 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 and is able to transmit or receive on only one of these access networks at a given time. A new voice call request is sent to the IMS using a special number known as STN-SR which is a unique number that is generated for each UE and is stored in the HSS. The number is sent to the MME by the HSS when the UE first informs the network. Receiving STN-SR number indicates to the Session Centralization and Continuity Application Server (SCC AS) that the corresponding call needs to be routed to a different network, and it starts the redirection process to the legacy endpoint. Fig 2.12 summaries the Voice Call Continuity Operation

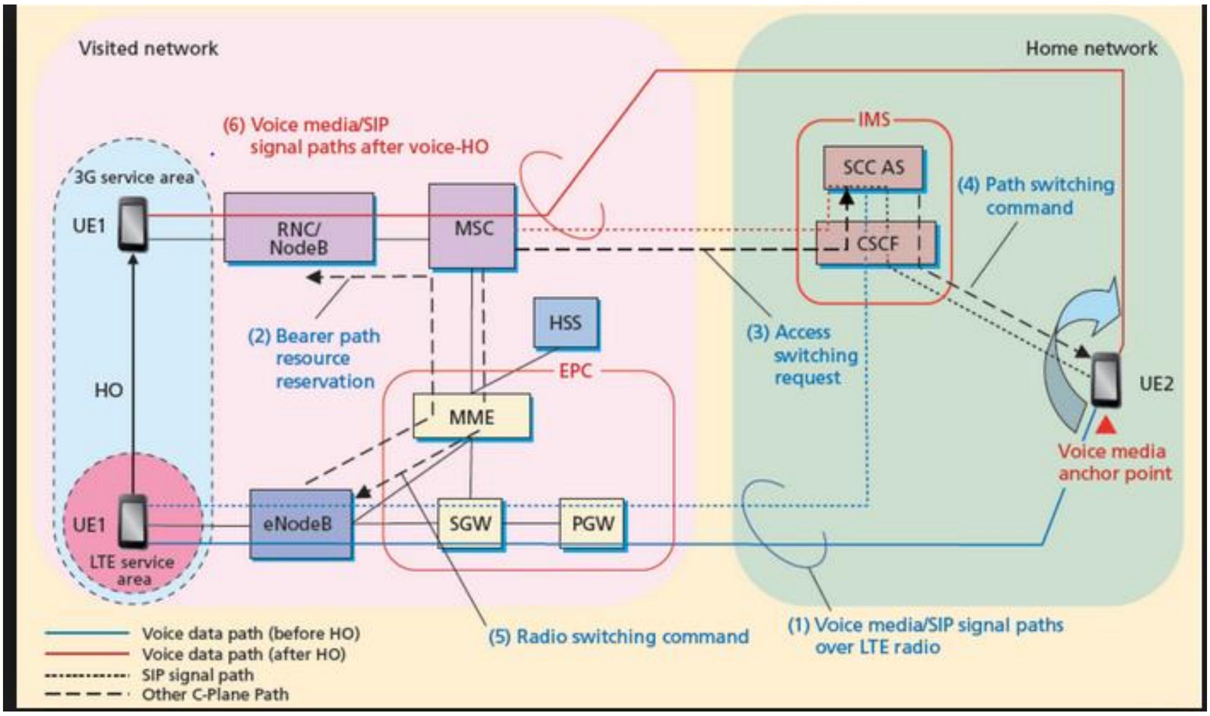

Fig 2.12 VCC Operation [15]

2.10.3 LTE coverage and into UMTS using SRVCC procedure

Fig 2.13 shows the procedures for LTE coverage into UMTS using the SRVCC handover

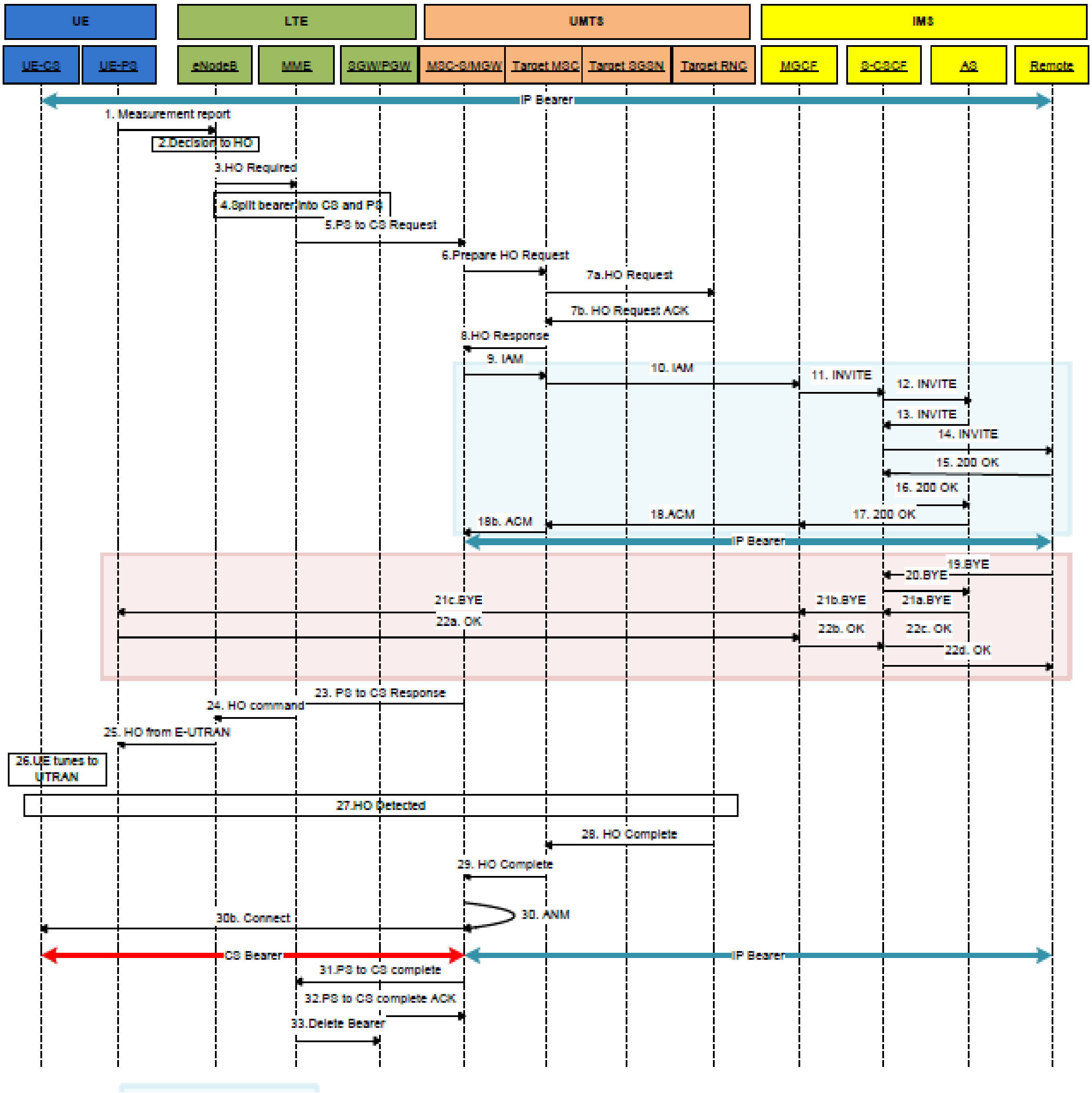

Fig 2.13 LTE INTO UMTS USING SRVCC

These Procedures do not really affect the delay of the handover (HO) process as they occur after establishment of the circuit [20].

2.11 Voice Call Continuity

This is an IMS application that provides capabilities to transfer voice calls between the Circuit-Switched(CS) domain and the IMS providing functions for voice call originations, voice call terminations and for Domain Transfers between the CS domain and the IMS and vice versa[20].

2.11.1 VCC Architecture

VCC is used for handover from UMTS to LTE. In this handover, all the 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. The domain transfer is triggered by an Intelligent Network making use of the CAMEL component [21]. Its architecture is shown in Fig 2.14

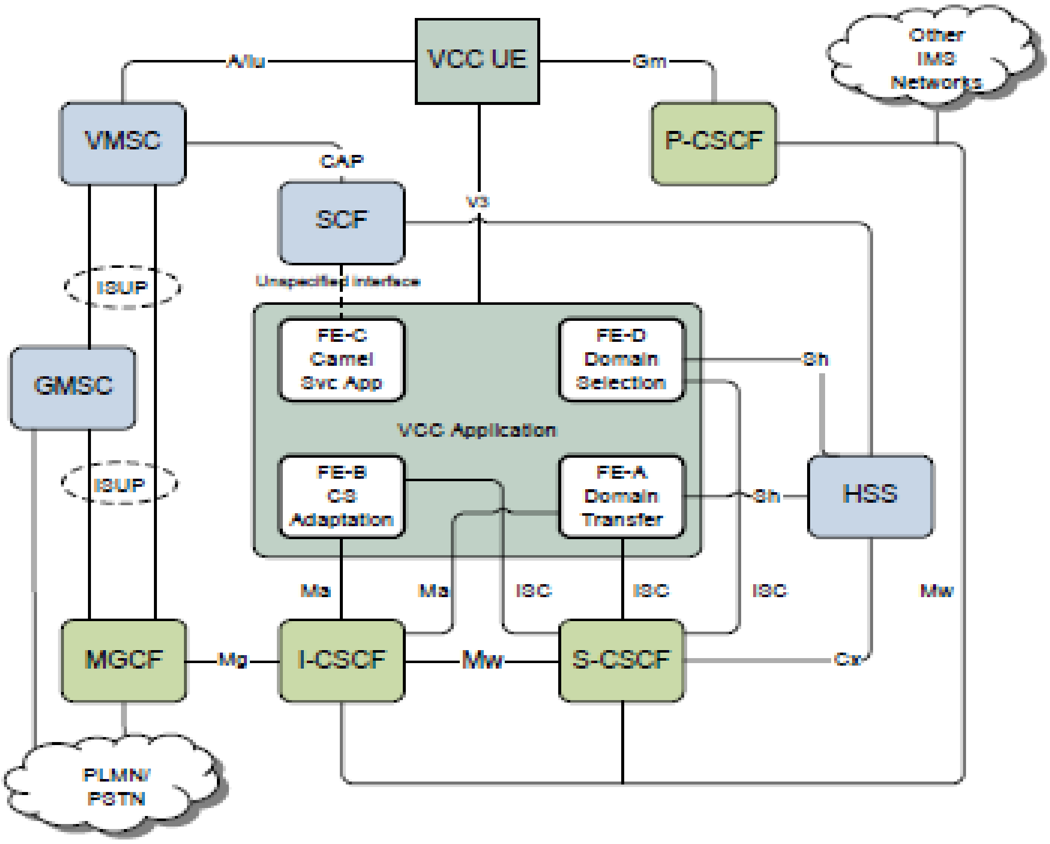

Fig 2.14 VCC Architecture [21]

2.11.2 How information flows for UMTS to LTE VCC handover

For UMTS to LTE handover to take place, the CS call must first be set up and then transferred to the IMS domain. The CAMEL is used for triggering of the VCC component. The information flow is shown in Fig 2.15

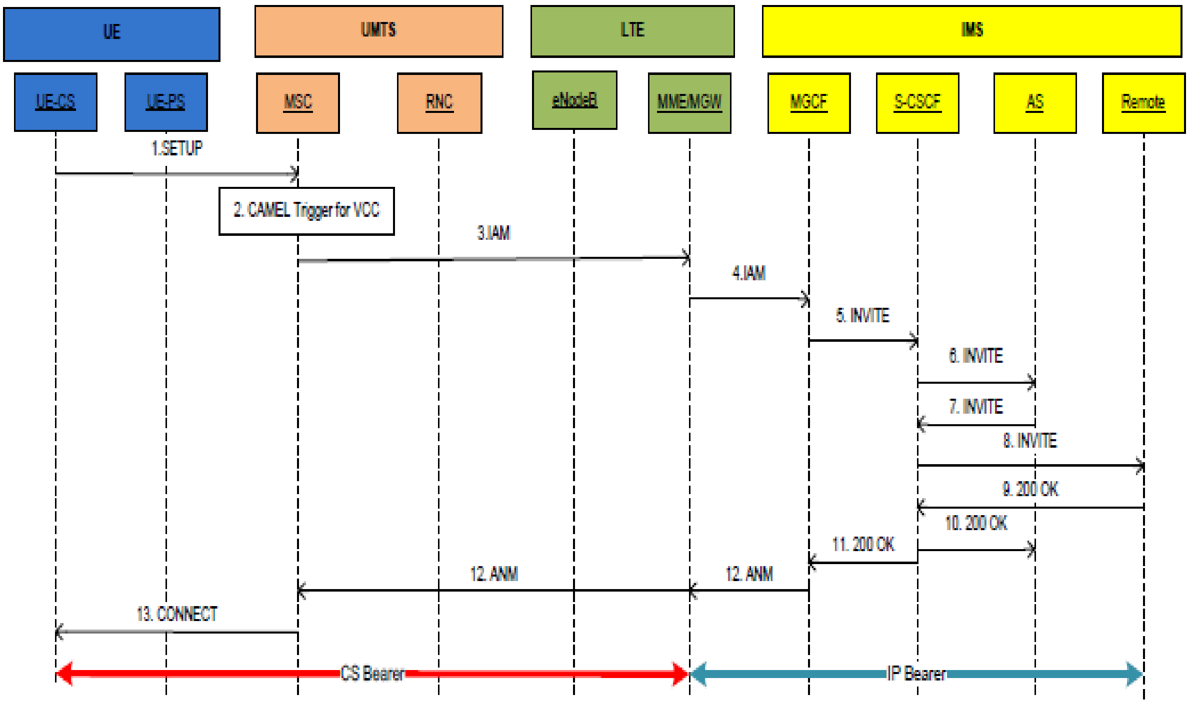

Fig 2.15 Information flow for UMTS to LTE VCC handover

The UE periodically measures the signal strength of neighbouring cells and sends a measurement report to the serving NodeB. The NodeB sends a report to Radio Network Controller (RNC) which decides if handover is necessary and sends Handover required message to MSC

Table 2.3 Different Parameters For Handover Schemes

Parameter	SRVCC	CSFB	VCC
Viable	Long Term solution	Temporary solution	Long Term solution
Time to implement	Standard Completed in Release 8, network fully implemented in 2012	Standard Completed in Release 8, network fully implemented in 2014 in Zimbabwe	Standard Completed Release 7, network to fully implemented in 2012
Operator Support	Supported by most operators	Supported by most European operators and a few African	Supported by Most Operators
Focus	VoIP controlled by IMS	No VoIP control	VoIP controlled by IMS
Cost	Cost is high but it is most feasible	Initial cost is low but performance is not the best, high cost handsets	Initial cost is high but sustainable

Table 2.2 Differences Between SRVCC and CSFB [10]

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.

a. Radio Network Optimization Process

Radio Networking Planning and optimization process play a very significant and vital role in optimizing an operational network to meet the ever-increasing demands from the customers The customer's quality expectations are very simple, they just want the availability of the service anywhere also anytime ,call setup time within limits, good speech quality during the call and normal termination of the call. Poor expectations are indicated by poor signal levels, high locking rates ,high bit error rates ,dropped calls/handover failures. If these expectations are not met, it forces one to perform some tests to come out with a possible solution. In order to understand the drive test on has to know about the radio network optimization flow chat shown in Fig 2.16

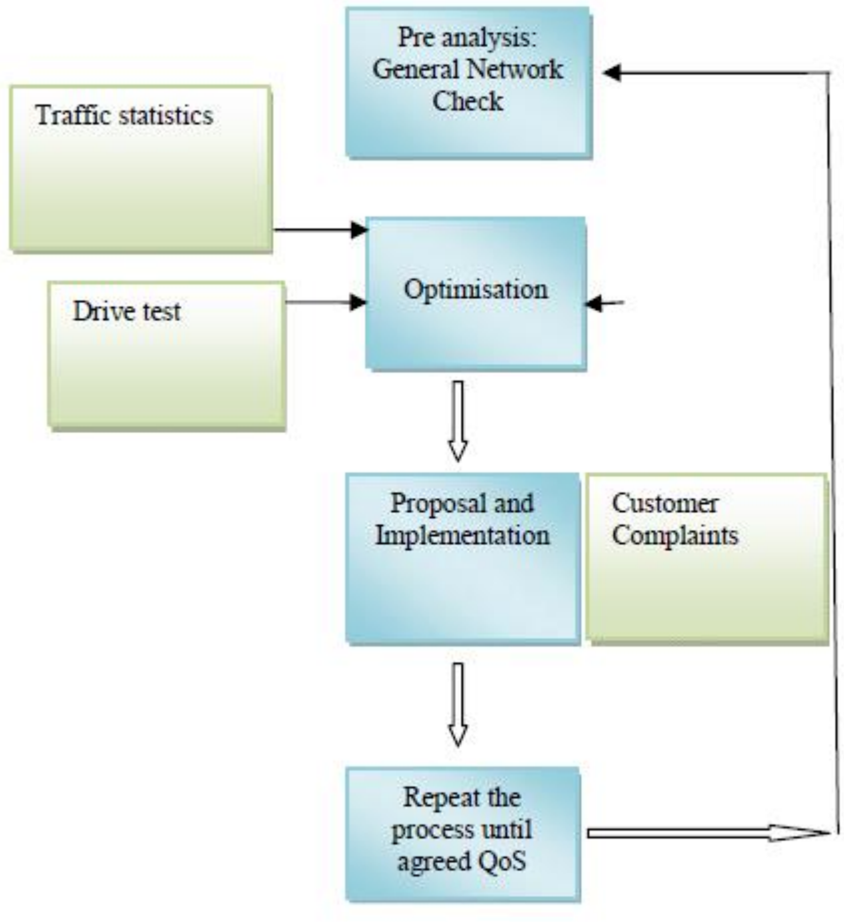

Fig 2.16 Optimization process [19]

It is also a relative process and requires a starting gauge of KPI's and goals. These can be derived from operator's individual outline rules, service requirement, client desire, market benchmarks and others. It plays an important role in optimizing an operational network to meet the ever-increasing demands from the customers. Fig 2.17 below shows the optimisation database

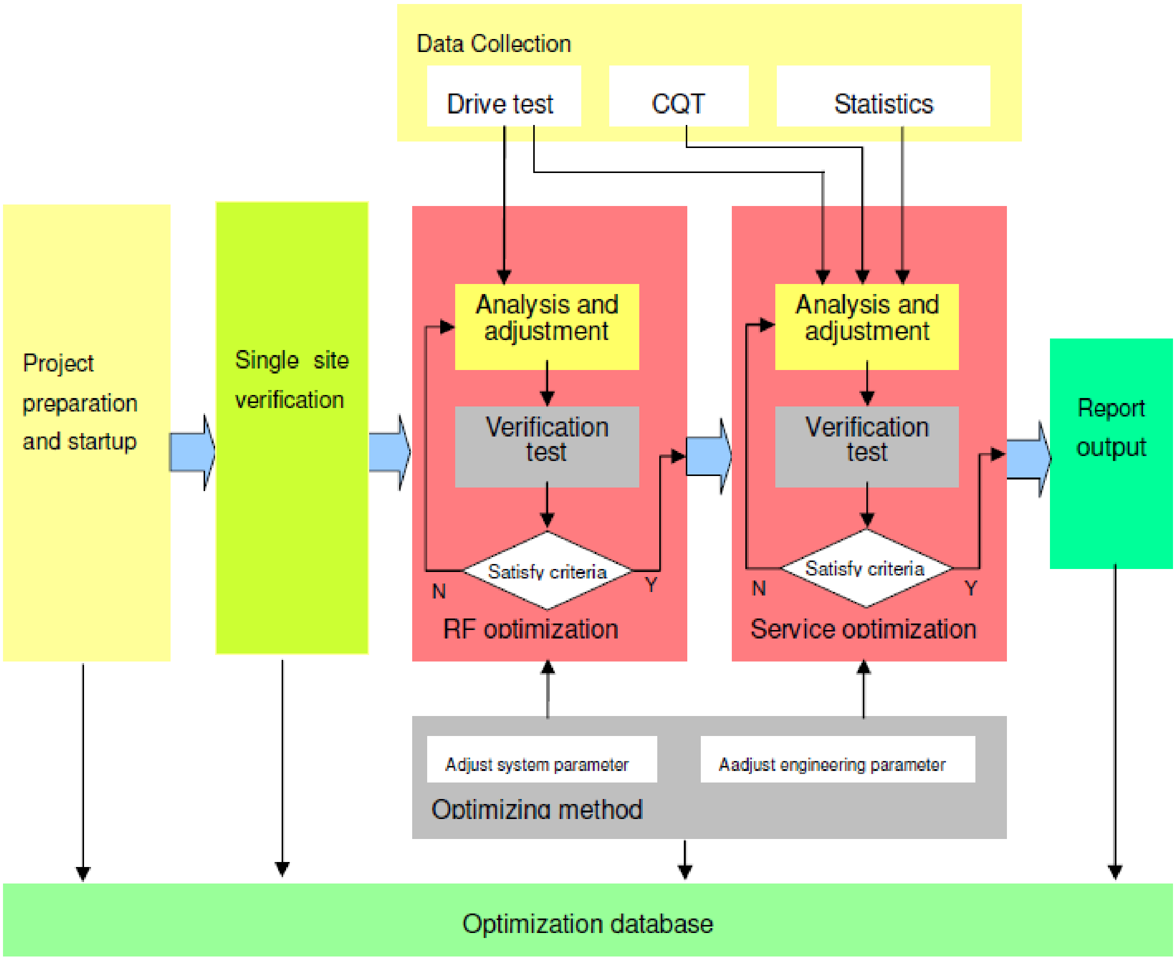

Fig 2.17 Optimization Database [21]

2.13.1 Mobility Management

This is one of the important issues of Heterogeneous Networks which set tasks required to supervise these mobile user terminal in a wireless network to check that it is always connected to the network even when moving [22]. This has several aspects which includes maintaining the Quality of Service, Handover Management, Location Management and power management. Also the aim of the mobility management is to track where the subscribers are allowing calls, SMS and other mobile phone services to be delivered to them.

CHAPTER 3

RESEARCH METHODOLOGY

3.1 Introduction

This chapter reviews the methods and research techniques used while conducting the tests determining the methods of gathering data and data presentation and data analysis in an attempt to address the critical question of comparing the evaluation of voice handover schemes used by the network providers in Zimbabwe.

The design of the methodology was more based on the research objectives of this research, therefore 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. As for the live network, in parallel with field test, you need to optimize the cells whose performance indicators have not reached the acceptance requirements.

In order to accurately evaluate the performance of a handover voice scheme, one must use a critical step by step approach. In this research, the first approach will be to perform drive tests in operational LTE, IMS and UMTS networks. Drive tests will 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. The second approach is by modelling UMTS and LTE network nodes their interfaces, properties, protocols and handover techniques using TEMS software. The third approach is through developing mathematical models that simulate the behaviour of the different message signal flows that occur during the voice handover schemes and this will be evaluated/modelled using the TEMS soft provided by the service provider (Econet Wireless)

3.2 Drive Test

Drive Testing is a system of measuring and surveying the scope, limit and Quality of Service (QoS) of a mobile radio system. Drive tests are carried out using a radio scanner to determine the distribution of UMTS and LTE coverages and quality samples in a defined geographical area. According to the network design and parameters, the TEMS software will evaluate how well the handovers are being performed and the evaluations include agreed test locations, coverage measurements, quality measurements, measurement of handover KPIs (success rate, drop rate, delays, etc.) analyses of rejection and failure causes on the basis of coverage and quality criteria as well as neighbour relations and proposal of corrective measures Drive tests will be done in two methods which are idle mode where the MS is ON but a call is not in progress and dedicated mode where the MS is ON and a call is in progress.

Table 3.1 Drive Test Call Procedures [15]

Test Methods	Purposes
Idle	Used for recording the network condition at the idle state and the level and Eo/Io
Dedicated –Short Call (180sec)	Mainly used for testing the accessibility and mobility of a network and for checking the successful completion of a call
Dedicated – Long Call (entire duration of drive test)	Used to test the retainability and sustainability for example Call drop rate and Success rate

Drive tests are done using a dual mode test mobile and a test server connected to the core network accessible over both the LTE and UMTS access networks. During the testing, the handover and reselection behaviour will be tested for both circuit switched and packet switched services. By doing these test and detailed analysis, one will be able to view the measurements for the timer-values, handover-relations and network parameter settings that control measurement and handover procedures.

Below is a model of how the equipment of performing the drive test was interconnected which include two test mobile phones, GPS receiver and the laptop which will be installed the TEMS software which is licenced and can only be used by licenced network operators. Also the base station analyser was used to perform some tests which were used to evaluate some of the results

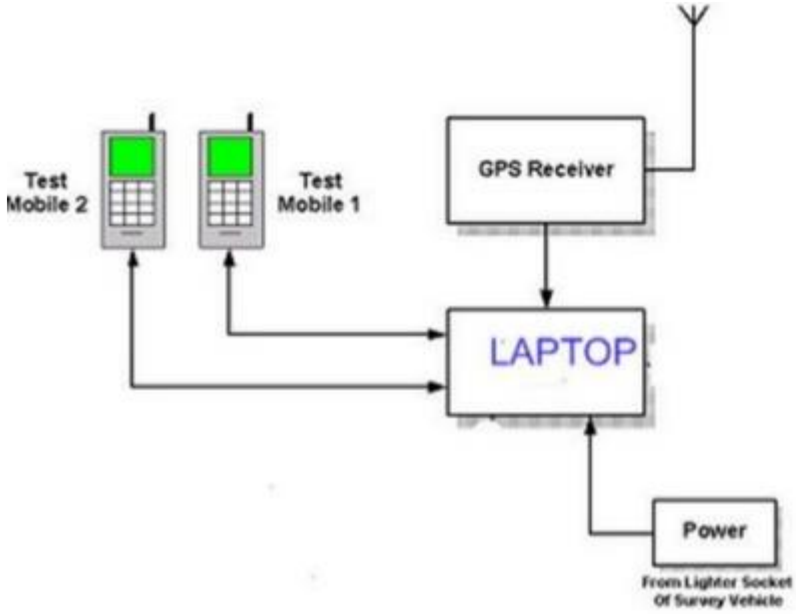

Fig 3.1 Test Setup [12]

Also the base station analyser was used to measure some parameters which were going to affect our evaluation of the voice handover schemes. Below are some of the parameters which the base station analyser can measure.

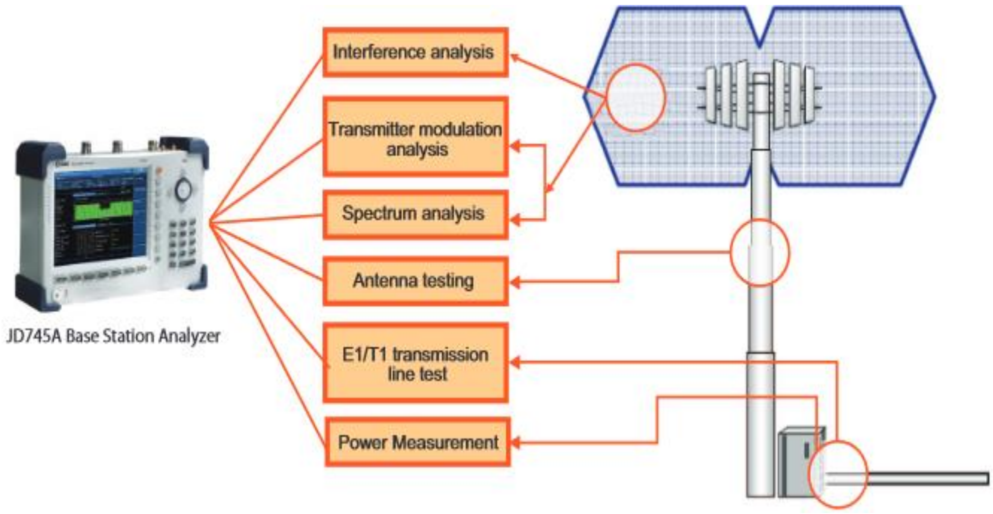

Fig 3.2 Base Station Analyser [21]

The drive test were done in Bulawayo CBD, since only in Zimbabwe LTE is only available in Bulawayo and Harare. Below is a coverage map for Bulawayo including base stations and the data was collected which was analysed letter after the completion of the test. The test were done using the TEMS software which was also used to perform simulations for the different behaviour of message signal flows

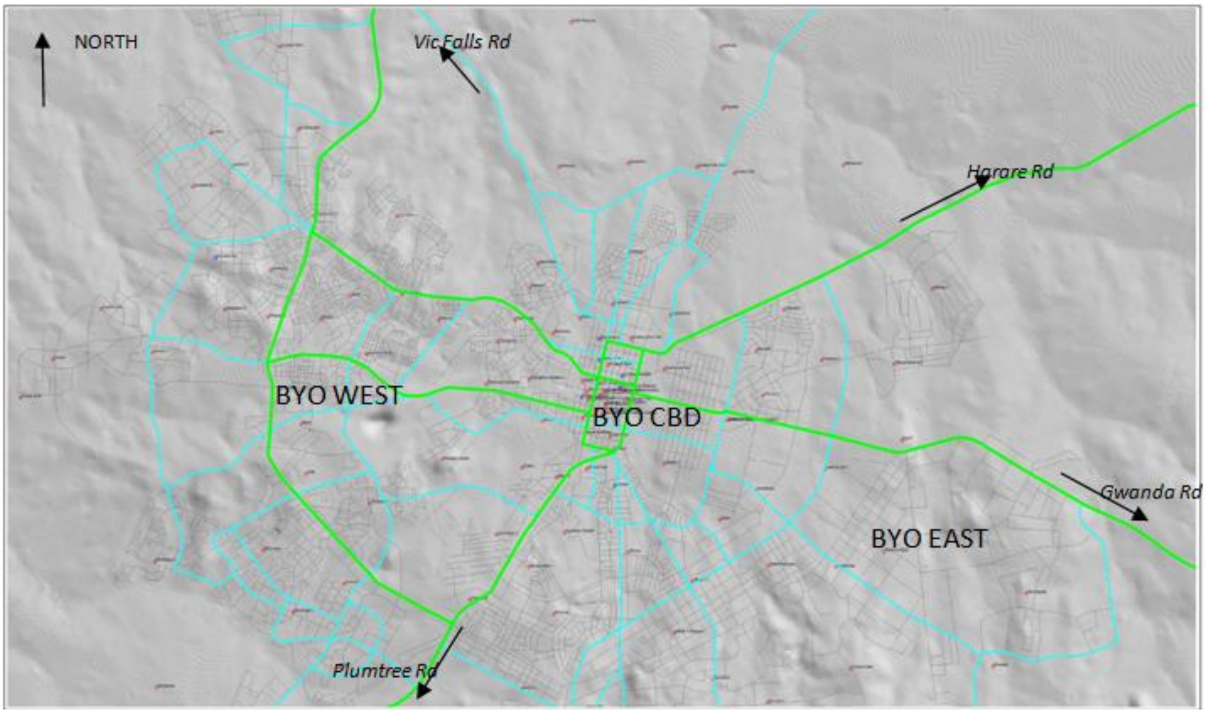

Fig 3.3 Bulawayo Drive Test Route

Also using the TEMS software, some of the snap shots for logging into the LTE and UMTS network were taken and are shown below.

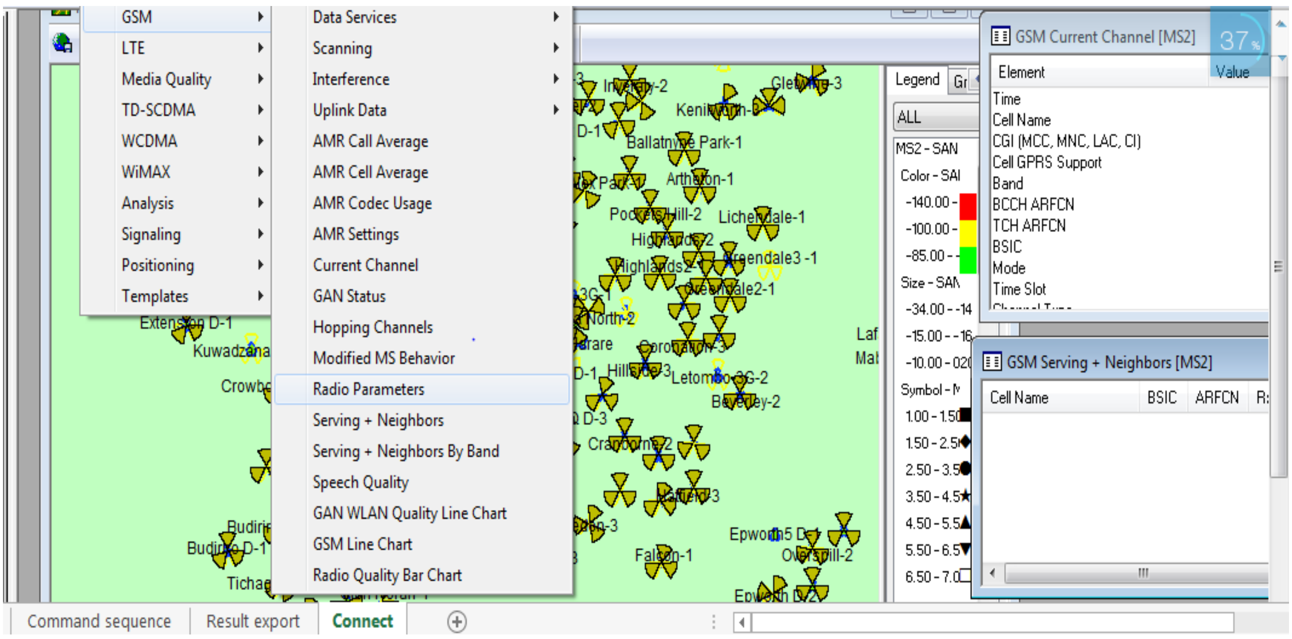

Fig 3.4 Logging in to specified technology

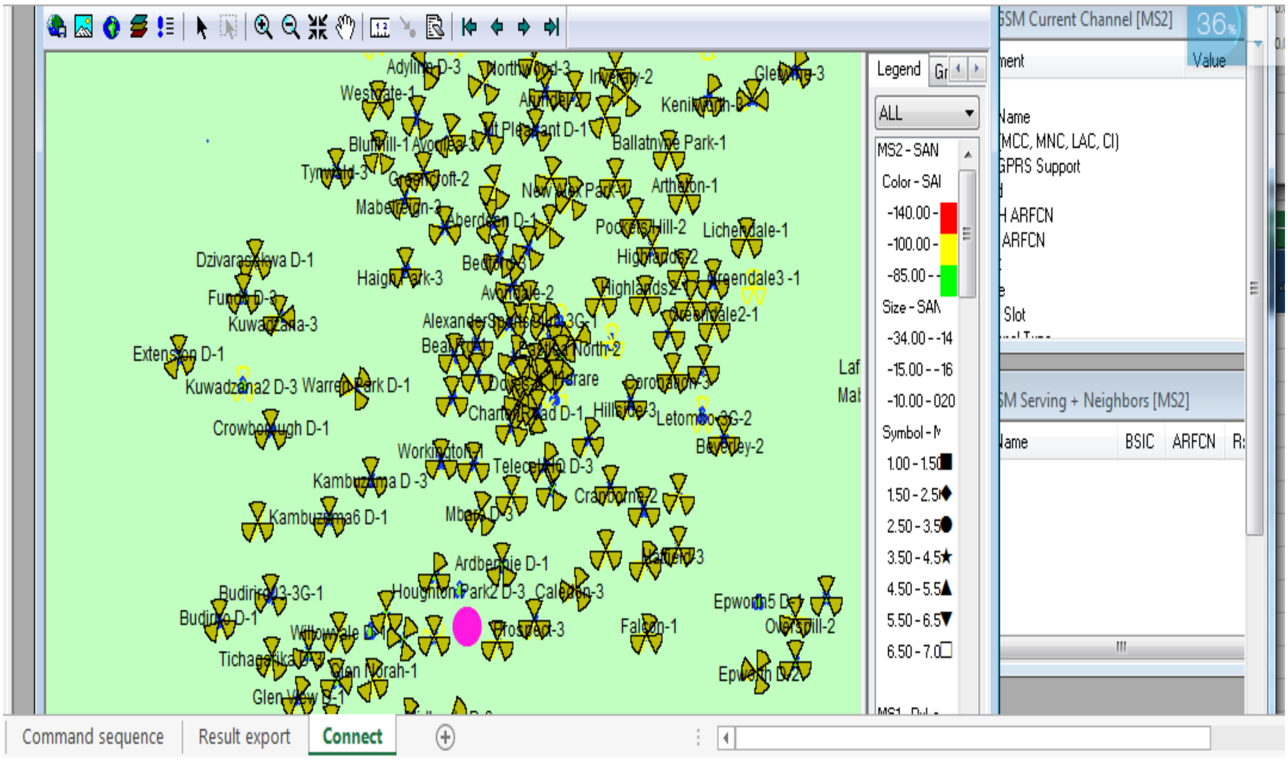

Fig 3.5 The Arrangement of Base Stations

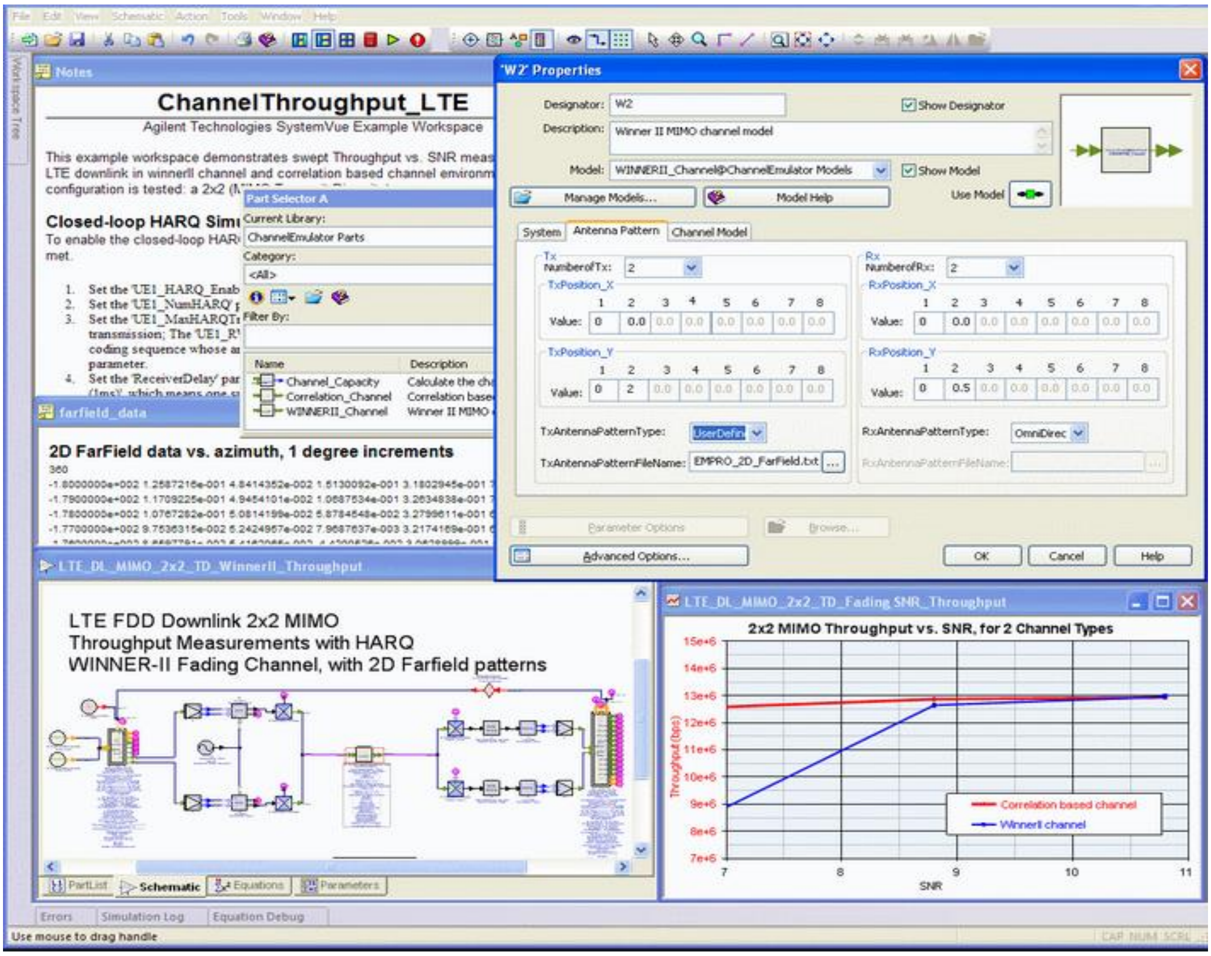

Fig 3.6 Channel throughput

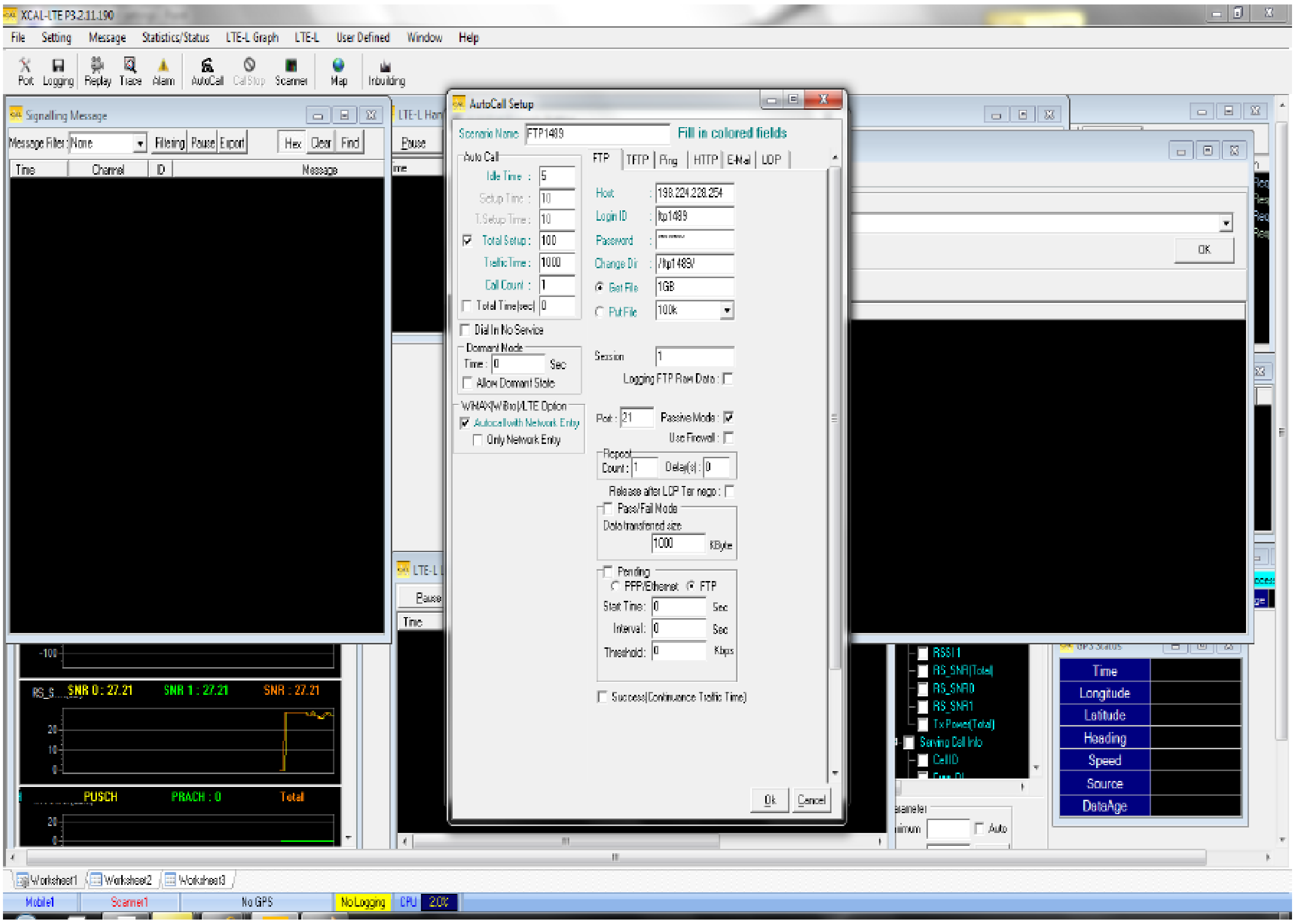

Fig 3.7 TEST Call Setup Parameters for Short Call

The fig below shows us different positions where dropped calls occurred due to some parameters which will be discussed in the next chapter

Fig 3.8 Drop Call Spots During Drive Test

3.3 Mathematical models that simulate the behaviour of the different message signal flows

With a specific end goal to evaluate the interference experienced by a call set from UMTS to LTE and vice versa, the handover delay will be split into two parts namely delay on the radio link and the network node queuing delay and each has a unique mathematical behaviour. These will be simulated using its own a mathematical behaviour

3.3.1 Radio Link Delay

For Inter-RAT handover to happen there must be correspondence between two radio advancements, for this situation its UMTS and LTE. Below is a diagram which is used to derive mathematical equations which were used to simulate for Radio Link Delay

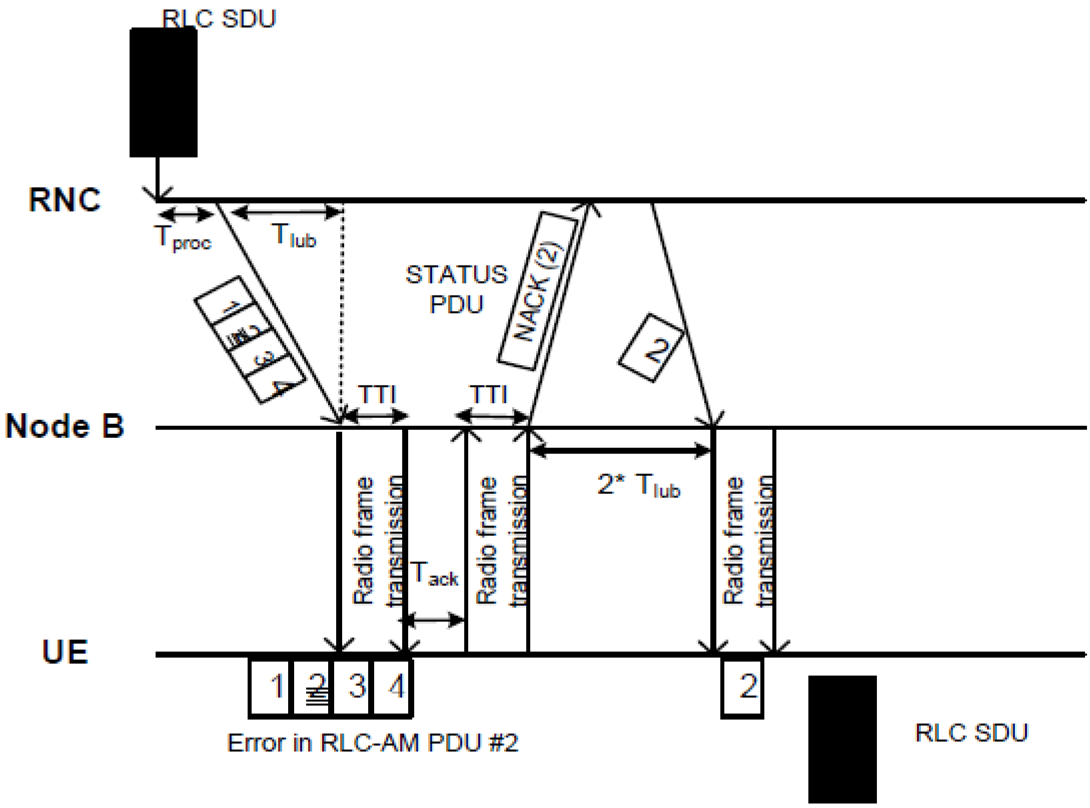

Fig 3.9 Frame transfer in UTRAN with RLC

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

$$D_1 = T_{proc} + T_{lub} + mT_{TI} \quad \dots\dots\dots\dots\dots\dots\dots\dots\dots\dots\dots\dots\dots\dots.. 3.1$$

The processing delay of an RLC frame

T_{Iub} - latency on the lub interface

TTI, the transmission time interval at the Node B

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

m is the number of TTI to send a frame

Also considering that T1 which is the time between the detection of an erroneous RLC sub-frame and the reception of its retransmission, the equation for the time of detection of erroneous of time RLC sub-frame can be written as

$T_1 = T_{ack} + 2 * (TTI + T_{Iub})$ ……………………………………………………………3.2

Accounting for errors we introduce a, which is the number of transmissions of the last correctly received sub-frame and therefore the delay after a retransmissions will be written as

$D_a = D_1 + (a - 1) * T_1$………………………………………………………………………………3.3

The overall equation after substituting D1 and T1 will be

$D_a = T_{proc} + T_{Iub} + mTTI + (a - 1)[T_{ack} + 2\ (TTI + T_{Iub})]$……………………………………………3.4

Delay corresponding to a transmissions

After calculating or simulating for the delay corresponding to a transmission, we also calculated the probability density function and considering that probability of receiving an error frame on the radio link (BLER) to be, p also the probability that a frame is correctly received after utmost *a* transmissions is (1-p^a). The mean delay for k frames after n transmissions will be

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

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

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

$$\overline{D} = \sum_{a=1}^{a_{max}} \left(D_a * P_{(D=D_a)}\right) \quad \text{……………………………………3.5}$$

The probability of successfully receiving a frame after n retransmission trials can be written as

$$P_s = 1 - p\big(p(2-p)\big)^{\frac{n(n+1)}{2}} \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots 3.6$$

And the overall delay over Radio Link Control (RLC) after n retransmissions can be written as

$$T_{RLC} = T_{Iub} + (k-1)TTI + \frac{k\big(P_s-(1-p)\big)}{P_s^2}\Big\{\textstyle\sum_j^n \sum_i^j \Big[P\big(C_{ij}\big)\Big(2jT_{Iub} + \Big(\frac{j(j+1)}{2} + i\Big) * TTI\Big)\Big]\Big\} \; 3.7$$

The simulation was done using the following parameters for both LTE and UMTS

Table 3.2 Parameters

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
T_{Iub}(UMTS, LTE)	3μs, 1 μs

3.3.2 Network Node Queuing Delay

Considering a 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.

$$\rho = \lambda/\mu \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots 3.8$$

Measure of demand on a queue

The average queue length can be found by using the formulae below

$$L = \rho/\big((1-\rho)\big) \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots 3.9$$

Also we applied Littles Theorem which states that L=Wλ, where *W* is the mean waiting time for a message waiting to be served by the MSC Server and the resulting average waiting time in a queue can be written as

$$W = \rho/\big(\lambda(1-\rho)\big) \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots 3.10$$

In the context of this research, the assumed values of arrival rate and service rate of network nodes in the UMTS, LTE and IMS networks are summarized in the table below

Table 3.3 Network Parameters

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

The simulations will be done in two phases, the3 first one is done under a static network and the second one is done under real time queuing conditions.

a) Static Network

It is assumed that we have a single user terminal with a predefined queuing behaviour that will be communicating with the UMTS, LTE, IMS and remote network nodes which also had static parameters. Below is a model which illustrates the behaviour of a static network

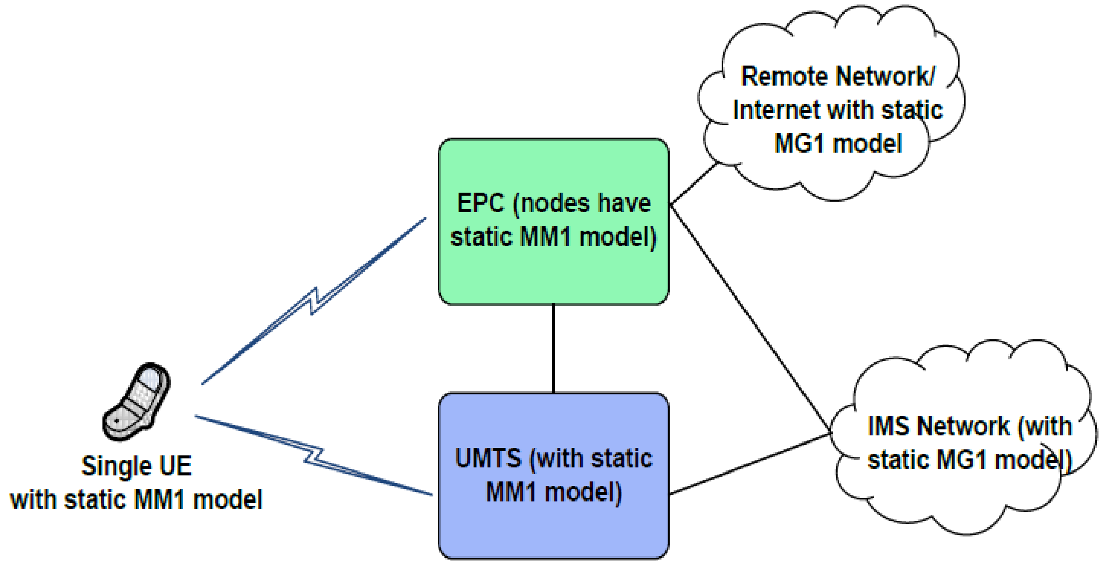

Fig 3.10 Static Network

b) Real time queuing conditions

The messages arriving at the network modes and remote network will be treated in real time and the observation will be done for every 300ms. This will capture the random nature of network traffic. The diagram below illustrates the behaviour in real time

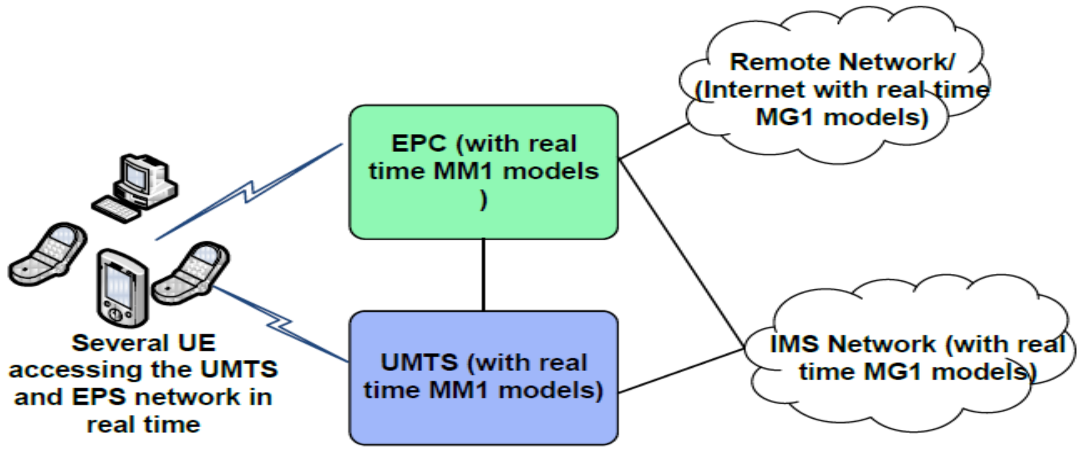

Fig 3.11 Real Time Conditions

3.2.2 Radio Resource Connection (RRC) Success Rate

RRC connection establishment is used to make the transition from RRC Idle mode to RRC Connected mode. UE must make the transition to RRC Connected mode before transferring any application data, or completing any signalling procedures, It is always initiated by the UE but can be triggered by either the UE or the network

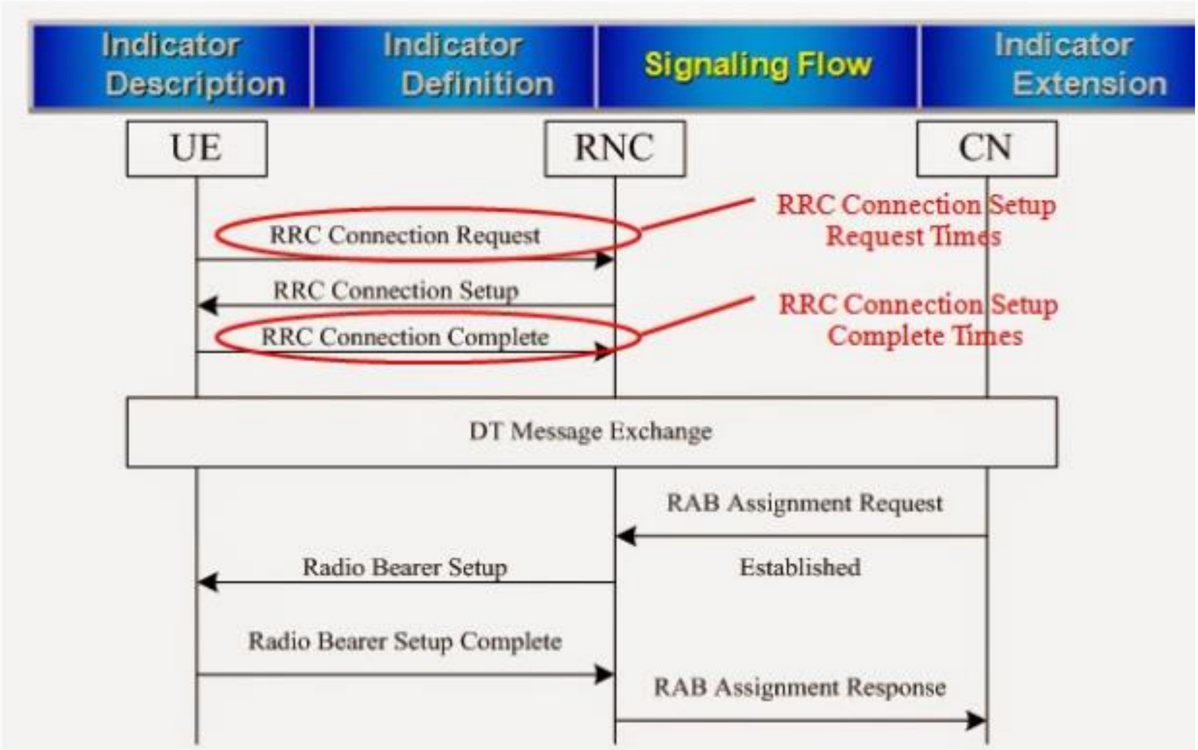

Fig 3.12 RRC Flow chat

RRC Establishment Success Rate= RRC Setup Complete Times/ RRC Connection *100

RRC Connection Request message is managed by the Non-Access Stratum (NAS) procedure for which the affiliation is being made. The relationship between establishment reason and NAS technique is shown by 3GPP TS 24.301

Table 3.4 RRC Request Message [21]

NAS Procedure		RRC Establishment Cause
Attach		Mobile Originating Signalling
Detach		
Tracking Area Update		
Service Request	User plane radio resources request	Mobile Originating Data
	Uplink signalling resources request	
	Paging response for PS core network domain	Mobile Terminating Access
Extended Service Request	Mobile originating CS fallback	Mobile Originating Data
	Mobile terminating CS fallback	Mobile Terminating Access
	Mobile originating CS fallback emergency call	Emergency

Radio Access Bearer (RAB) establishment success rate

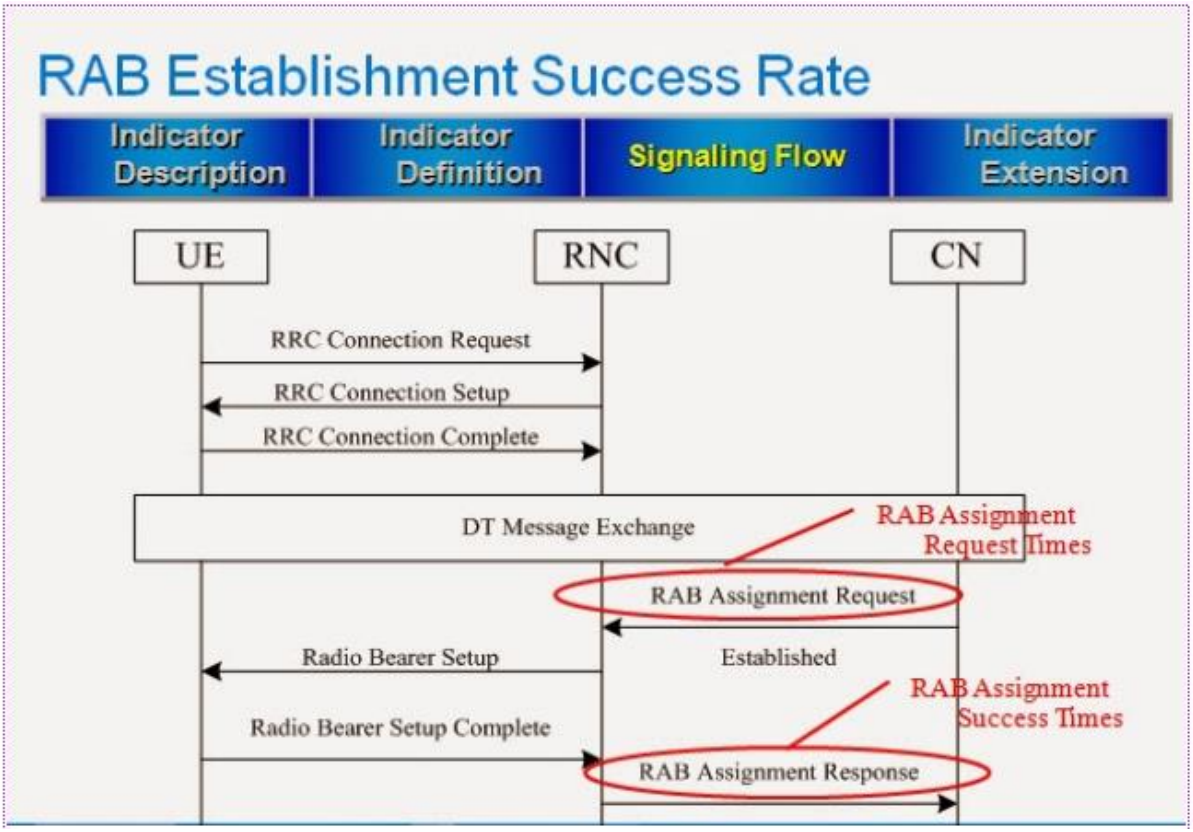

Fig 3.13 RAB Flow Rate Diagram

RAB Establishment Success rate = (CS RAB Assignment Success Times + PS RAB Assignment Success Times)/(CS RAB Assignment Request Times + PS RAB Assignment Request Times) *100

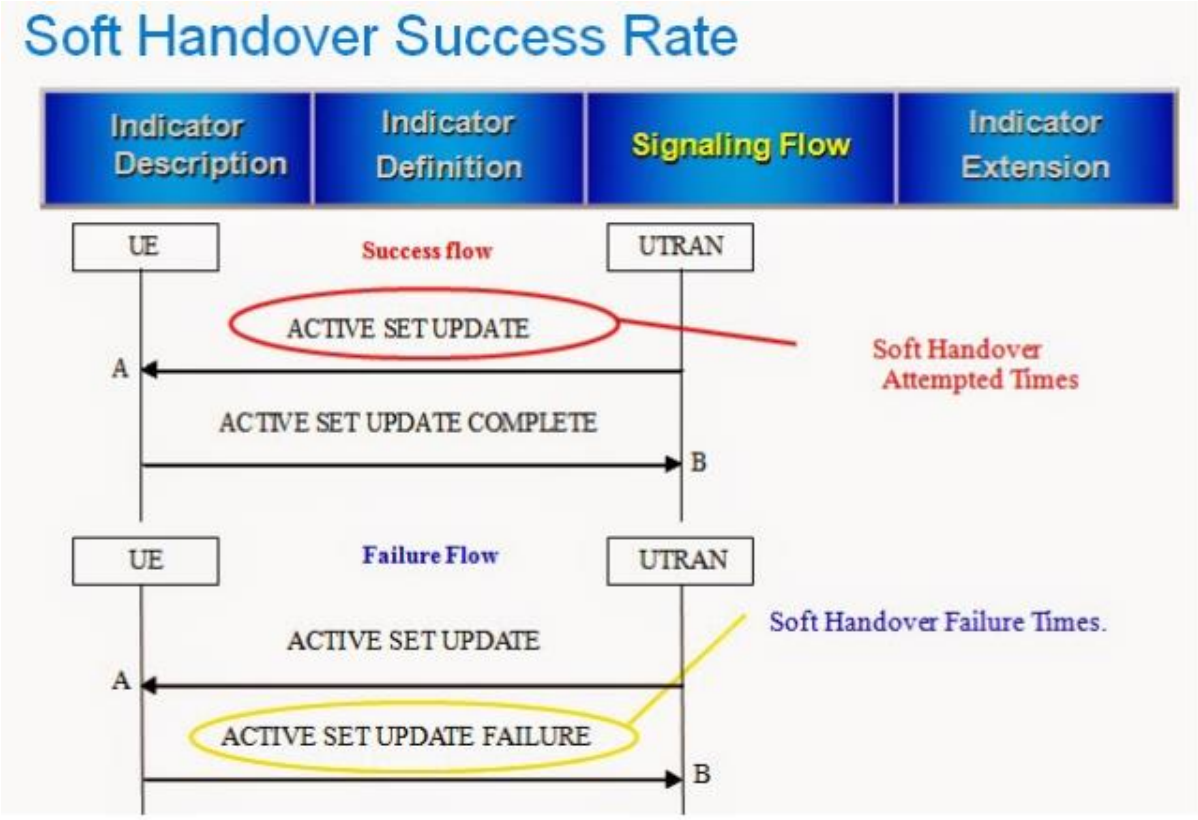

Fig 3.14 Soft Handover Success Rate

Soft Handover Success Rate = (Soft Handover Attempted Times-Soft handover Failure Times)/Soft Handover Attempted Times *100

CHAPTER 4

RESULTS AND ANALYSIS

4.1 Introduction

This chapter is divided in sections which summaries the results obtained. In section 4.2 disscusion of results of handover success rates of different handover schemes followed by section 4.3 which summaries the total number of SMSs per subscriber, total number of calls per subscriber and the mobile penetration rate of Zimbabwe telecommunications. Lastly section 4.4 which gives results of the simulations done using TEMS software licenced by Econet Wireless. Inorder to evaluate the correct type of voice handover which will be feasible and usable to be used in Zimbabwe, some critical analysis of the results were obtained which will be analysed and will be compared to the 3rd Generation Partnership Project which is an international board. The project had to be done through performing experiments on live network and perfoming simulations for different scenarios. In order to correclty evaluate the type of voice handover to be used, the bandwidth level which is allocated by POTRAZ and the number of subcribers also played a significant role in the analysis of the data

4.2 Success Rate for different Handover Schemes

The results of Handover Success Rate for different handover schemes were obtained from the data which was captured whiledoing drive tests and the graphs were drawn using data from Appendix 1 .

4.2.1 Circuit Switched fall back (CSFB)

From Fig 4.1, it is clear that the average of the CSFB handover is just above 94% which is not bad for a well organised Mobile operating company. Although the percentage of CSFB is meets the targeted Key Permonce Indicator an LTE device camped onto a femto cell would need to fall-back to 3G in the femto itself and adding 2G into femtos is extremely complex, while adding 3G increases cost, complexity and time-to-market. Athough CSFB brings about a good capital base to the company and also the users will be happy with the good Quality of service this handover scheme provides. As shown from Fig 4.1 below

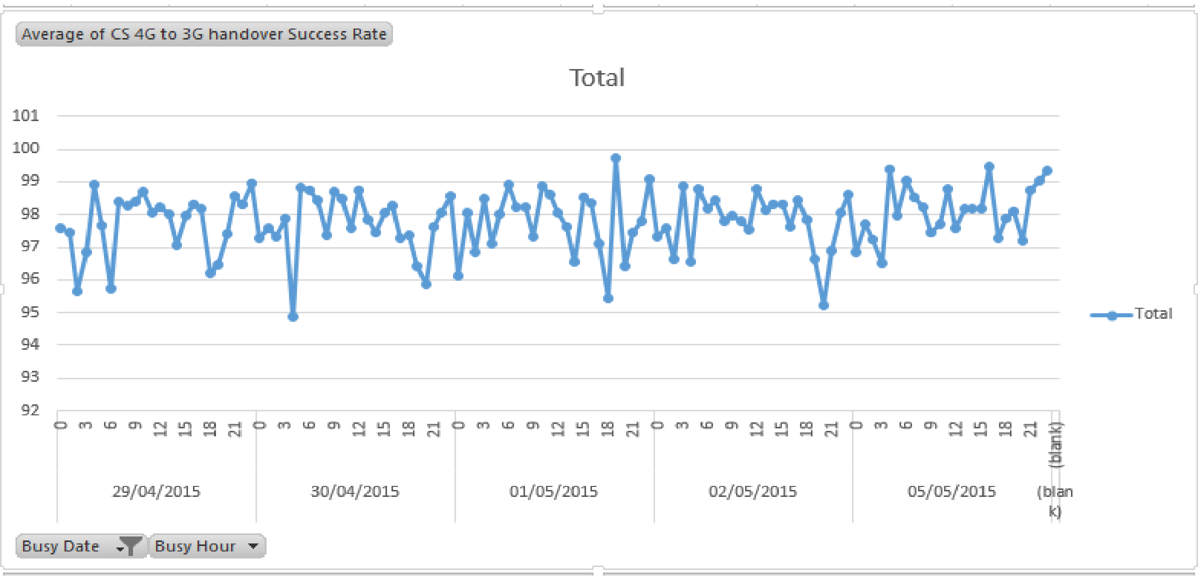

Fig 4.1 Average of CSFB Handover success rate

CSFB has disadvantages that it requires extra time to make or receive a call and one of the worst-case scenario probably occurs when two LTE handsets communicate with each other, and both need to fall back to 3G. Although some of the procedures may occur in parallel, the overall call set-up time is likely to be unreasonably poor. As seen from the graph on the 2nd of May 2015, that drop was due to fact that CSFB dropped any concurrently-running LTE data connection in the process.

4.2.2 Soft handover Success Rate

This is the most common handover used by many mobile operators in Zimbabwe and from the graph, it show a good success rate. This is due to the fact that it reduces the UE power up to 4db hence decreases interference and increases the battery life, also due to the fact that it also reduces the Node B (base station) power which also decreases interference and increases the capacity for many subscribers

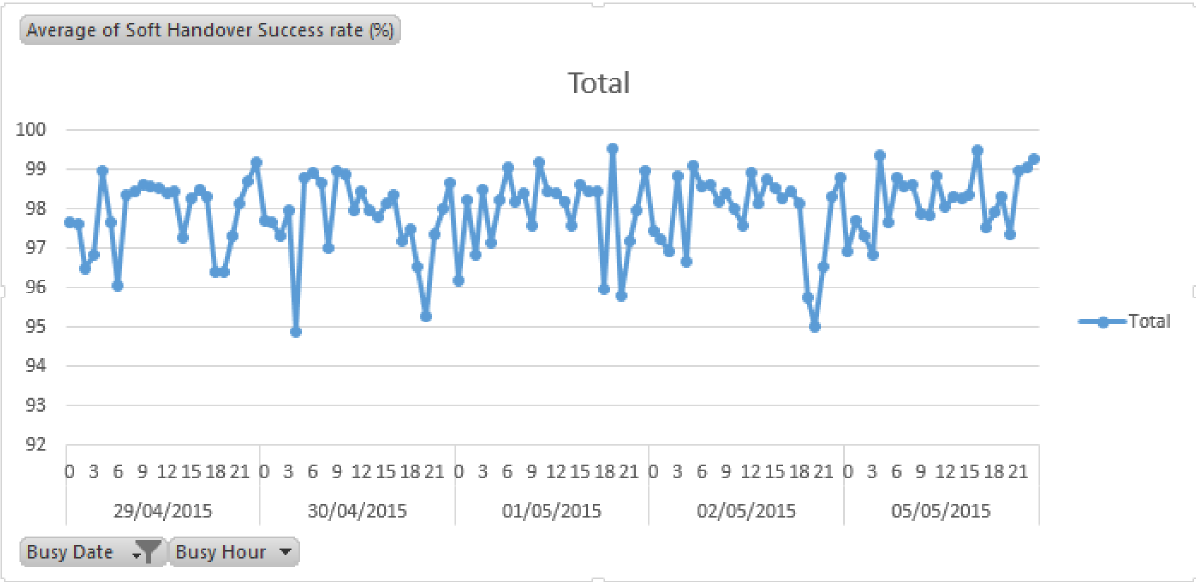

Fig 4.2 Average Soft Handover Success Rate

From Fig 4.2 it is clear that they is no consistence of results, this is mainly due to the fact that soft handover decreases the power needed by mobiles at the cell boundaries, also its effects on the downlink interference are quite complicated depending on such factors as the location of the mobile, the radio attenuation and the power division strategy are employed which will cost the company a lot of revenue

4.2.3 Single Radio Voice Call Continuity (SRVCC)

From Fig 4.3, one can notice that SRVCC handover can meet the required KPI standards in such a way that its average handover success rate is above 90%. As mobile operators, they consider the benefit to the company and also the best quality of service it gives to its customers

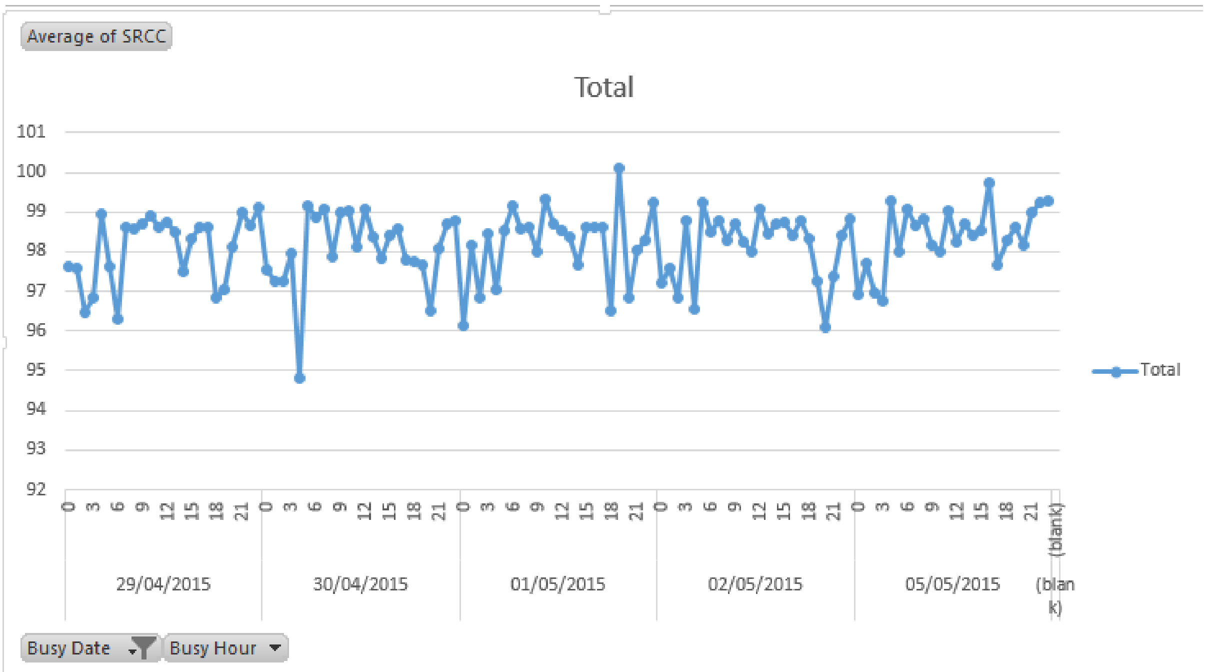

Fig 4.3 Average SRVCC Handover Success Rate

The main problem with SRVCC is that only a single radio User Equipment (UE) can access IMS-anchored voice call services which 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 gives rise to many complains from the customers where they will experience major call drops and many abnormal call sessions. Although the company can generate a lot of revenue from using this handover in internet access, they will not be making justice its valid customers.

4.2.4 Voice Call Continuity (VCC)

Fig 4.4 clearly shows us that the average voice call continuity success rate is just above 88% and does not meet the targeted key performance indicator of mobile operators which is 90%. The main reason for all those drops was due to the fact that it is not able to manage efficiently voice services during transfer i.e hold sessions are lost and the service configuration aspects are not covered due to lack of user to network interface

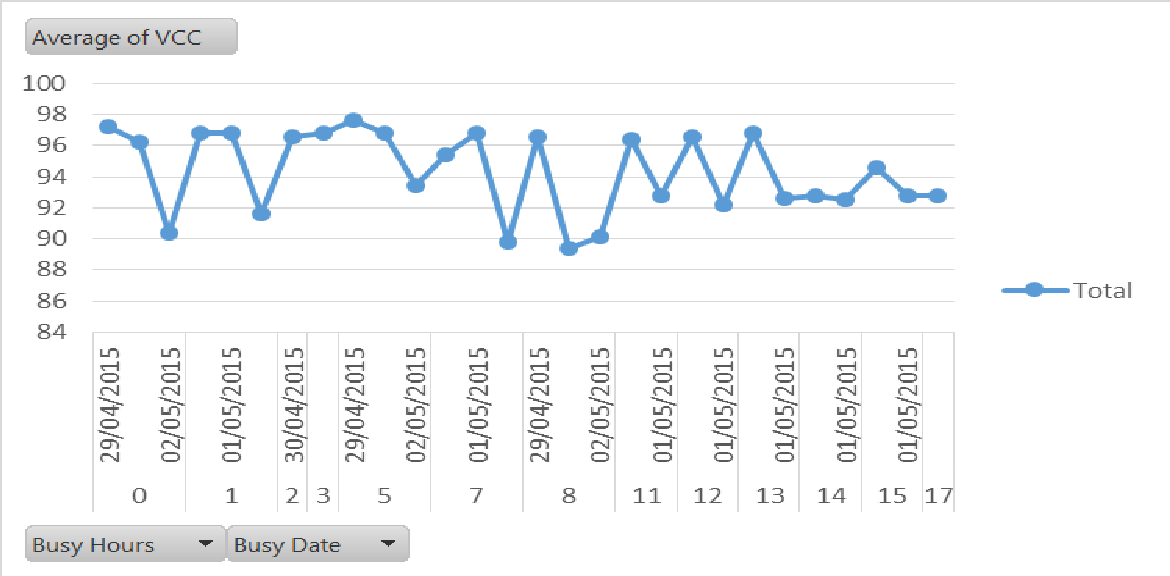

Fig 4.4 Average Voice Call Continuity

It is only limited to voice session only and requires VCC enabled terminals with dual radio capabilities which will be more expensive to maintain hence will not benefit the operator and below in Fig 4.5 are the parameters which were measured for VCC

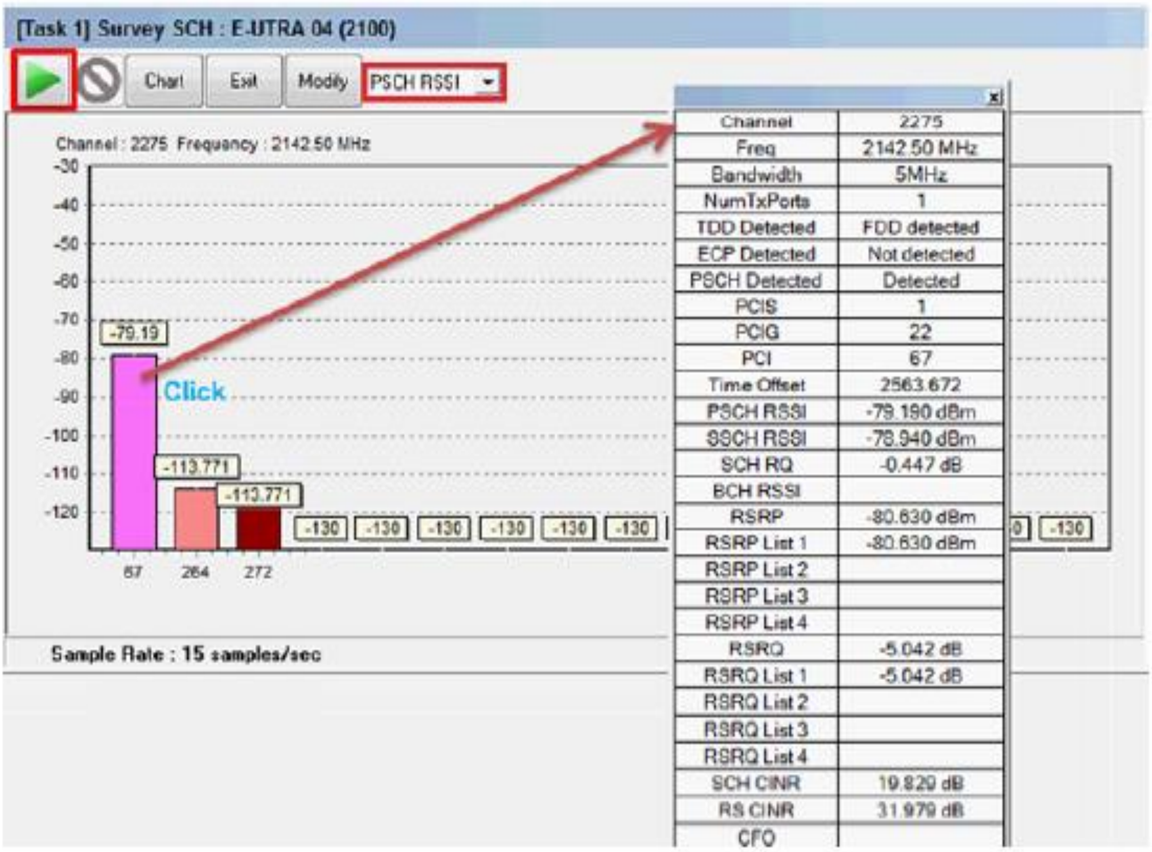

Fig 4.5 Analysis of Voice Call Continuity

4.2 Summary of SMS, Calls, number of base stations and mobile penetration

According to the report from Postal and Telecommunications Regulatory Authority of Zimbabwe , mobile network operators' revenues fell by 18% to close at $907.3 million for the year 2014. Fourth Quarter revenues also took a slump, with the 11.3% decline closing the quarter at $219.7 million [23]. This is against the third quarter revenue total of $248.7 million. The biggest decline in revenue was witnessed by Econet which experienced a massive 15.45% decline in revenue to finish the quarter with a total of $158.2 million. This was so because the number of subscribers continued to increase as they did not improve the type of voice handover techniques they used, people then opted to start using texts for convenience sake and with the introduction of instant messaging like WhatsApp and other alternatives. Below is the table showing the statistics of that.

4.3.1 Quarterly Total number of SMS

From table 4.1 below we can deduce that the number of sms per subscriber is also decreasing which is also due to the penetration in the market of some other affordable and user friendly social media applications

Table 4.1 Quarterly Total number of SMS [20]

	2nd Quarter	3rd Quarter	4th Quarter	Quarterly Variation
Total SMS	373,021,590	354,583,260	270,851,684	23.6%
SMS per subscriber	29	26	20	(6)

4.3.2 Quarterly Total number of calls

From the statistics shown in table 4.2, it can be noted that the number of total call and call per subscriber decreased more than that of SMS, some mobile operators may argue that this is because of some poor voice handover schemes which are not well maintained. Also the rapid increase of subscriber's without increasing the signalling bandwidth hence most subscribers will opt to use sms as a form of communication which is convenient to customers

Table 4.2 Quarterly Total number of Calls [20]

	2nd Quarter	3rd Quarter	4th Quarter	Quarterly Variation
Total Calls	280,567,476	252,432,123	221,145,908	35%
Calls per subscriber	14	10	8	4

4.3.3 Total number of Base Stations

As shown by the above table and the Statistics given by POTRAZ, in order to manage a good voice handover scheme, one must take into consideration the switching capacity and also the number of base station provided for each location. Some handovers discussed in this research drains the UE power if it does not find the next site given that the coverage area of that base station is almost out of range and this causes call drops. Also considering the number of subscribers at a certain area given a certain signalling bandwidth.

Table 4.3 Total number of Base Stations 1st Quarter 2015 [20]

OPERATOR	2G	3G	LTE	TOTAL BASE STATIONS	SWITCHING CAPACITY (subscribers)
Telecel	646	315	-	961	5,000,000
NetOne	513	144	-	657	6,068,000
Econet	2,125	841	19	2,985	9,280,000
Total	3,284	1,300	19	4,603	20,348,000

4.3.4 Mobile Penetration

Acording to the mobile penetration statistics graph in Fig 4.6 , it can be seen that it has increased over the past five years. This is mainly due to the availaibility of Subscriber Identity Module

(SIM) card. Also to increase the percentage of mobile penetration, the country should increase the network perfomance and to increase the strategies to improve on their voice handover schemes hence this will reduce the number of call drops and also improve on the revenue that will be generated.

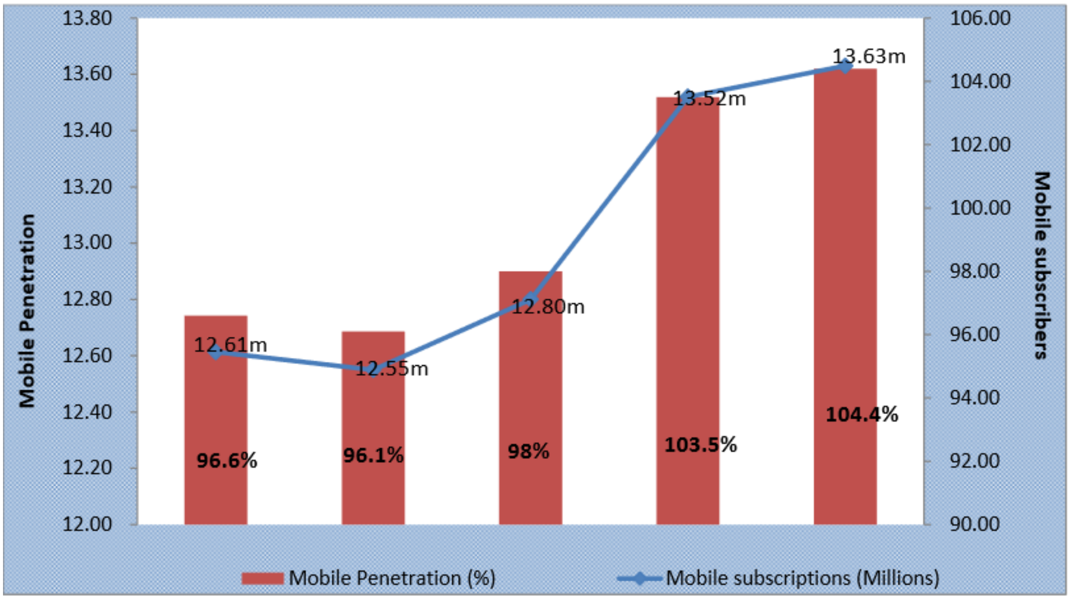

Fig 4.6 Mobile Penetration in Zimbabwe [20]

4.4 Results obtained from Simulations

The results in this section are from the simulations done using equations in section 3.3 of chapter 3,below are the parameters used for the simulations

Table 4.4 Simulation Parameters

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

4.4.1 Results of Latency vs Block Error Rate

From Fig 4.5 below it can be seen that for a given Block Error Rate (BLER), the service interruption time reduced considerably with increased data rate, some data rates affect the service interruption time of voice handover schemes. We can conclude that for a good handover scheme, it should operate with a considerably higher data rate to perform well and this will benefit the customers since they will not experience high level of call drops.

4.4.2 Graph for Service Interruption Time vs Block Error Rate (BLER)

From Fig 4.5 the data rate of 64kb yielded a service interruption time of slightly below 80ms which was acceptable for a good handover scheme and for data rate of over 64kbps, there was no significant decrease in service interruption time because Client Server messages have a specific frame length and 64kbps is sufficient to transmit this length.

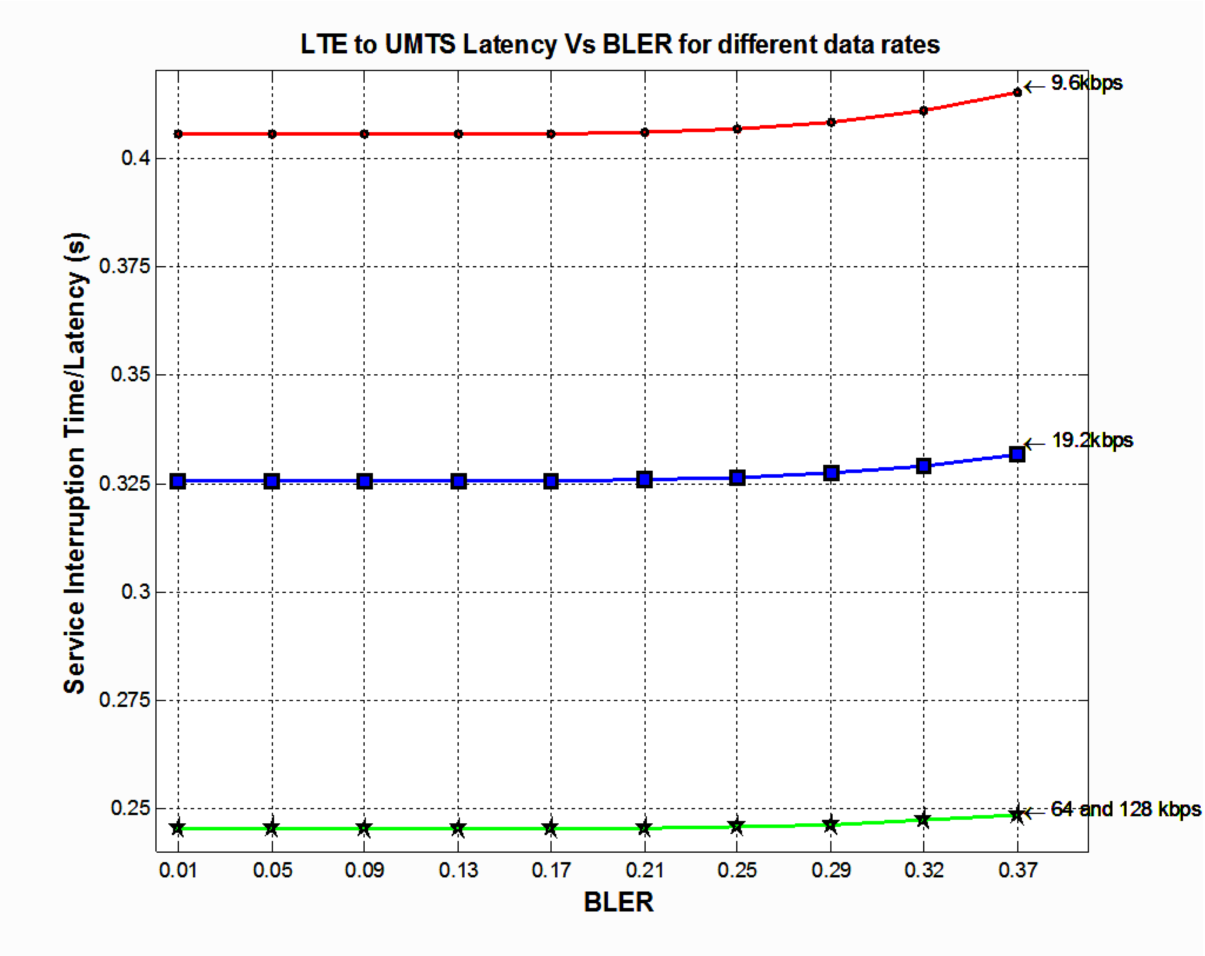

Fig 4.5 Graph of LTE-UMTS Latency vs BLER

.

4.4.2 Results of UMTS to LTE Handover Latency vs BLER

Fig 4.6 shows results of the simulations which were done when the User Equipment (UE) assisted in the handoff decision by measuring the neighbouring cells and reporting the measurements to the network, which in turn decides upon the handoff timing and the target cell/node. In this simulation the Quality of Service (QoS) is maintained not only before and after a handover, but during the handover as well. 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 results into more retransmissions thus a higher radio propagation delay

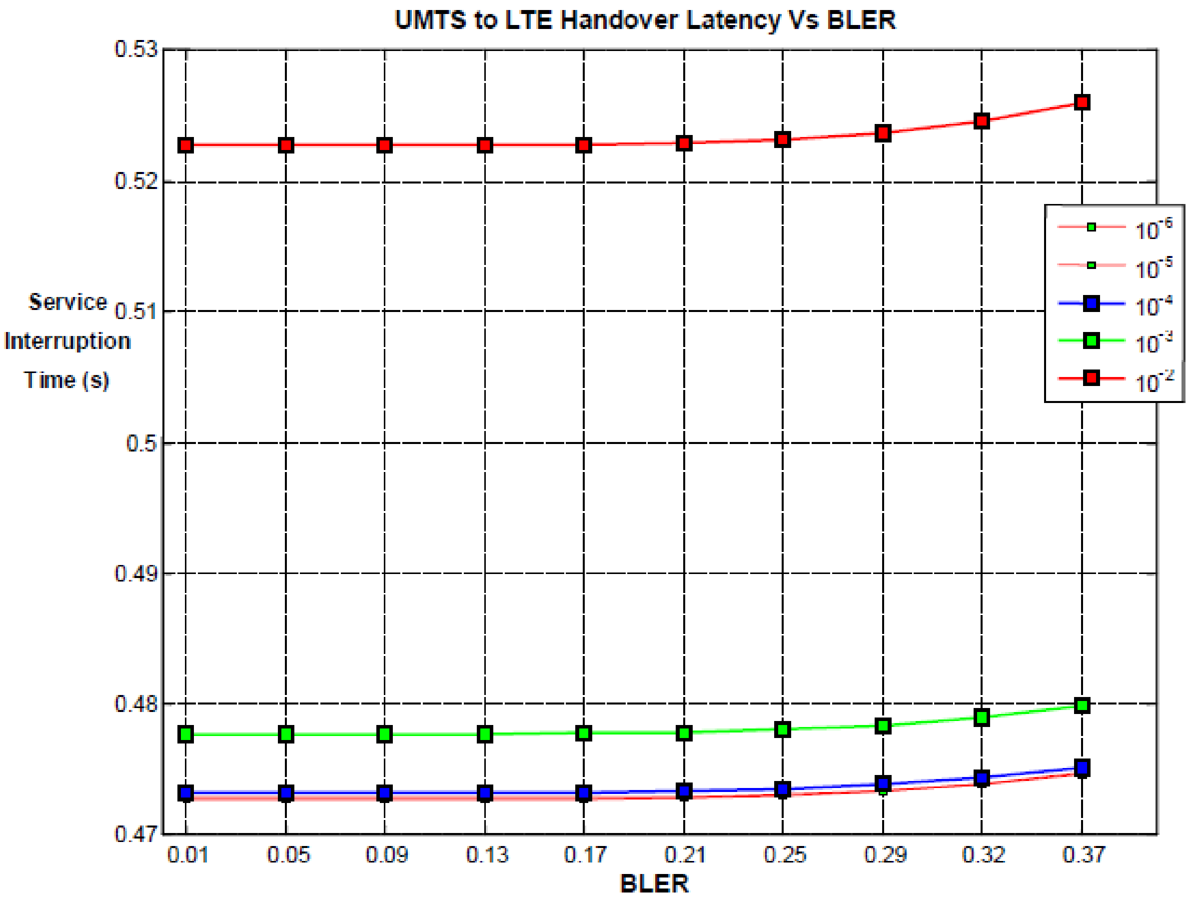

Fig 4.6 Graph for UMTS to LTE Handover

4.5 Results For Call drops, Average RRC and Session Success Rate

These are the result which also measures the network performance, they are the Key Performance Indicators (KPI) which help in trying to identify the possible handover scheme to be used.

4.5.1 Call/ Session Setup Success Rate

The graph in fig 4.3 shows the average call/session setup success rate which is the fraction of the attempts to make a call that result in a connection to the dialled number. From the graph we can see that it is above 97.5% which is above the standard one of 90-98%. Call set up success rate is a key performance indicator which is used by network operators to assess the performance of their network.

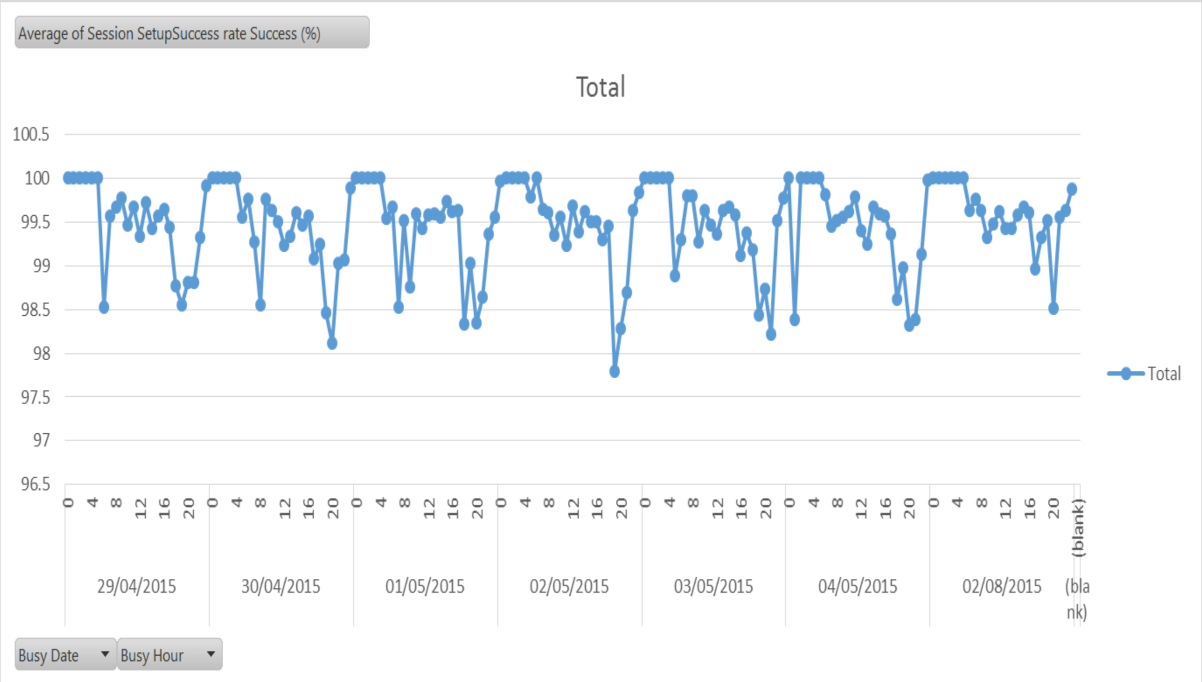

Fig 4.7 Call Setup Success Rate

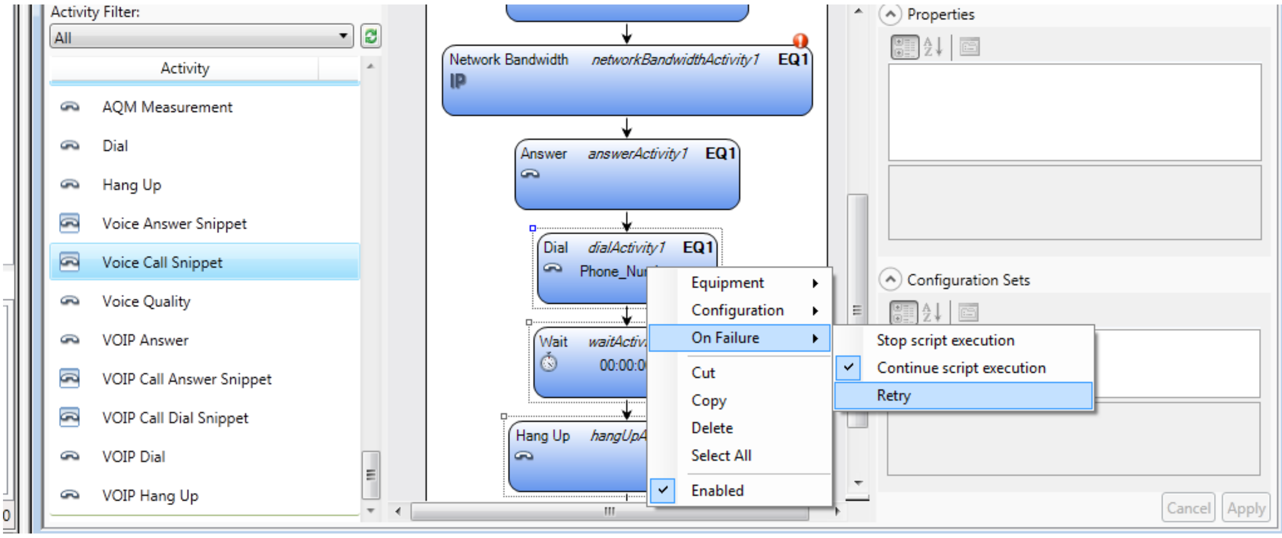

Fig 4.8 Analysis for the Call Success Rate

The call setup success rate has a direct influence on the customer satisfaction with the service provided by the network and its operator. The average success or failure of a call setup directly impacts the quality level for delivering the service by the networks as shown by the graph which is just above 97%. From the analysis of the graph we can see that at some point the call success rate dropped to 97%, this was due to poor radio coverage in the downlink of the base station.

4.5.2 Dropped Calls

Fig 4.4 shows the analysis of average drop calls for soft handover. One can see that it is just below 2.9% which is very fair for the subscribers and also benefiting the mobile operator since it increases the performance of a network and improves smooth operation of voice handover schemes. The main reason why the call drop rate increased to around 3% was due to the fact that when we were doing the tests we passed through a dead zone where there was no reception and also handover failure due to bad Rx quality.

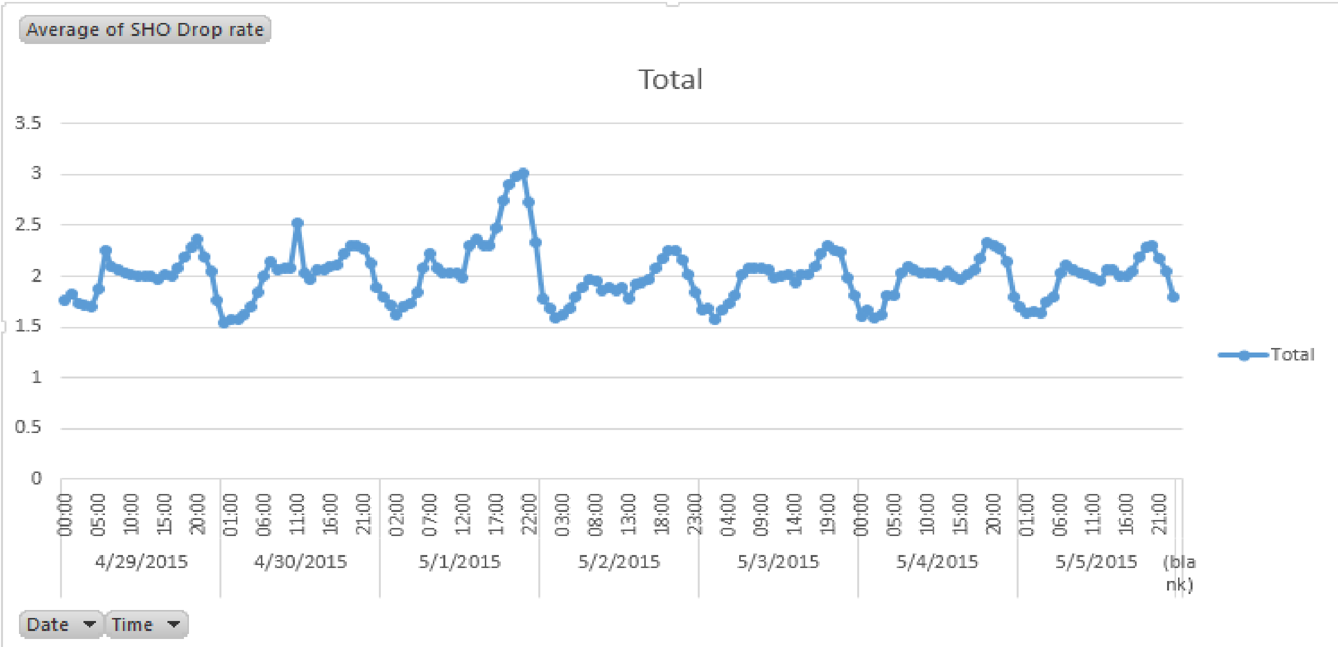

Fig 4.9 Soft handover Average Dropped Calls

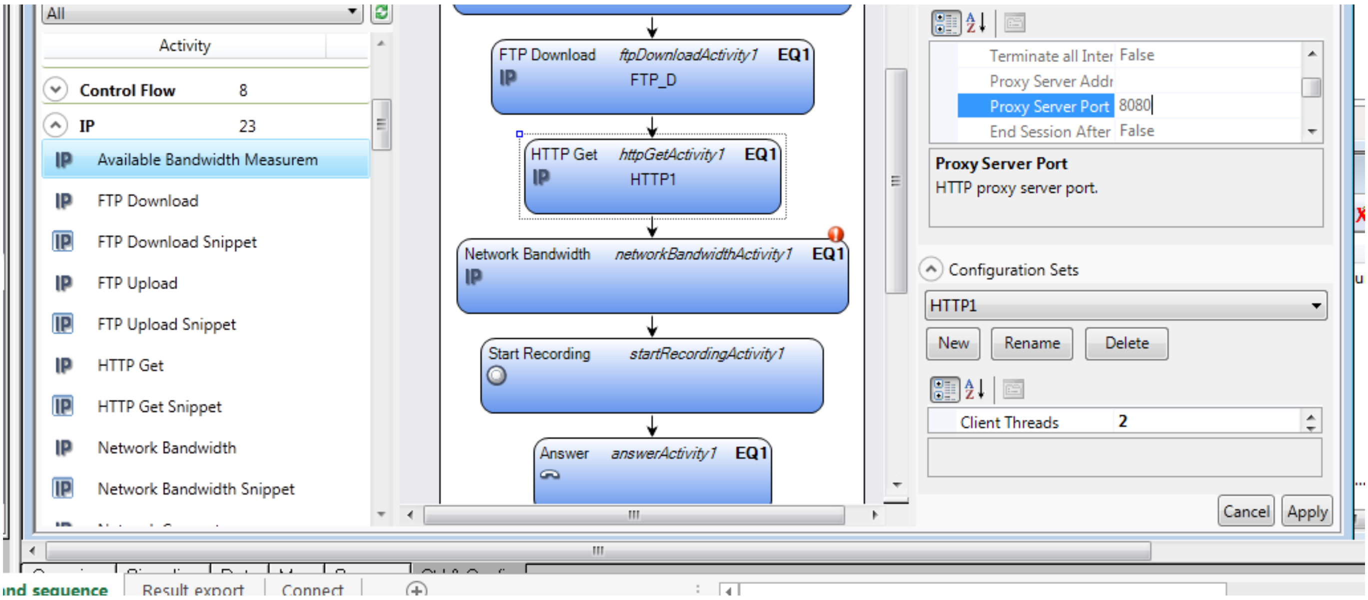

Fig 4.10 Results Analysis for Call Drops

4.5.3 Comparison Of average call drop between SHO and VCC

Voice call continuity (VCC) and Soft handover are the two handover techniques which are used for 3G-4G by mobile operators in Zimbabwe. As shown in Fig 4.10, it is clear that for SHO the call drop is around 2.9 % and for VCC it is around 9.8%

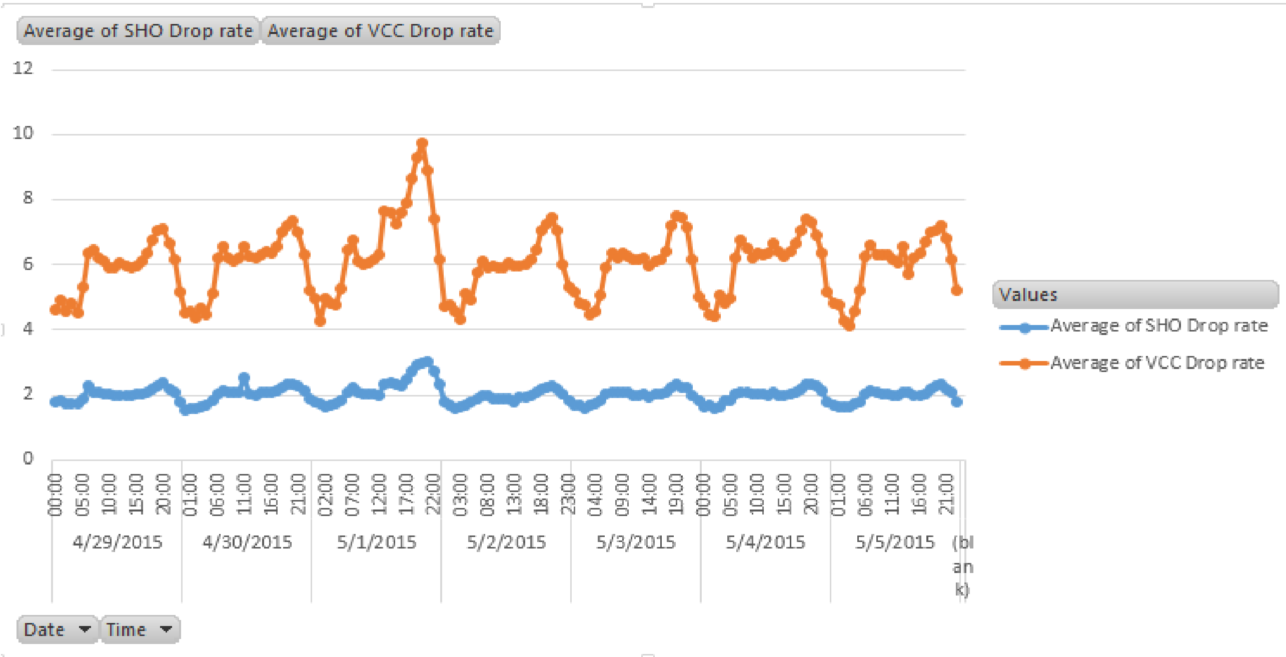

Fig 4.10 Average dropped calls for VCC and SHO

4.5.4 Comparison Of average call drop between CSFB and SRVCC

CSFB and SRVCC are the handovers used to measure the performance of a network between LTE-3G.Fig 4.11 shows the average call drops for these two handovers. SRVCC has got a percentage drop of around 9% compared to CSFB which has got just below 5%

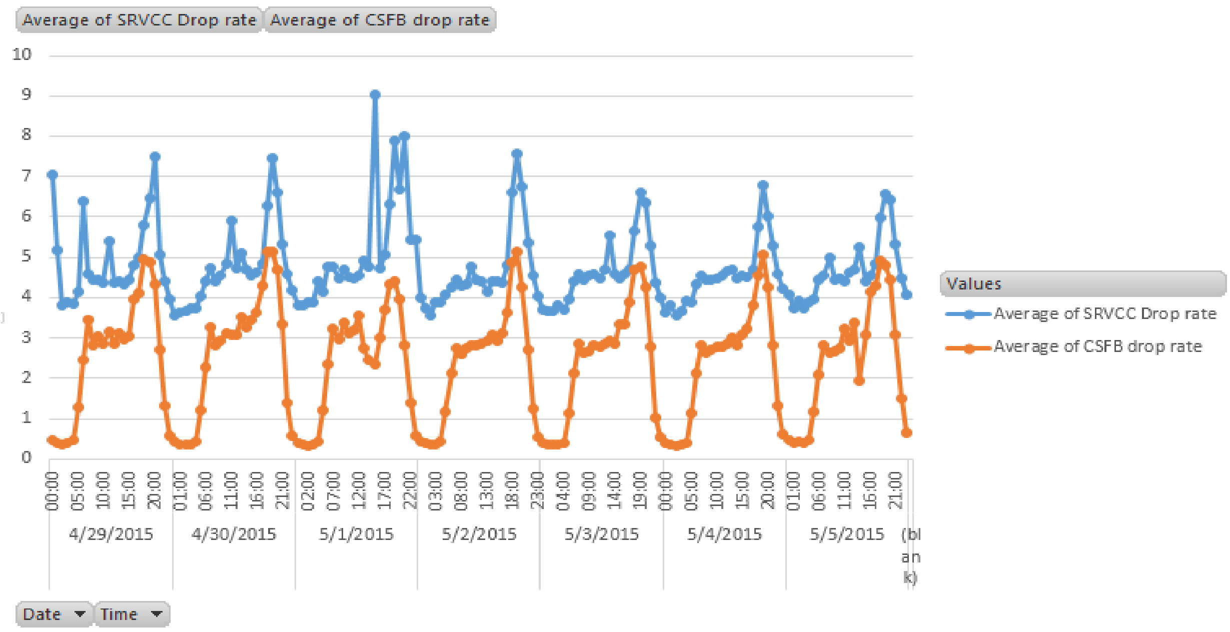

Fig 4.11 Comparison of Call drops between SRVCC and CSFB

4.5.5 Radio Resource Control Success Rate

From Fig 4.6, the average RRC success rate is just around 97%, the main reason for this RRC setup is to make sure that the network knows exactly the capability of a UE and provide the required services for each UE in the network so that it will not lead to a communication drop.

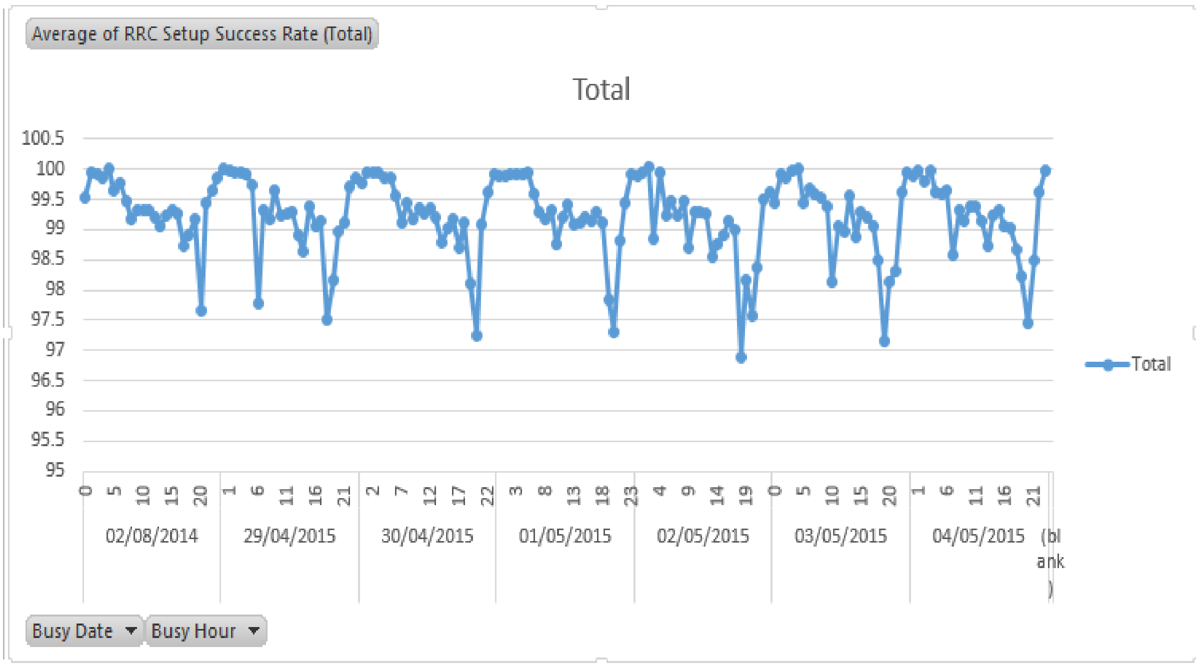

Fig 4.12 RRC Setup Success Rate

4.6 Conclusion

From the analysis of the graphs, it can be seen that for 3G-4G handover, the preferable handover scheme should be Circuit Switched Fall Back and for 3G-3G the handover which should be used is Soft handover. The critical analysis and key factors which a mobile company uses to choose a suitable handover scheme will be discussed in Chapter 5.

4.6.1 Findings

The introduction of LTE services into UMTS networks creates new challenges to network planning engineers. One major critical challenge comes from the requirement for providing a certain quality of service for LTE traffic without significantly degrading the performance of existing UMTS services. In a UMTS/LTE integrated network, it becomes necessary to reserve exclusive channels for LTE in order to provide base-line QoS for LTE users

CHAPTER 5

CONCLUSION

5.1 Introduction

This chapter clearly outlines the conclusion of which voice scheme should for 3G-3G, 3G-4G also 4G-4G which benefits both the operator and their customers, the advantages of implementing LTE all over the country since it is only available in Harare and Bulawayo. Also the future recommendation of improving the current schemes available at the moment. Network operators chose a handover scheme based on several factors which include

- Handset capabilities
- Technical Manpower
- The cost of implementation
- The easy of deployment

5.2 3G-3G (UMTS-UMTS) Schemes

Currently for 3G-3G technology, tests were done for soft handover and Voice Call Continuity (VCC). Firstly one have to mention that these schemes are well defined by an international board called 3GPP. From the graphs for soft handover and VCC, the average handover success rate are 98% and 88% respectively for both schemes, but taking into consideration some other parameters for both schemes, in mobile network industry the most expensive thing to maintain is signalling band, for VCC although the success rate is not that good, it does not require the terminal to fall back to circuit switched each time a call is made and therefore this reduces the amount of network signalling thus a cost saving on the network resources. For software handover it removes the need to camp on two radio networks simultaneously which reduces the user terminal complexity. Above all the recommendations VCC did not meet the recommended 3GPP service interruption time of 120ms which made soft handover the required voice handover scheme for UMTS-UMTS.

5.3 3G-4G (UMTS-LTE)

For 3G-4G, the handover techniques which were under test are Circuit Switched Fall Back (CSFB) and Single Radio Voice Call Continuity (SRVCC).As shown from the graphs in chapter 4, the average handover success rate for both graphs is 94%. The main problem with the current LTE network in Zimbabwe is that, it cannot hold a call for more than 20seconds in the LTE network. For SRVCC to work properly we had to improvise some aspects of Voice over LTE (VoLTE) which was very expensive to maintain. For the time being Econet Wireless is relying on Circuit Switched Fall Back (CSFB) for the provision of voice services. When there is a need for a voice call (either incoming or outgoing), an LTE handset falls back from the LTE network to either 3G WCDMA or 2G GSM and establishes a traditional circuit switched connection. This allows an LTE subscriber to benefit from the well-established voice services available on existing networks. However, the approach has some disadvantages

- Call setup times are extended, because of the need to switch radio networks as well as establishing a circuit switched call.
- 2G and 3G radio technologies use spectrum less efficiently than LTE, so it is wasteful of capacity to use GSM or WCDMA when LTE coverage is available.
- It is quite possible that a handset operating with a strong LTE signal, for example indoors at 800MHz, might fall back to a weak 3G signal at 2.1GHz. This could result in a poor quality voice call even if the handset was showing a perfectly good LTE signal immediately prior to making the call.
- A quirk of deploying LTE in one of its lower frequency bands is that it can improve the indoor coverage of mobile data services, which are often already well catered for but not voice services.

Although SRVCC involves the transfer of a live call from LTE network it places significant demands on the network in order to achieve reliability, minimal voice interruption and consistent call quality. With Single Radio Voice Call Continuity (SRVCC) it only maintains voice calls as mobile users move from LTE to non-LTE coverage areas but it does not initiate a call. After considering all these situations the company opted to use CSFB for 3G-4G and vice versa though SRVCC is a long term plan to maintain and to initiate voice calls in LTE

5.1 4G-4G (LTE-LTE)

Currently as it stands in Zimbabwe the 4G-4G tests are the ones which are currently being tested right now and the statistics at the moment are not yet approved to be disclosed but what we can simply say is that just managed to come up with a possible solution of just using the CSFB to make and to receive a call in the LTE network.

5.2 Future Recommendations

Mobile companies can create a sustainable business in which they have to implement network sharing and out sourcing. For mobile operators to ensure a long term profitable, they have to mainly focus on reducing network cost. The network and IT expenditure are believed to constitute 75% of the capital expenditure and 45% of the operational cost per annum of a mature mobile operator [23]. This implies that site sharing would reduce the overall cost by 30% and network sharing can reduce costs by targeting overcapacity in low-traffic areas

Network operators can also avoid the disadvantages of CSFB by delivering voice calls over their LTE infrastructure using Voice over LTE (VoLTE) technology. VoLTE has a number of benefits over CSFB, including faster call connection (less than half a second), greater spectral efficiency, high definition (HD) wideband voice quality. VoLTE services have other features that can help mobile network operators to differentiate them from “over the top” voice over IP services. . The quality of over the top services can be affected by IP traffic loads in both networks and devices, whereas VoLTE is protected by prioritisation. VoLTE also benefits from global interoperability of services and being part of the standard telephone numbering system. VoLTE requires network operators to invest in an IP Multimedia Subsystem (IMS) supporting Mobile Multimedia Telephony and to undertake upgrades to the LTE radio access network and core network which will make it much easier to integrate with other systems. All IP network converges the native power of IP to deliver enhanced capacity, value-added services and interoperability across network access methods and operators.

5.3 Potential Benefits of Deploying LTE all over Zimbabwe

With the introduction of smart meters and smart grid sensors it demands cost effective and easily deployment which implies that LTE would offer low latency, high throughput and the Quality of service differentiation in a single access technology[24]. In Zimbabwe they might be a broad ecosystem where many players will be involved which includes handset suppliers, software developers, and infrastructure vendors if a proper channel is employed. Also they will be vast posts to be advertised which will result in increased employment rate. Also the cost of maintenance of the network will eventually come down as one Base Transceiver Station (BTS) can cover a very long distance. This can give an opportunity of infrastructure sharing will can benefit all parties involved in network maintenance. LTE will meet customer expectations and the demand for speed and capacity enabling more data demanding and latency sensitivity applications such as interactive TV [25].

References

[1] "http://www.cleardoubts.com/technology/what-is-1g-or-first-generation-of-wireless-telecommunication-technology/".

[2] "Tinnirello Ilenia, Bianchi Giuseppe. Wireless Networks. Wireless Networks. [Online] University of Parlemo, 2005. [Cited: 14 february 2015.] http://www.tti.unipa.it/~ilenia/course/00-intro.pdf".

[3] A. Pashtan, "http://www.awarenetworks.com/publications/6.108.10.CellularTelephony.pdf. http://www.awarenetworks.com/publications/6.108.10.CellularTelephony.pdf. [Online] 2014. [Cited: 20 january 2015.] http://www.awarenetworks.com/publications/6.108.10.CellularTelep".

[4] in *2G and 3G cellular networks:*, 2008.

[5] "http://www.mobilecomms-technology.com/projects/gsm/".

[6] K. H. A. Mohammed A., "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.".

[7] H. Vishwanath, "Mobile Handset Cellular Network," 2011. [Online]. Available: https://www.google.co.zw/url?sa=t&rct=j&q=&esrc=s&source=web&cd=1&cad=rja&uact=8&ved=0CCoQFjAA&url=http%3A%2F%2Fwww.cse.ohio-state.edu%2F~xuan%2Fcourses%2F694%2F694_CELLULAR_NETWORK.ppt&ei=WLJTU".

[8] "http://www.mobilecomms-technology.com/projects/gsm/".

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

[10] http://www.engineersgarage.com/articles/what-is-3g-technology-specifications.

[11] 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&lan.

[12] http://www.techopedia.com/definition/2920/forth-generation-wireless-4g.

[13] K. A. K. S. Dinan Esmael, "Dinan Esmael, Kurochkin Aleksey, Kettani Sam.".

[14] "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.".

[15] "3GPP. Universal Mobile Telecommunications System (UMTS) RRC Protocol Specification. Sophia Antipolis France : ETSI, 2010. TS 25.331 Version 9.4.0 Release.".

[16] "http://www.lightreading.com/document.asp?doc_id=692828".

[17] "3GPP. Circuit Switched Fallback in Evolved Packet System. Sophia Antapolis : ETSI, 2009. TS 23.272 V8.4.0".

[18] "http://www.radio-electronics.com/info/cellulartelecomms/lte-long-term-evolution/srvcc-single-radio-voice-call-continuity.php".

[19] "http://lteworld.org/blog/evolution-single-radio-voice-call-continuity-srvcc".

[20] "Digital cellular telecommunications system ,UMTS VCC Between CS and IMS systems.".

[21] "Digital cellular telecommunications system ,UMTS VCC Between CS and IMS systems".

[22] G.Giannattasio, A Guide to the Wireless Engineering Body of Knowledge, 2009.

[23] T. McEwan, creating-sustainable-business-for-mobile-operators-through-network-sharing-and-outsourcing.

[24] ericson, "lte-for-utilities," *http://www.ericsson.com/res/docs/whitepapers/wp-lte-for-utilities.pdf.*

[25] ericsson, *http://www.ericsson.com/thecompany/press/mediakits/lte/information.*

[26] A. A. L. L. N. S. N. V. Kaaranen Heikki, UMTS Networks., 2001.

[27] A. A. L. L. N. S. N. V. Kaaranen Heikki, UMTS Networks., 2001.

[28] A. A. L. L. N. S. N. V. Kaaranen Heikki, UMTS Networks, 2001.

[29] S. A. France, Universal Mobile Telecommunications System (UMTS) RRC Protocol Specification.

[30] 3. U. M. T. S. (. R. P. Specification., Sophia Antipolis France.

APPENDIX

	A	B	C	D	E	F
1	Scenario	MO	Parameters Name	Scenario Value	Unit	Resolution
2	LTE to 3G Session Continuity	ReportConfigEUtraBadCovPrim	a2ThresholdRsrpPrim	-118	dBm	1
3	LTE to 3G Session Continuity	ReportConfigEUtraBadCovPrim	triggerQuantityA2Prim	0 (RSRP)		
4	LTE to 3G Session Continuity	ReportConfigEUtraBadCovPrim	hysteresisA2Prim	20	0.1 dB	5
5	LTE to 3G Session Continuity	ReportConfigEUtraBadCovPrim	reportAmountA2Prim	0		
6	LTE to 3G Session Continuity	ReportConfigEUtraBadCovPrim	timeToTriggerA2Prim	640	ms	0,40,64,8 0,100,128 ,160,256, 320,480,5
7						
8						
9	*FAJ 121 0493 , WCDMA Session Continuity, Coverage-Triggered* should be licensed and activated					
10						
11						
12	Scenario LTE to 3G Session Conti	UE detects bad coverage on Serving cell via event A2 when RSRP(LTE)+hysteresisA2Prim<a2ThresholdRSRPPrimary				
13						
14		RSRP(LTE)<-118-2=-120dBm				
15		time to trigger A2 :timeToTriggerA2Prim =640ms then the UE stats sending Meas Reports				
16		In case the UE detects RSRP(LTE) - hysteresisA2Prim>a2ThresholdRSRPPrimary before the timer 640ms is out :				
17		Then RSRP(LTE) >-118+2=-116dBm then the UE Leaves Event A2 and stays connected on the serving LTE cell				
18						

A	B	C	D	E	F
Scenario	MO	Parameters Name	Scenario Value	Unit	Resolution
LTE to 3G Cell Reselection	EUtranFreqRelation	cellReselectionPriority	7		
LTE to 3G Cell Reselection	EUtranCellFDD	systemInformationBlock3_sNonIntraSearch	16	dB	2
LTE to 3G Cell Reselection	EUtranCellFDD	threshServingLow	6	dB	
LTE to 3G Cell Reselection	EUtranCellFDD	systemInformationBlock6_tReselectionUtra	2	s	
LTE to 3G Cell Reselection	EUtranCellFDD	systemInformationBlock1_qRxLevMIn	-128	dBm	2
LTE to 3G Cell Reselection	UtranFreqRelation	threshXLow	3		
LTE to 3G Cell Reselection	UtranFreqRelation	qRxLevMin	-105	dBm	2
LTE to 3G Cell Reselection	UtranFreqRelation	cellReselectionPriority	6		
Scenario LTE to 3G Cell Reselection	Set WCDMA frequency lower than LTE priority as set in the table above				
	start measuring on UTRAN Freq when RSRP-qRxLevMin< sNonIntraSearch	RSRP<-128+16<-112			
	UE reselect from LTE to 3G for **two conditions**				
	RSRP-qRxLevMin<threshServingLow	RSRP<-128+ 6<-122	to keep more the LTE device in LTE in idle mode		

Result : 3G Cell KPI voice (1hr)HRE(03132014 0122)
Save Time 03/04/2015

User Name Tatenda kwava

Total 4,392 Records

Period (Min	NE Name	BSC6900U	AMR RAB	VP RAB S	CS Call Drop Rate ('	VP Call Dr	Soft Handover Success	CS 3G to 2G Handover	Cell Service Erlang (hour) (Erl)
60	HRE_BSC	Label=3G_	-	-	-	-	100	-	0
60	HRE_BSC	Label=3G_	-	-	-	-	-	-	0
60	HRE_BSC	Label=3G_	-	-	-	-	100	-	0
60	HRE_BSC	Label=3G_	100	-	0	-	99.576	-	0.067
60	HRE_BSC	Label=3G_	100	-	0	-	100	100	0.008
60	HRE_BSC	Label=3G_	100	-	0	-	98.913	100	0.067
60	HRE_BSC	Label=3G_	100	-	0	-	100	100	0
60	HRE_BSC	Label=3G_	100	-	0	-	100	-	0.1
60	HRE_BSC	Label=3G_	100	-	0	-	100	-	0.867
60	HRE_BSC	Label=3G_	-	-	-	-	100	-	0
60	HRE_BSC	Label=3G_	100	-	0	-	100	100	0.55
60	HRE_BSC	Label=3G_	100	-	-	-	100	-	0.025
60	HRE_BSC	Label=3G_	-	-	-	-	100	-	0
60	HRE_BSC	Label=3G_	-	-	-	-	100	-	0
60	HRE_BSC	Label=3G_	100	-	0	-	100	100	0.633
60	HRE_BSC	Label=3G_	100	-	0	-	100	100	0.008
60	HRE_BSC	Label=3G_	100	-	0	-	100	100	0.017

NE Name	BSC6900U	PS RAB S	HSDPA R/	HSUPA R/	PS Servic	Soft Handover Suc	Voice Call continuitu	SRVCC	PS DonwL	PS UpLink
HRE_BSC	Label=3G_	92.939	92.939	92.34	1.605	99.98	43.808	9.693	0.303	1.745
HRE_BSC	Label=3G_	100	100	100	2.532	100	2.219	0.23	0.006	0.813
HRE_BSC	Label=3G_	99.526	99.526	98.684	0	100	3.567	0.298	0	0.978
HRE_BSC	Label=3G_	99.828	99.823	99.808	1.067	99.956	645.106	54.459	20.187	7.526
HRE_BSC	Label=3G_	99.747	99.746	99.927	0	99.982	47.556	34.844	2.118	0.988
HRE_BSC	Label=3G_	99.549	99.51	99.432	2.36	99.973	208.03	147.125	20.467	6.089
HRE_BSC	Label=3G_	99.365	99.261	99.044	0.921	99.943	208.206	18.68	15.917	13.081
HRE_BSC	Label=3G_	99.922	99.92	100	0.314	99.975	58.761	9.755	0.739	5.02
HRE_BSC	Label=3G_	99.835	99.82	99.848	0.433	99.953	428.25	84.851	6.332	5.043
HRE_BSC	Label=3G_	99.932	99.919	100	0.136	99.966	12.582	3.204	5.755	2.451
HRE_BSC	Label=3G_	99.785	99.763	99.783	0.768	99.976	264.76	42.606	2.719	13.658
HRE_BSC	Label=3G_	99.549	99.486	99.829	0.657	99.956	69.789	8.267	4.807	7.836
HRE_BSC	Label=3G_	99.642	99.598	99.416	0.912	99.941	21.721	6.359	4.351	2.179
HRE_BSC	Label=3G_	99.753	99.74	99.757	1.188	99.889	65.571	16.233	13.698	8.817
HRE_BSC	Label=3G_	99.854	99.839	99.95	0.367	99.95	41.33	17.718	8.186	7.825
HRE_BSC	Label=3G_	99.771	99.784	99.754	0.232	99.957	120.329	15.499	5.654	15.998
HRE_BSC	Label=3G_	99.968	99.966	99.943	0.317	99.922	41.996	6.223	2.251	3.265
HRE_BSC	Label=3G_	99.817	99.809	99.901	0.673	99.949	288.52	16.897	2.998	9.01
HRE_BSC	Label=3G_	97.892	97.892	97.778	0	100	0.895	0.591	0.002	0.054
HRE_BSC	Label=3G_	99.922	99.922	99.92	0.465	100	40.437	3.855	0.006	3.444
HRE_BSC	Label=3G_	100	100	100	0	100	7.997	0.777	0.002	0.024
HRE_BSC	Label=3G_	100	100	100	1.093	100	2.975	0.119	0.175	0.462
HRE_BSC	Label=3G_	100	100	100	0	100	0.766	0.109	0	0.01

PS RAB S	HSDPA R/	HSUPA R/	PS Service	Call drops	Radio Resource Contr	Call Succe	PS DonwL	PS UpLink
92.939	92.939	92.34	1.605	99.98	43.808	96	0.303	1.745
100	100	100	2.532	100	2.219	97	0.006	0.813
99.526	99.526	98.684	0	100	3.567	100	0	0.978
99.828	99.823	99.808	1.067	99.956	645.106	95	20.187	7.526
99.747	99.746	99.927	0	99.982	47.556	98	2.118	0.988
99.549	99.51	99.432	2.36	99.973	208.03	94	20.467	6.089
99.365	99.261	99.044	0.921	99.943	208.206	97	15.917	13.081
99.922	99.92	100	0.314	99.975	58.761	95	0.739	5.02
99.835	99.82	99.848	0.433	99.953	428.25	100	6.332	5.043
99.932	99.919	100	0.136	99.966	12.582	100	5.755	2.451
99.785	99.763	99.783	0.768	99.976	264.76	99	2.719	13.658
99.549	99.486	99.829	0.657	99.956	69.789	92	4.807	7.836
99.642	99.598	99.416	0.912	99.941	21.721	90	4.351	2.179
99.753	99.74	99.757	1.188	99.889	65.571	89	13.698	8.817
99.854	99.839	99.95	0.367	99.95	41.33	98	8.186	7.825
99.771	99.784	99.754	0.232	99.957	120.329	94	5.654	15.998
99.968	99.966	99.943	0.317	99.922	41.996	93	2.251	3.265
99.817	99.809	99.901	0.673	99.949	288.52	99	2.998	9.01
97.892	97.892	97.778	0	100	0.895	99	0.002	0.054
99.922	99.922	99.92	0.465	100	40.437	100	0.006	3.444
100	100	100	0	100	7.997	89	0.002	0.024
100	100	100	1.093	100	2.975	87	0.175	0.462
100	100	100	0	100	0.766	90	0	0.01