"Performance Analysis of an LTE-4G Network Running Multimedia Applications"

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Abstract - In this thesis, we are interested in performances of LTE-4G Network running multimedia applications based on some QoS parameters. Our research was conducted using OPNET Network Simulator. We study the behavior of VoIP and Video Conferencing applications over LTE at static position of the nodes and when the nodes are moving at speed of 30 m/s using a standard Random Waypoint Mobility Model from OPNET.

An increase in the demand for very low latency and QoS satisfaction of the current bandwidth greedy multimedia applications over cellular and mobile devices is on the rise, which brought about the latest step in the UMTS family to develop LTE by the 3GPP. LTE is a new wave of frequency for the current 4G network that is an All-IP based radio frequency. Meaning that, all applications contending the network have to share the same narrow band IP-based network.

Keywords: LTE, QoS, GBR, NGBR, VoIP, Video Conferencing

1.1 INTRODUCTION

Mobile phones and smart devices are continuously evolving, seemingly at an accelerating rate of innovation and adoption from first generation (1G) to the current fourth generation (4G) network. The earlier generations of mobile technologies were only meant to guarantee QoS for voice communications but the progressive improvement in the functionalities of mobile devices with the recent 4^{th} generation (4G) wireless technology aims at providing a high quality video as well as voice communications that can co-exist over an all IP network [1] [2].

In 4G network today, WIMAX and LTE are both contending as the potential candidate's platform for 4G. Moreover, since LTE has evolved from existing 3G systems, it has been widely accepted by many service providers for 4G deployment.

The major aim of developing LTE by the 3GPP is to guarantee QoS for real-time multimedia applications, which have zero tolerance for delay. According to 3GPP, LTE is the evolution of the third generation of mobile communications, UMTS. The intention of LTE is to create a new radio-access technology, which will provide high data rate, low latency and a greater spectral efficiency most especially for multimedia applications over the newly innovated smart devices. To address these needs, the 3GPP has defined a Multimedia Broadcast/Multicast service, which extends the existing architecture by the introduction of MBMS bearer service as well as MBMS user service. Further release by the 3GPP specifically in release 8 specified a more advanced service called Enhanced MBMS (EMBMS) service. This provides higher frequency efficiency and more reliable point-tomultipoint transmission for LTE network [4] [6].

In our Scenario Two, we have conducted a related simulation to scenario one for Video Conferencing application using the same mobility model. We found out that Video Conferencing users experienced the same degree of QoS performances just as that of VoIP application in Scenario One. We found out that the Packet Delay Variation of Video Conferencing users at speed 30 m/s is far less than those at static nodes owing to the same condition of HARQ retransmission given-up we experienced in Scenario One.

In our further Scenario, we have conducted a simulation on the user perceived QoS when LTE network is under varying loads in Scenario Three. We have conducted a QoS performance analysis in the network scenario when VoIP and Video Conferencing users are running on LTE network alone and when they share the narrow band all-IP network with background traffic by introducing Ftp and Http applications. Our results show that different level of loads caused by best effort traffic have varying effects on the overall QoS performance of VoIP and Video Conferencing users in LTE network.

1.2 LTE: An Introduction

The advent of cellular devices has a great influence in our lives and has impacted nearly all public and private service sectors. Various evolutions have been witnessed in advancing series of mobile telecommunication system. One of the recent is the Long Term Evolution LTE of the UMTS [2].

LTE is the access part of the Evolved Packet System (EPS). It is a wireless broadband designed to support roaming internet access via cellular and handheld devices. [7]. The development of LTE was due to the huge increase in the

number of smart devices users and an increase in the number of low latency tolerance applications like online gaming, video-conferencing, video streaming and VoIP that has zero tolerance for jilter or delay. This rapid increase had lead the 3GPP to work on the LTE on the way towards 4th Generation network (4G).

The 3GPP in their release 8 improved on its standardization work on initial Evolved Packet System (EPS) by introducing Evolved Packet Core EPC and Evolved UMTS Terrestrial Radio Access Network (RAN) [8].

The invention of Internet Protocol (IP) as the key protocol to transport all services was developed by 3GPP during the design of 4G network. As a result of this new innovation, voice would have to be carried by IP instead of initial CS domain which pose a threat on the architecture of the system [7].

The major objective of LTE as mentioned in the release 8 of 3GPP is to guarantee users with significantly increased and instantaneously peak data rate of 100Mbps on the downlink and 50Mbps on the uplink in a 20MHZ channel as illustrated in Table compared with release 6. This will cause an improvement in user's throughput for about a factor of 3 and 2 for the downlink and uplink respectively.

LTE Network Architecture

The aim behind the development of LTE is to design a network architecture that will be based only on packet switching services in contrast to the earlier generation circuit-switching network. With the packet switching advantages of LTE, it will offer a seamless internet protocol IP between the User Equipment UE and PDN without any interference or disruption to users even during mobility [2] [9].

According to [10], all the network interfaces of LTE are based on IP protocol. The All-IP protocol is an evolution of the 3GPP system to fulfill the increasing demands for speed of the cellular communication devices. The implementation of all IP network protocol offers convenient access system for various vendors and networks with provisions for reduced system latency and user guarantee satisfaction [9]. The relationship between the UE, E-UTRAN and the EPC is depicted with Figure 2.1 while Figure 2.2 gives brother view of the relationship and the components found in them.

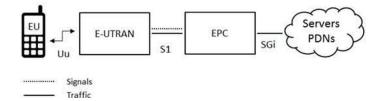


Figure 2.1: LTE High Level Network Architecture

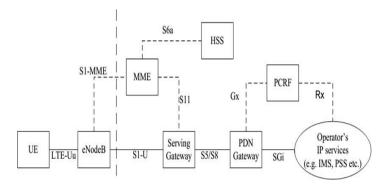


Figure 2.2: The EPS Network Architecture

The E-UTRAN

This is responsible for the handlings of the radio communications between the UE and the EPC. E-UTRAN consists of one single entity known as the eNodeB, which is the base station that connect and control the activities of the UE with the EPC. Figure4 depicts the fundamental structure of the access network of LTE: Evolved- UTRAN. E-UTRAN comprises of eNodeBs, which are interconnected with each other through an interface known as X2. The E-UTRAN also comprises of MMEs which are connected to the eNodeBs through the S1 gateway as illustrated in Figure 2.3 [12] [2].

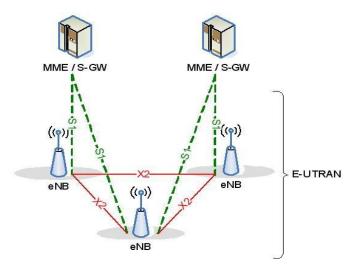
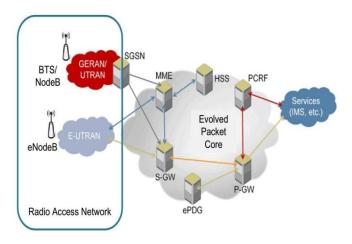
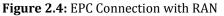


Figure 2.3: Evolved-UTRAN (E-UTRAN) Inter connection

The Evolved Packet Core EPC

EPC also known as Core Network (CN) is responsible for the overall control of the UE and establishment bearers. It comprises of three logical nodes: P-GW, S-GW and MME and some other supporting nodes which includes HSS, PCRF [2] [13] [14]. Figure 2.4 depicts the structure of EPC and its connection to the RAN. I will briefly highlight the functions and components of these three logical nodes but you can refer to [2] for details and broader understanding.





LTE Radio Protocol Architecture Development

In this section, I will shed light on the protocol stacks of LTE model user plane and control plane architectures. The creation of data packets and their processing by different protocols such as IP, TCP and UDP are carried out within the User plane and Control planes respectively.

Figure 2.5 represents a radio protocol architecture containing the user plane and the control plane. The data packets created at the application layer are being processed by IP, TCP and UDP protocol through the help of PDCP, MAC and RLC at the user plane. The control plane is where the signaling messages that are being exchanged between the based stations are processed and finally passed to the physical layer for transmission with the help of the RLC [11].

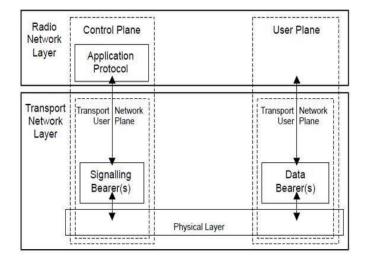


Figure 2.5: LTE Radio Protocol Architecture

Figure 2.6 gives an illustration of the architectural structure of LTE User Plane while Figure 2.7 represents the LTE control plane architecture. Both User and Control planes have similar features. I will briefly explain all these components in the next sub-section.

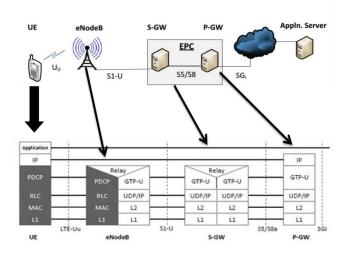


Figure 2.6: LTE User Plane Architecture

PDCP

The main functions of PDCP are header compression and decompression of IP data and processing.

RLC

RLC uses ARQ for data transfer in upper layer PDU and error corrections in AM only. In UM and AM, it also does concatenation, reassembling and segmentation of SDUs. There are three mode of operation of RLC. These are Transport Mode(TM), Acknowledged Mode (AM) and Unacknowledged Mode (UM) [14] [11]. The TM is a range between RLC SDUs to RLC PDUs. It is responsible for broadcasting system information and paging messages

control [14]. The AM gives bi-directional data whenever RLC transmits in both uplink and downlink mode. It is also responsible for giving acknowledgement in delay sensitive non-real time application such as web browsing. On the other hand, UM perform the distribution and continuity of RLC SDUs and reordering of RLC PDUs in real time traffics such as VoIP and MBMS [14].

RRC

This can only be found in the control plane. Its main function is to broadcast system information related to NAS, AS, paging, maintenance and establishment of RLC connection between eNodeB and UE. RRC also perform some security functions which includes key management, configuration, establishment, maintenance and releasing of P2P radio bearers [11].

1.3 QUALITY OF SERVICE OF LTE NETWORK

Providing the required end-to-end QoS for mobile devices is one of the challenges of wireless network. QoS service refers to the ability of network to deliver predictable and guarantee performance for the applications that are running over the network [18]. However, to guarantee the required QoS of various multimedia applications over wireless medium is very difficult. Multimedia applications like VoIP and Video are bandwidth greedy and can only tolerate very low latency in order to serve the end users better [14]. Because of this, different models and policies have been used in LTE to serve these applications better. Some of the policy is the used of Scheduling algorithm that prioritize applications over wireless network medium based on the Type of Service ToS. This algorithm used Quality Class Identifier (QCI) to classify and divide application using Traffic Forwarding Policy (TFP) among all the applications on the network.

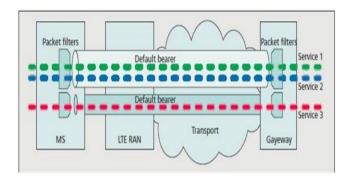


Figure 3.1: LTE QoS Framework showing Default and Dedicated Bearers

Standardized QoS Class Identifiers (QCIs) for LTE

In LTE network, Traffic Forwarding Policy TFP defined a set of standardized parameter between the UEs and the SGW. The allowed values range from 1-9. These values

are called standardized QCI. They are defined to reference specific QoS characteristics such as priority, packet delay budget and packet error rates between the UEs and GW [12] [14]. According to [12], the major goal for this set standard is in ensuring that applications along the LTE network received the right QoS services that guarantee users minimum requirements and that the applications receive the same minimum level of QoS in multi-vendor network deployments and in case of roaming.

The Table 3.1 below depicts the mapping of standardized QCI values to their standardized characteristics.

QCI	Resource type	Priority	Packet delay budget	Packet error loss rate	Example services
1		2	100 ms	10 ⁻²	Conversational voice
2		4	150 ms	10 ⁻³	Conversational video (live streaming)
3	GBR	3	50 ms	10 ⁻³	Real time gaming
4		5	300 ms	10 ⁻⁶	Non-Conversational video (buffered stream- ing)
5		1	100 ms	10 ⁻⁶	IMS signaling
6	Non-GBR	7	100 ms	10 ⁻³	Voice, Video (live streaming), Interactive gaming
7		6			Video (buffered streaming),
8		8	300 ms	10 ⁻⁶	TCP-based (e.g., www, e-mail, chat, ftp, p2p
9		9	500 113		file, sharing, progressive video, etc.)

1.4: Simulation Design and Implementation

This section describes the OPNET, a discrete event simulation framework for modeling the LTE as a fourth generation (4G) mobile broadband wireless technology. The OPNET simulator has a well-organized and detailed simulation environment for simulating LTE in partnership with OPNET LTE Consortium for application performance analysis and protocol design.

Opnet Simulator

OPNET, which stands for (Optimized Network Engineering Tool), is the de-facto standard for network R&D, modeling and simulation, defense organizations and network equipment's manufacturing. It is an important network simulator developer and solution provider for application and network management issues [23] [24].

According to [16] and [23], OPNET Modeler is an easy-touse application with a comprehensive developing features and graphical user interface that ease the development of design of real life scenario and simulating the network models.

In this study, OPNET modeler 17.5 is used for its reliability and efficiency in simulating both an object oriented and discrete event system (DES).

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The use of OPNET Modeler is adopted for this research study due to its flexibility in LTE Modeling and simulation, although, there are many standard network simulators that are readily available for an LTE study, for example, NS-2. The advantages illustrated below

Configuring the Network Model

OPNET 17.5 contains some standard tools that are readily available for editing and modification to simulate network problems. In this section, I will briefly highlight the network elements used in this study work.

Scenario One: Simulation of VoIP Application at varying speeds

We illustrate below an OPNET simulation environment for users of VoIP application over an LTE-4G network to investigate the performance of VoIP mobile users at specific mobility conditions.

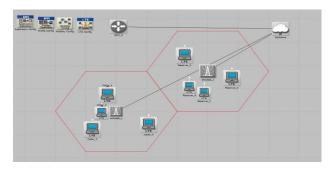


Figure 4.2: VoIP Application at varying speed network scenario setup

The application configuration is shown with the Figure 4.3 and 4.4 below where the VoIP application is configured using the parameters in the Table 4.2 below:

Тур	e: utility			
	Attribute	Value		
0	: name	Application Config		
	model	Application Config		
õ	-x position	-2,980		
Õ	- y position	3,876		
	- threshold	0.0		
õ	- icon name	util_app		
õ	- creation source	Object Palette		
õ	- creation timestamp	18:47:34 Apr 28 2014		
٢	- creation data			
	- label color	black		
٢	Application Definitions	()		
	■ MOS			
3	Voice Encoder Schemes	All Schemes		
	- hostname			
	- minimized icon	circle/#708090		
3	I. role			
M M C C	tended Attrs. Model Details Object D atch: Look in: Egact I Names Substring Values RegEx I Jacobile values I Tags	ocumentation 		

Figure 4.3: Application Configuration Attributes

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Figure 4.4: Application Configuration VoIP Attributes

Packet End to End Delay of VoIP users at Varying Speed.

The Table 4.3 below gives results of the average Packet E2E Delay Performance of VoIP users at speed 0 m/s and 30 m/s respectively.

Table 4.3 Average Packet End to End Delay of VoIP Users
at Varying Speeds.

Speed of the Mobile Users				0 m/s	30 m/s	
Packet variatio			End	Delay	0.7733	0.12644

We observed that the E2E performances of Static Nodes are higher than the mobile nodes and the speed of the mobile nodes have little or no effect on the average Packet E2E delays.

We went further to collate results from some selected nodes from the simulation in order to observe the variation of their E2E delay performances in relation to their speeds. The results obtained in second is tabulated in Table 4.4 and plotted in Figure 4.7 below.

Selected Nodes	Static	30 m/s	
Caller 1	0.4572	0.2551	
Receiver 1	0.8027	0.1287	
Caller 3	0.9225	0.1287	
Receiver 3	1.0508	0.1282	

VoIP Application Packet Loss Performance

As earlier stated in our previous chapter, the Packet Loss Performance (PLP) of users can be calculated from the following formula as

Number of Sent acket-Nu *100 Packet Loss **Received** Packet

Number of Sent Packet

Future Scope

This research has been conducted on OPNET network simulator, we recommend that other network simulation tools like NS2, OMNET++ , MATLAB Sim and LTE Sim, should also be used to study the QoS performance of multimedia applications in LTEnetwork.

Future Scope

Video Conferencing Performance under Light and **Heavy Loads**

In this section, we are interested in analyzing the performances of Video Conferencing under Light and Heavy Load, just as we have done in our early section for VoIP application. Here, we will study Video Conferencing application.

CONCLUSION

In this research work, we have conducted an investigation on the QoS performance of two vital multimedia applications (VoIP and Video Conferencing) over LTE-4G network. LTE, a new standard for wireless communication, has so many promises in terms of speed and performance metric of multimedia applications users.

We set up four different scenarios using Opnet 17.5 network simulator to investigate the QoS performances of VoIP and Video Conferencing based on the effect of mobility of end users on their QoS and based on the effect of loads created by best efforts (Ftp and Http) applications while sharing the narrow networks with the low latency and jittery intolerance multimedia applications.

We further conducted a research and found out that the LTE traffic class specifications of the 3GPP also contributed to the QoS performance of multimedia application. In our simulation, we discovered that VoIP application give very poor MOS and PLP results when allowed to share a default channel with background traffic created by Ftp and Http. But, when allocated to the best prioritized GBR channel based on ARP and ToS specification, the performance was guaranteed for VoIP user satisfaction. In addition, in the case of Video Conferencing, the QoS performance in both shared channel and prioritized channel has very little effect. This implies that Video Conferencing respond better than VoIP application, when they are allowed to share defaults LTE channel.

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