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

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ABSTRACT

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.

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

We went further in our study to investigate how these multimedia applications respond to different degree of congestions most especially when NGBR applications like Ftp and Http are contending the channel with the GBR bearers. We finally study the various class of ToS defined in the LTE network to see how this can aid the QoS performance of multimedia applications on the network.

Our results show that sometimes, mobile nodes give better packet end-to-end delay and Mean Opinion Score than the corresponding static nodes because of the HARQ retransmission given up in the mobile node case for VoIP application. Furthermore, our simulation showed that Video Conferencing application sometimes gives less delay at mobile nodes that static nodes because of accumulation of traffics at the static nodes that lead to poor end-to-end performance and high delay variation.

We also found out in our study that the present of best effort traffics in the narrow band LTE network caused VoIP and Video Conferencing applications to give low QoS performances while the LTE traffic classes' standard in 3GPP improve the QoS of these multimedia applications over LTE network.

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

ÖZ

Çok düşük gecikme ve hücresel ve mobil cihazlar üzerinden mevcut bant genişliği açgözlü multimedya uygulamaları QoS memnuniyeti için talep artışı 3GPP tarafından LTE geliştirmek için UMTS ailesinin üzerinde son adımı getirildi. LTE All-IP tabanlı radyo frekans akımı 4G ağ için frekansı yeni bir dalgadır.Ağ yarışma tüm uygulamaları aynı dar bant IP tabanlı ağ paylaşmak zorunda anlamına gelir.

Bu tezde, bazı QoS parametrelerine dayalı multimedya uygulamaları çalıştıran LTE-4G Ağı performansları ilgilendirir. Bizim araştırma OPNET Ağ Simülatörü kullanılarak gerçekleştirilmiştir. Biz düğüm statik pozisyonda LTE üzerinden VoIP ve Video Konferans uygulamaları davranışlarını incelemek ve düğümleri OPNET bir standart Rastgele Noktası Hareketlilik Modeli kullanılarak 30 m / s hızında hareket ettirmek için çalıima yaptık.

Biz bu multimedya uygulamaları FTP ve HTTP gibi NGBR uygulamalar GBR taşıyıcıları ile kanal yarışma çoğu özellikle tıkanıklığı farklı derecesi nasıl yanıt araştırmak için bizim çalışmamızda da ileri gitti. Biz nihayet bu ağda multimedya uygulamaları QoS performansını nasıl yardımcı olduğunu görmek için LTE ağı tanımlanan ToS çeşitli sınıfını çalışmasını yaptık

Bizim sonuçlarımız, mobil düğümler çünkü VoIP uygulaması için mobil düğüm durumda vazgeçmiş HARQ yeniden iletimin daha iyi paket uçtan uca gecikme ve ilgili statik düğümler daha Mean Opinion Score verdiklerini göstermektedir. Ayrıca, simülasyon Video Konferans uygulama bazen mobil düğümleri biraz gecikme

gösterdiğinden dolayı kötü uçtan uca performans ve yüksek gecikme varyasyon yol statik düğümlerde trafikleri birikimi statik düğümleri oluşmuştur.

Biz 3GPP LTE trafik sınıfların standart LTE ağı üzerinden bu multimedya uygulamaları QoS iyileştirmek için dar bant LTE ağında iyi çaba harcadık ki bugünkü düşük QoS performans vermek için VoIP ve Video Konferans uygulamalarını çalışmalarımızda öğrendim.

Anahtar kelimeleri: LTE, QoS, GBR, NGBR, VoIP, Video Konferans

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To my darling wife, Doris, my daughter, Ayomide and my son

Ayokunle

TABLE OF CONTENTS

LIST OF FIGURES

LIST OF TABLES

LIST OF ABBREVIATIONS

AGBR	Associated Guaranteed Bit Rate
AM	Acknowledged Mode
AMC	Adaptive Modulation and Coding
ARP	Allocation and Retention Priority
BCCH	Broadcast Control Channel
BCH	Broadcast Channel
CCCH	Common Control Channel
CDMA	Code Division Multiple Access
CN	Core Network
COMP	Coordinated Multi-point
DES	Discrete Event System
DCCH	Dedicated Control Channel
DFT	Discrete Fourier Transform
Diff-Serv	Differential Service
DL-SCH	Downlink Shared Channel
DTCH	Dedicated Traffic Channel
DTS	Data Transport Service
DWPTS	Downlink Pilot Time Slot
eMBMS	Enhanced Multimedia Broadcast Multimedia Service

EPC	Evolved Packet Core
EPS	Evolved Packet System
E-UTRAN	Evolved Universal Terrestrial Radio Access Network
FDD	Frequency Division Duplexing
FDM	Frequency Division Multiplex
FDMA	Frequency Division Multiple Access
GBR	Guaranteed Bit Rate
GP	Guard Period
GSM	Global System for Mobile
HARQ	Hybrid Automatic Repeat Request
HLR	Home Location Register
HSPA	High Speed Packet Access
HSS	Home Subscriber Service
HTTP	Hypertext Markup Language
IDFT	Inverse Discrete Fourier Transform
I P	Internet Protocol
IRC	Interference Rejection Combining
LTE	Long Term Evolution
MAC	Media Access Control
MBMS	Multimedia Broadcast Multimedia Service
MC	Multicast Channel

MCCH	Multicast Control Channel
MCS	Modulation and Coding Scheme
MIMO	Multiple Input Multiple Output
MME	Mobile Management Entity
MT	Mobile Terminating
MTCH	Multicast Traffic Channel
NAS	Non-Access Stratum
NGBR	Non-Guaranteed Bit Rate
NGMN	Next Generation Mobile Network
OFDMA	Orthogonal Frequency Division Multiple Access
PCCH	Paging Control Channel
PCH	Paging Channel
PCRF	Policy and Charging Rule Function
PDU	Protocol Data Unit
PDCP	Packet Data Convergence Protocol
PDN	Packet Data Network
P-GW	Packet Data Network Gateway
QCI	QoS Class Identifier
QoE	Quality of Experience
QoS	Quality of Service
RACH	Random Access Channel

RAN	Radio Access Network
RB	Resource Block
RRC	Radio Resource Control
SAE	System Architecture Evolution
SC-FDMA	Single Carrier Frequency Division Multiple Access
SDF	Service Data Flow
SDU	Service Data Unit
S-GW	Serving Gateway
SIM	Subscriber Identity Module
TDD	Time Division Duplexing
TDMA	Time Division Multiple Access
TFP	Traffic Forwarding Policy
ToS	Type of Service
UE	User Equipment
UDP	User Datagram Protocol
UICC	Universal Integrated Circuit Card
UMTS	Universal Mobile Telecommunication System
UP-SCH	Uplink Shared Channel
W-CDMA	Wideband Code Division Multiple Access
WIMAX	Worldwide Interoperability for Microwave Access
3GPP	Third Generation Partnership Project

Chapter 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 4th 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**].

The world has experienced great and tremendous innovation with the invention of smart devices and tablets built with the latest technologies that allow users to access various multimedia applications such as live video streaming, online gaming, voice over IP (VoIP), mobile TV and so on. These applications require high data rate and high bandwidth because to guarantee QoS to the end users, adequate provision of bandwidth and low latency network must be in place [**3**].

To meet these needs, the Third Generation Partnership Project (3GPP) commenced research on the best solution to offer guarantee and quality service to the end users [**4**] [**5**].

This brought about the introduction of HSPA that is currently using 3G. Recently, they introduce a LTE as a potential candidate for 4G network.

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-to-multipoint transmission for LTE network [**4**] [**6**].

Despite the available provisions made by the 3GPP, the greediness and aggressiveness of smart and mobile devices for bandwidth is on the high side. Meeting the required QoS of these devices and QoS requirements imposed by various services along the wireless medium call for holistic investigation. LTE use coordinated multi-point (COMP) transmission to provide access to multiple applications with different requirements of the quality of service (QoS). For example, video telephony, file transfer and web surfing require high-speed data rate whereas, voice over IP (VoIP) demands relatively low-rate applications though posed critical requirement on latency [**2**].

Sharing of single channel by real-time and non-real-time applications pose great challenge on the LTE network. How the sharing of the LTE resources is achieved using numbers of queuing disciplines which are widely deployed is investigated in this thesis and the performance analysis of an adaptive bandwidth provisioning and content aware algorithm used in the eNodeB will be simulated using OPNET simulator to improve in the latency and packet drop and retransmission. [**5**]

We have conducted a research using OPNET simulation tool to investigate the QoS performance of VoIP and Video Conferencing applications users over LTE-4G network. In our study, we have set-up four different scenarios to conduct our analysis. In the first scenario, we simulated VoIP users at static and 30 m/s speed using random waypoint mobility. The result shows that the Packet End-to-End delay performance, Mean Opinion Score and Packet Delay Variation of VoIP users at 30 m/s case give better performance than that of users at static nodes. This is because of the accumulation of traffics at the static nodes case since the nodes try to admit as many as possible traffics. While in the mobile nodes case, the HARQ retransmissions are given-up hence, the users experience high loss and less traffic successfully traverse from sending nodes to destination nodes.

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.

We finally conducted a simulation to study the 3GPP standard LTE traffic classes as stated in 3GPP release 7 and 8 in our Scenario Four. We set-up a simulation using the idea of dedicated and default channel QoS allocation system. In the first case, we simulated VoIP, Video Conferencing Http and Ftp to share a default channel with QCI-6 as Non-Guaranteed Bit Rate (NGBR) bearers. In the second case, we used the Allocation and Retention Priority and Quality of Service Class Identifier (QCI) to allocate each bearer based on their ARP as stated in the 3GPP QoS standard. We found out that, VoIP and Video Conferencing applications users give better QoS performances when they are allocated to their respective priorities than when they share a default channel with best efforts. More also, Video Conferencing can manage to give fair QoS performance while sharing the scenario with best efforts traffics, VoIP performances are totally unacceptable in this case.

1.1 Research Motivation

The LTE 4G network is a pure all IP-based network that replaces network switches with routers and computer. Unlike the 3G network that is based on packet and circuit switching, LTE 4G networks is based purely on packet switched network, which is mainly designed for high-speed data transfer across the network.

For LTE to offer the QoS require by real-time multimedia applications like voice over IP (VoIP), the architecture has to be redesigned in a top-down form where by priority will be given to real-time application contending the network channel over non-real-time packets. Each individual packet on the network medium must be identified with a Type of Service (ToS) flag, which will serve as content aware for the network along with the differential service (Diff-Serv) to achieve the required QoS of individual applications on the network. The Diff-Serv architecture which is commonly used in IP based network to classify the various ToS need to be integrated with the LTE QoS architecture to guarantee good end to end performance.

This research work tries to examine the performance of real-time application i.e. Voice over IP (VoIP) and Video conferencing over the LTE network. Investigation of the QoS performance opens more researches to create an adaptive measure to streamline the provisioning of bandwidth to various applications across the network in a content-aware mode and help to think of the better strategies to treat multimedia applications to give better service and QoS required by the end users.

1.2 Aim and Objective

Looking at the trends and innovations in smart and mobile devices and their QoS demands, many researchers have developed ideas using various scheduling algorithms and queuing disciplines to improve on the overall performances of user's applications over LTE network.

Since the aim of developing LTE by the 3GPP is to guarantee better QoE of users, proper prioritization of real and non-real time application must be guaranteed because of the demand posed by low tolerance multimedia applications that are currently running over the various smart phones and mobile devices. As these multimedia applications may be susceptible to delays (e.g. voice-over-IP), loss (e.g. video streaming), or both (e.g. video conferencing) proper quality of service (QoS) needs to be guaranteed.

The aim of this thesis is to investigate the performance of various multimedia applications over LTE. The design will involve the performance analysis of VoIP and Video Conferencing applications when they are allowed to share the narrow-band all-IP channel at varying network loads. Some of the cause of poor QoE of users, which is the effect of background traffic, will also be investigated by introducing an FTP application to share the same channel with the since LTE is an all IP network. This will help us to identify the effect of background traffic on the overall performance of real time application over LTE network.

Chapter 2

LTE BACKGROUND AND THEORETICAL KNOWLEDGE

2.1 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 EPC is the latest evolution of the 3GPP core network architecture. EPC is based on packet switching only. It is an all IP-network according to [**8**], which aids the end-to-end architecture for supporting smart and mobile devices networks.

In the past evolutions, GSM is based on Circuit Switching (CS), which establishes circuit between calling and called parties throughout the telecommunication network. Likewise, in GPRS, packet-switching (PS) was used to transport data as an addition to CS. Although, CS is used to transport voice and SMS in some cases.

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 2.1 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.

Other benefits that come with LTE are improved data rate at cell edge, improved spectral efficiency, scalable bandwidth, compatibility with other earlier releases and with other system, reduced delays in terms of connection establishment and transmission latency, seamless mobility including between different radio-access technologies and reasonable power consumption for the mobile terminal [**2**] [**7**].

Table 2.1: 3GPP Specification for 4G network

		Absolute requirement	Comparison to Release 6	Comment
Downlink	Peak transmission rate	>100Mbps	7 x 14.4 Mbps	LTE in 20 MHz FDD 2x2 spatial multiplexing.
	Peak spectral efficiency	>5 bps/Hz	3 bps/Hz	
	Average cell spectral efficiency	>1.6-2.1 bps/Hz/cell	3-4x0.53 bps/Hz/cell	LTE: 2 x 2 spectral multiplexing, Interference Rejection Combining (IRC) receiver
	Cell edge spectral efficiency	>0.04-0.06 bps/Hz/user	2-3 x 0.02 bps/Hz	As above, 10 users assumed per cell
	Broadcast spectral efficiency	>1 bps/Hz	N/A	Dedicated carrier for broadcast mode
Uplink	Peal transmission rate	>50 Mbps	5 x 11 Mbps	LTE in 20MHz FDD, single antenna transmission
	Peak spectral efficiency	>2.5 bps/Hz	2 bps/Hz	
	Average cell spectral efficiency	>0.66-1.0 bps/Hz/cell	2-3x 0.33 bps/Hz	LTE: single antenna transmission
	Cell edge spectral efficiency	>0.02-0.03 bps/Hz/user	2-3 x 0.01 bps/Hz	As above, 10 users assumed per cell
System	User plane latency (two radio delay)	<10 ms	One fifth	
	Connection set-up latency	<100 ms		-Idle state- active state
	Operating bandwidth	1.4-20MHz	5MHz	(initial requirement started at 1.25)
	VoIP capacity	NGMN preferred is > 60 Sessions/MHz/cell		

2.2 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**].

It also offer lower costs compared to earlier system. Although, some of the danger in the all IP network is its porosity and vulnerability to hackers , virus and all sort of intruders due to its openness as an all IP-network [**2**] [**6**] [**10**][[**9**].

The LTE comprises of the User Equipment **UE**, evolution of the radio access network Evolved-UTRAN (**E-UTRAN**) and the Non Radio Access counterpart knows as System Architecture Evolution (**SAE**), which comprises of the Evolved Packet Core (**EPC**) [**2**].

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.

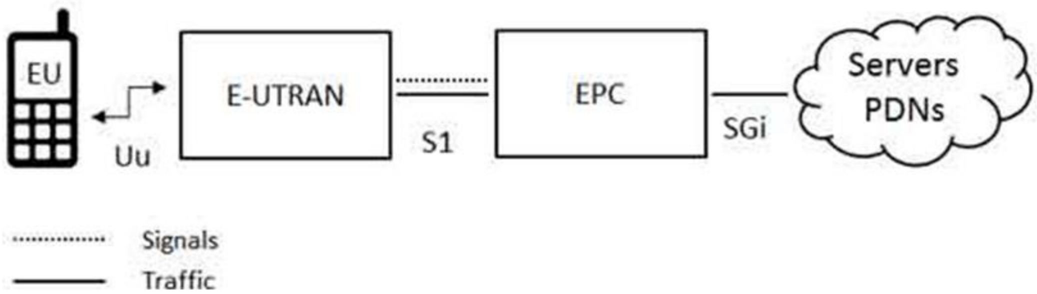

Figure 2.1: LTE High Level Network Architecture

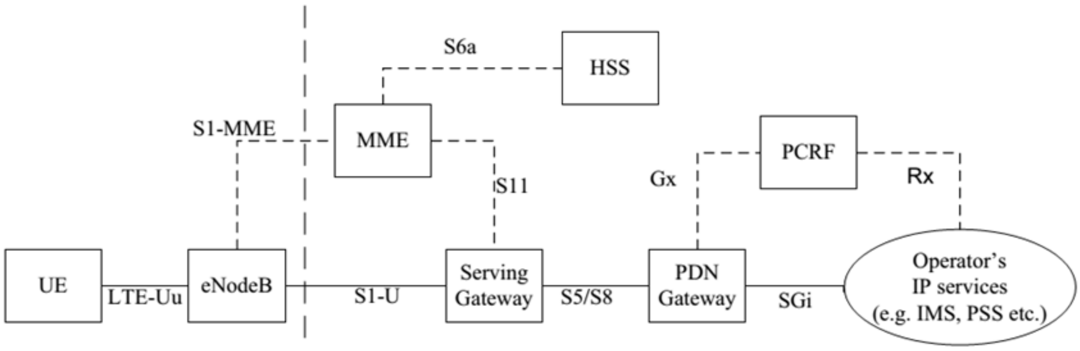

Figure 2.2: The EPS Network Architecture

2.2.1 The User Equipment UE

This is actually the Mobile Equipment (ME) [**11**] which could either be a hand-held cellular phone, a laptop equipped with a mobile broadband adapter or any such device that are used by the end-users for wireless communication. The relationship between the UE, the E-UTRAN and the EPC is as illustrated in the Figure2.1 and Figure 2.2. The architecture of the UE is similar to the one of UMTS and GSM and it comprises of MT that handles all the communication functions, the TE that terminates the data streams and the UICC also referred to as SIM that run an application known as USIM. USIM is used by the SIM part of the UE to store information of the user like the phone number, the home network identity and the security keys [**2**] [**11**].

2.2.2 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**].

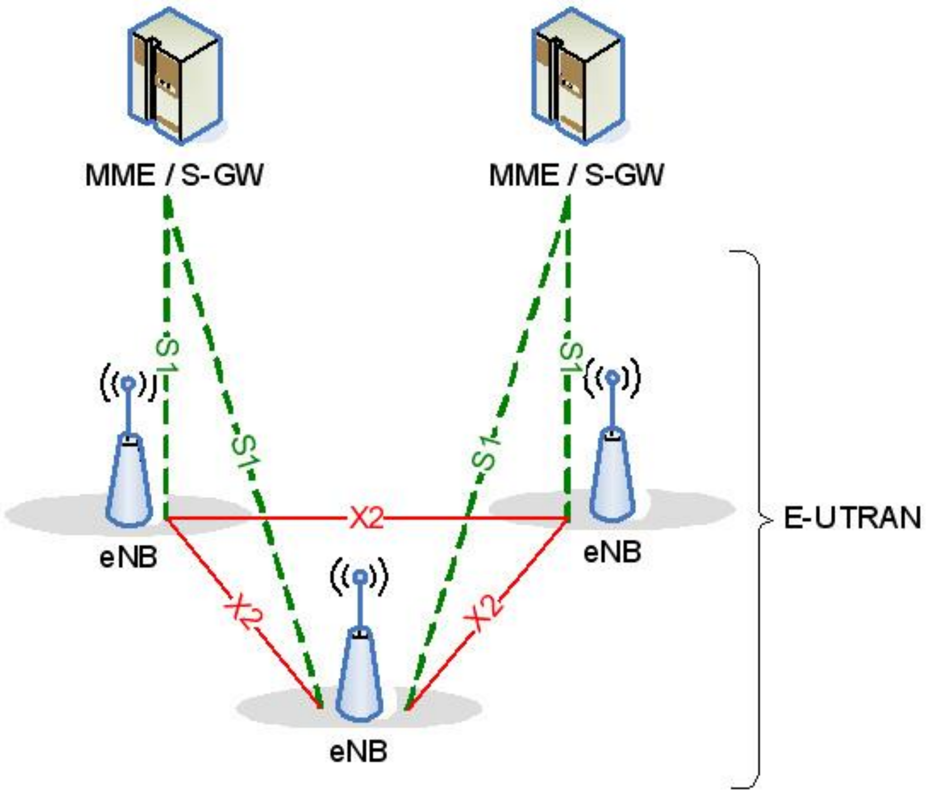

Figure 2.3: Evolved-UTRAN (E-UTRAN) interconnection

The sending and receiving of radio transmission packets between the UE and the EPC is coordinated by the eNodeB by using the analog and digital signal processing functions in its interface [**11**]. eNodeB also handles various UE low-level operations such as handover.

2.2.3 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.

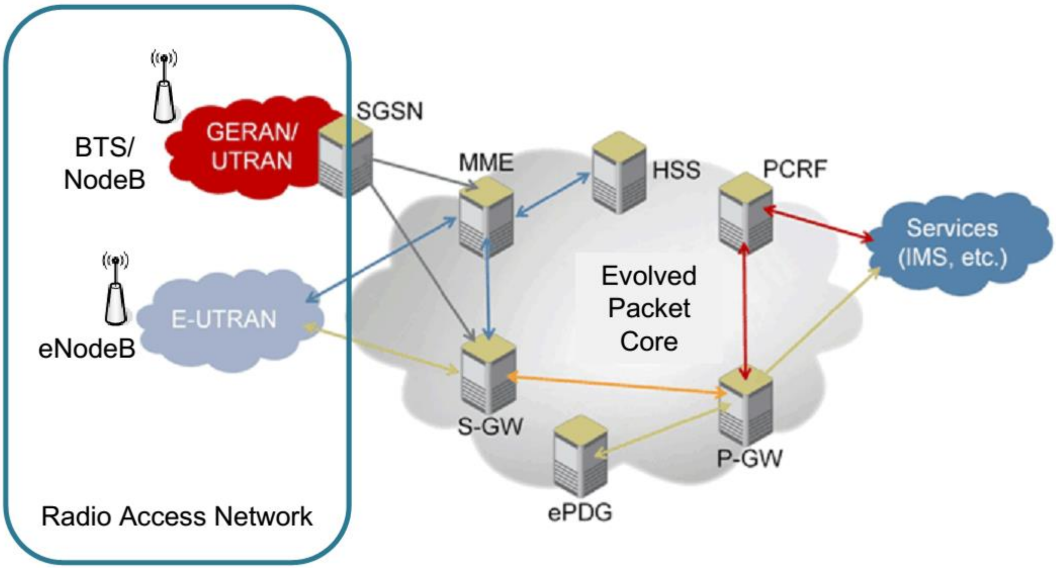

Figure 2.4: EPC Connection with RAN

2.2.3.1 PDN Gateway (P-GW)

P-GW communicates with the outside world using SGi interface. It play a major role of allocating IP addresses to various UEs and help the EPC to enforce the required QoS to individual traffics based on the report from the PCRF on the downlink side by IP filtering. [**2**] Due to the sensitivity of the GBR bearers, P-GW helps to enforce their required QoS and serve as bedrock for other non-3GPP technologies to co-exist with LTE [**2**] [**13**].

2.2.3.2 Serving Gateway (S-GW)

S-GW acts as a router by forwarding data between the RAN and the PDN gateway. It also serves as a mobility anchor for inter-working with other 3GPP GSM and UMTS [**13**]. It serves as the local mobility anchor when UE is idle. It helps to keep information about the volume of data sent or receives in form of charges at the visiting network. The MME uses the information kept by the S-GW to reestablish an idle UE during UE reestablishment at the downlink section [**2**].

2.2.3.3 Mobility Management Entity (MME)

MME uses signaling message and HSS to control NAS protocol, a high-level operation between the UE and the CN [**11**].

It is also responsible for NAS control, idle state, security, and EPS bearer control [**13**]. According to [**15**], MME also helps in tracking user location in form of security and various paging procedure.

Home Subscriber Support (HSS): HSS serves as the central database server for the LTE that contains identity and subscription related information for the home users. HSS perform the same function as the HLR in the 3G technologies [**15**].

Policy and Charging Rules Function (PCRL): PCRL perform three vital functions. These are policing, decision making for the EPC and control of the flow-based charging functionalities [**11**] [**15**]. PCRL is also responsible for the control of QoS of UE and reports to the P-GW and S-GW [**16**].

2.3 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**].

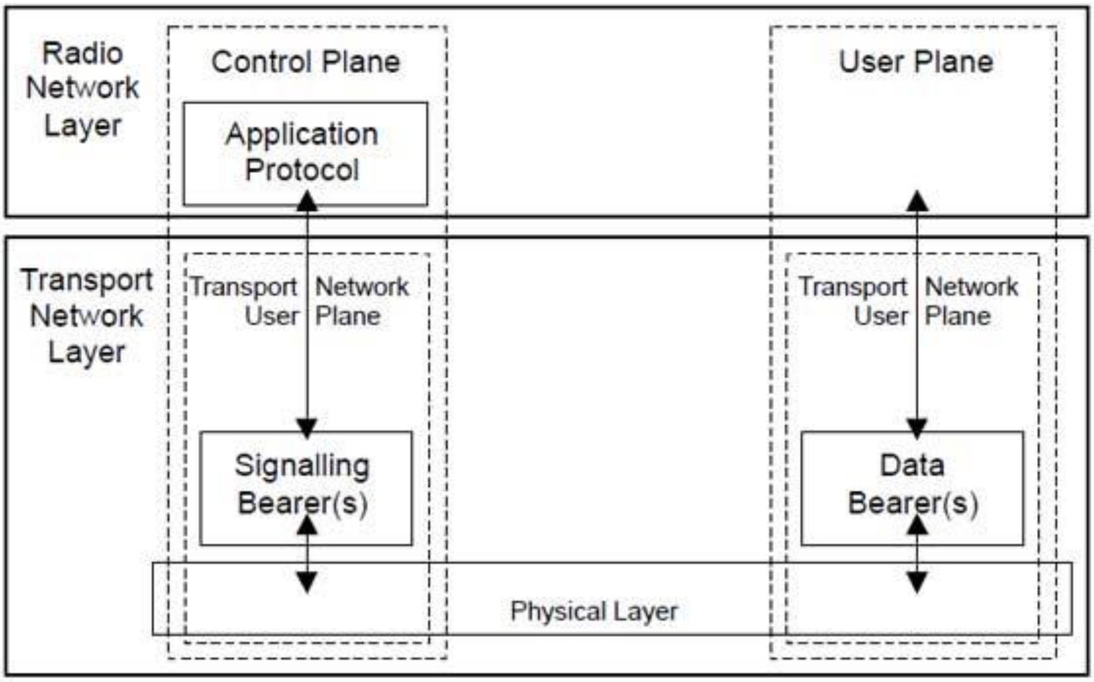

Figure 2.5: LTE Radio Protocol Architecture

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.

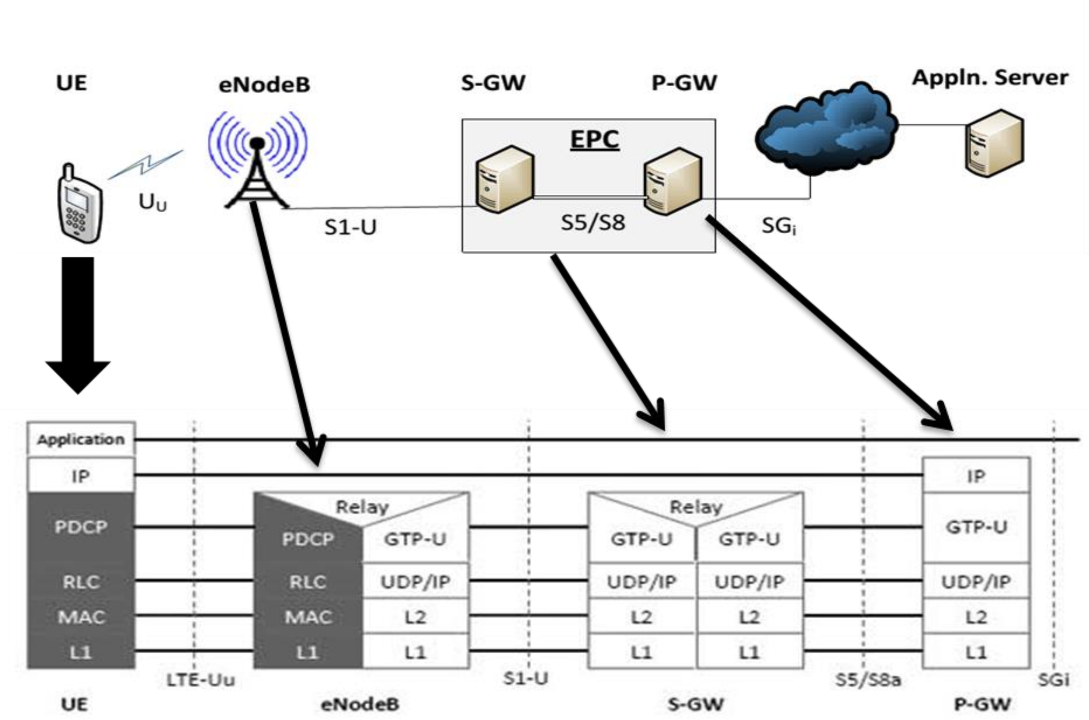

Figure 2.6: LTE User Plane Architecture

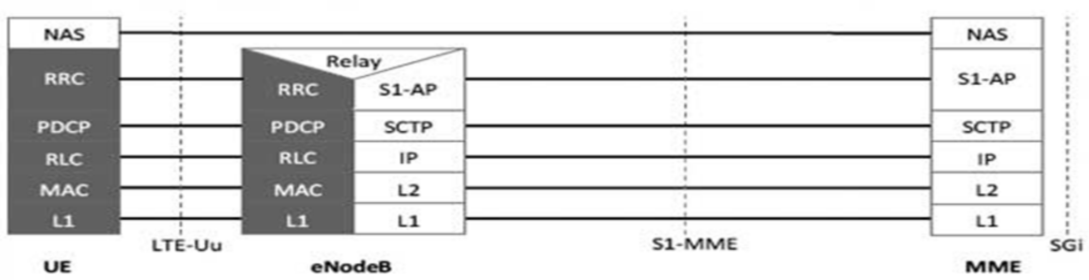

Figure 2.7: LTE Control Plane Architecture

2.3.1 PDCP

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

2.3.2 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 Un-acknowledged 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**].

2.3.3 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**].

2.3.4 NAS

This is also present in Control plane only [**14**]. It is a protocol than runs between MME and UE and helps in mobility management and bearer settings. It helps to establish IP connectivity between UE and PDN GW. It is the highest stratum of the control plane [**14**] [**11**].

2.3.5 MAC Layer

MAC is a vital layer in LTE model as part of the logical layer that is responsible for the mapping of information between the Logical and the Transport channels. It multiplex MAC PDUs to the physical layer and as well receives SDU from physical layer over transport channel and logical connection between RLC layer and Logical channel [**2**] [**14**] [**11**]. Some of the important functions of MAC layer according to [**11**] are error correction using HARQ, scheduling and priority handling of applications in the logical channel of each UE and prioritization of logical channel.

MAC belongs to the Layer 2 in the reference model.

2.3.6 Physical Layer

Physical layer uses the Data Transport Service DTS to carry information from the MAC transport channel over the air interface. It is the one responsible for the cell search initial synchronization and handover triggering. It also maintains the power control and link adaptation (AMC) together with transferring of dependable signal over radio access between the UE and eNodeB [**11**].

The Physical layer of LTE uses SC-FDMA for uplink and OFDM for downlink radio resources transmission between UEs and eNodeBs respectively. It also supports the use of MIMO for higher data rate at the downlink section [**2**] [**6**].

During transmission, the signals to be transmitted are divided into two frames (FDD and TDD.

The FDD frame structure supports both half-duplex and full duplex transmission with two-carrier frequencies domain, one for uplink and the other for downlink [**16**] as illustrated in Figure 2.8. FDD frames have 10ms period slots and each of the slots is 0.5ms. Each of the FDD radio frame contain 20 slots and a sub frame contains two slots out of the 20 slots in an FDD radio frame which amount to 10 sub-frame in LTE FDD frame structure [**6**] [**16**].

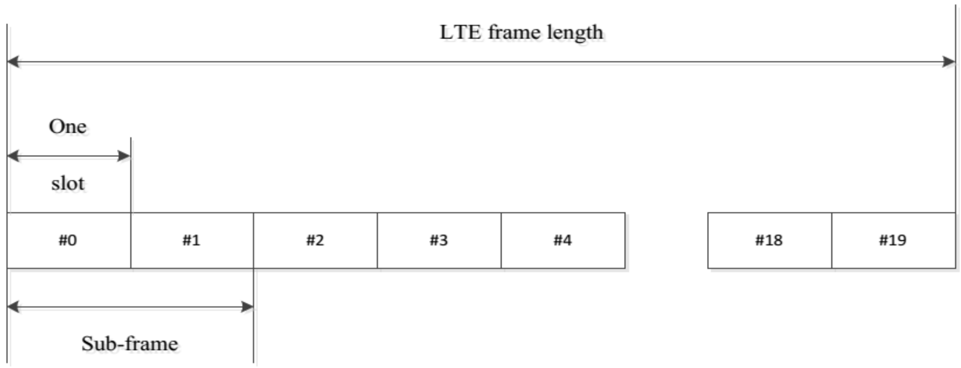

Figure 2.8: FDD LTE frame Structure

TDD on the other hand uses the same frequency bands, which are separated in time for uplink and downlink transmission [**6**]. TDD consists of three sub frames namely: Guard Period (GP), Downlink Pilot Time Slot (DWPTS) and Uplink Pilot Time Slot (UPTS), which both have a length of 1ms [**16**]. TDD frame structure is composed of two identical half-frame with duration of 5ms for each of them. Both of the half-frames are further sub-divided to another set of five sub-frames with the duration of 1ms. The sub-frame further splits to two slots of 0.5ms each. Figure 2.9 illustrates the TDD frame structure and sub-frames.

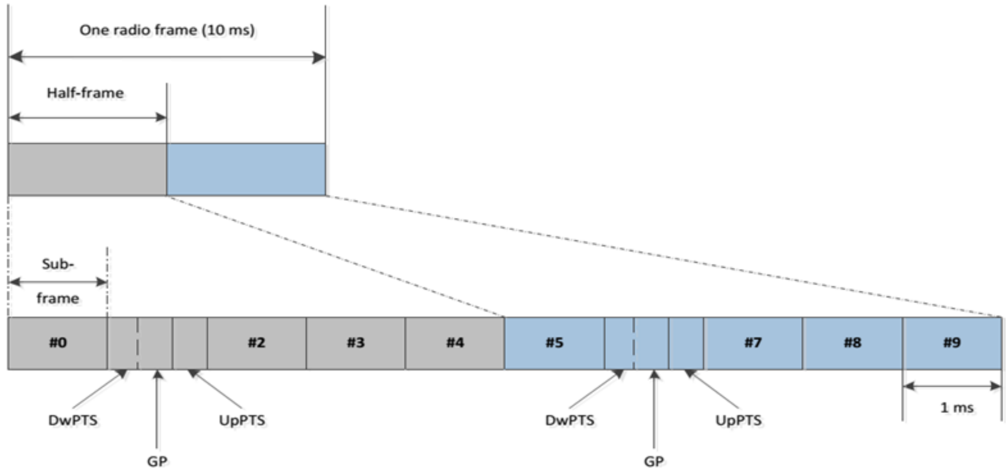

Figure 2.9: TDD LTE Frame Structure

2.3.7 Structure of Resource Block

This is what we use to represent transmission in LTE. It is the smallest time-frequency resource that can be allocated over the air in LTE. A unit of resource block (RB) consists of on sub-carrier over one OFDMA symbol [**6**]. During transmission in the both uplink and downlink, a scheduler located at the eNodeB allocates resource in the units of RB either in FDD or TDD mode, which has a bandwidth of 180 kHz over a single time slots of 0.5ms [**6**] [**16**]. Figure 2.10 shows the overall structure of LTE frame RB. RBs are structured in Time-Frequency grid fashion and contains of NSC in successive subcarriers in frequency domain and time slots in time domain as illustrated in Figure 2.10.

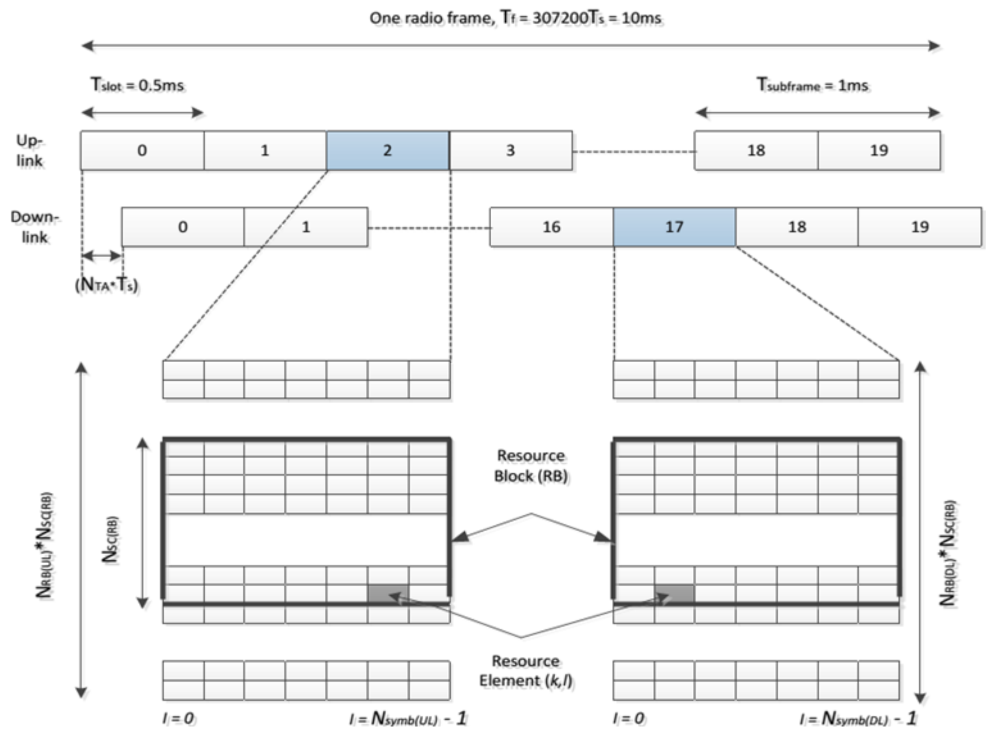

Figure 2.10: LTE Resource Block structure

2.4 Overview of the multiple access and modulation techniques used in LTE network

Multiple access transmission in LTE network is based on Frequency Domain Multiplexing (FDM). Basically, Single Carrier Frequency Division Multiple Access (SC_FDMA) is used for uplink transmission and Orthogonal Division Multiple Access (OFDMA) is used for downlink transmission respectively [**17**]. These multiple carrier techniques show some high level of advantages over the earlier techniques like CDMA. Some of these advantages are their robustness in communication and their stable interference management strategies. Multiple access techniques open room for exploiting multiuser diversity at smaller granularities than the earlier CDMA-based network. They use modulating orthogonal subcarriers to enhance frequency selective fading, which help to support different users during mobility at different communication conditions [**6**] [**17**]. I will also briefly talk on OFDM as a principle used in OFDMA.

2.4.1 OFDM

An advanced in access technologies that facilitates higher transmission rates with a reasonable equalization and detection complexities using some numbers of modulated narrowband orthogonal subcarriers. [**2**]. It is a method of digital modulation that splits a signal into several narrowband channels at different frequencies [**17**].

2.4.2 OFDMA

This modulation technique is used in the downlink transmission section of LTE network for accessing mobile broadband wireless system in 4G [**14**]. According to [**16**], OFDMA is an OFDM-based multiple access that combine the techniques of TDMA and FDMA for LTE downlink transmission. This is shown in Figure 2.13 below. It allocates fraction of system bandwidth to each users in each specific time slot and guarantees better spectral efficiencies and better resources scheduling based on the frequency responses and channel time [**6**] [**16**].

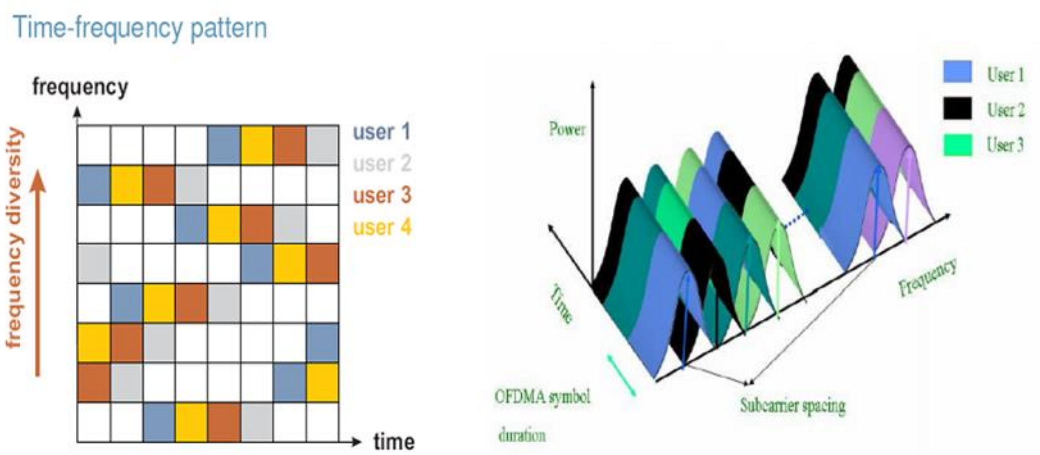

Figure 2.11: LTE OFDMA Basic Operations [**16**]

2.4.3 SC-FDMA

SC_FDMA is chosen as the multiple transmission technique for LTE uplink section. SC-FDMA is a very sophisticated modulation technique that combines various advantage of OFDM that includes the efficient frequency allocation, low peak to average power ratio with multiple path resistance for transmission at the uplink sections of LTE network [**17**]. One of the advantages of SC-FDMA is power management in the UE during uplink transmission [**6**]. Although, OFDMA has advantage of better utilization of the available narrow band over SC-FDMA according to [**2**] [**16**], SC-FDMA on the other hand is less sensitive to the channel frequency-selective fading than OFDMA due to its ability to spread each modulated symbol very efficiently across the total channel bandwidth [**16**] SC-FDMA other advantages is its low PAPR. Figure 2.14 shows the transceiver comparison of OFDMA and SC-FDMA.

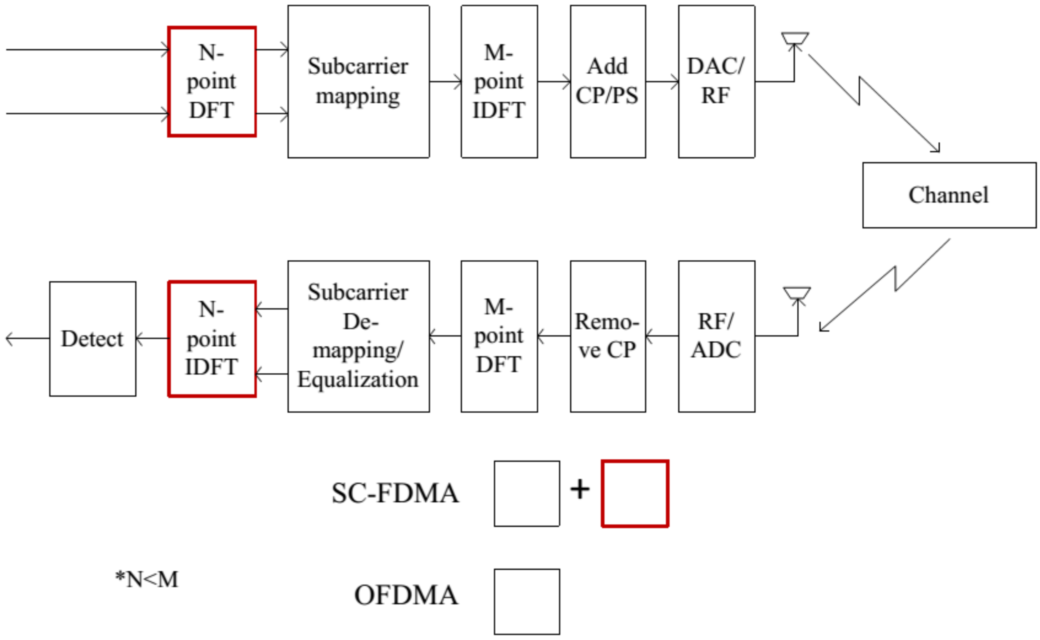

Figure 2.12: Transceiver Comparison between OFDMA and SC-FDMA

Chapter 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.

In LTE, EPS bearer has been defined to enable the EPS to guarantee QoS requirements of different traffic flow between the PDN gateway and the UEs. This EPS bearer serves as the basic element with the LTE QoS support concept. This concept helps in today's mobile communication system when a user performs multiple services with different QoS on a single device. A user can perform Video Streaming and at the same time browse the internet. The QoS policy help to assign all these services based on the QoS requirements in order to guarantee user satisfaction [**17**].

3.1 QoS Classification in LTE Network

In [**6**], [**16**] and [**17**], LTE classify flows into Guarantee Bit Rate (GBR) and Non Guarantee Bit Rate (NGBR). These flows are mapped into radio bearers which are the Over-the –Air connections. There are two types of bearer in LTE network: these are default and dedicated bearers.

3.1.1 Guarantee Bit Rate (GBR)

As the name implies, the bearer guarantee a minimum bit rate for their services. This is accomplished because of special attributes of the bearer that enforce other units to reserve resources (bandwidth) for them. They are used for special applications such as VoIP, Video Conferencing and Online Gaming [**16**]. A GBR has what we called Associated Guarantee Bit Rate (AGBR) which reserves some certain amount of bandwidth for GBR bearers whether being used or not. We also have Maximum Bit Rate (MBR), on the other hand is the maximum bit rate that can be expected to be provided by a GBR [**12**] [**6**].

3.1.2 Non Guarantee Bit Rate (NGBR)

These bearers do not guarantee bit rate for the users. They are kept for application like best effort service. In this classification, resources are not kept for the applications along the flow. They are only served with the left over resources after the GBR applications have been allocated resources [**17**]. Because of these therefore, a Non-GBR bearer may experience packet loss in case of congestion [**12**].

There are also two types of bearer in LTE Network that are associated with the LTE network flows as illustrated in the Figure 3.1 below. These are the Dedicated and Default bearer.

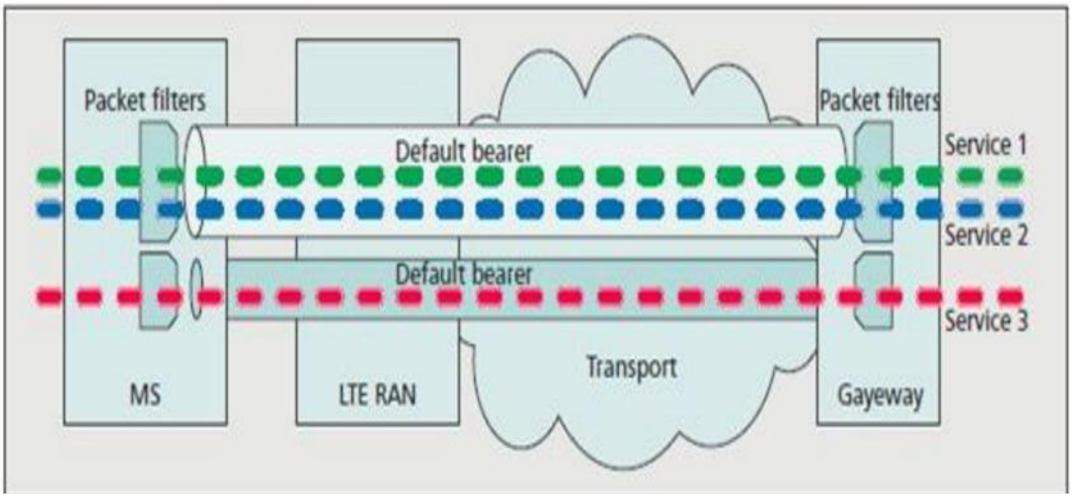

Figure3.1: LTE QoS Framework showing Default and Dedicated Bearers

3.1.3 Default Bearer

This is a Non-GBR bearer, which does not provide bit rate guarantee. This type of bearer is established at the start-up for all traffic along the network [**12**] [**2**] before the GBR applications are re-allocated to the Dedicated channel. Traffic Flow Template (TFT) is used to associate dedicated bearers with their corresponding QoS parameters. Nevertheless, a default bearer may or may not be associated with TFT based on HSS according to [**19**].

3.1.4 Dedicated Bearer

This can operate as either GBR or Non-GBR bearer. Dedicated bearer as GBR allows UE through the help of TFT to specify the required guaranteed bit rate, packet delay and packet loss error rate. This allows the direct mapping of service data flow in both downlink and uplink. The mapping at the downlink is carried out at the eNodeB and UEs while that of the downlink is performed at the S-GW or the P-GW respectively.

3.2 QoS Parameters of an EPS Bearer

3.2.1 Allocation and Retention Priority ARP

ARP is the feature in LTE network that determines whether a bearer request for establishment can be accepted or rejected because of resources limitation [**12**]. It represents the symbolic allocation and retention of radio bearer for call admission control during congestion [**2**]. It is also important to state here that, once a bearer is established, the function of ARP end there. The scheduling and rate control are solely done by the QCI, GBR and NGBR respectively [**16**] [**17**].

3.2.2 QoS Class Identifier QCI

This scalar value specifies the class that the bearer belongs. It helps to determine packets forwarding characteristics. QCI depends on Scheduling weights, Admission Thresholds, Queue Management Thresholds and Link layer Protocol Configuration [**12**] [**14**] [**16**].

3.3 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.

Table 3.1: Standardized QCI values and their characteristics

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

Chapter 4

MODEL METHODOLOGY AND RESULTS

4.1 Introduction

Many researches have been conducted related to modeling MAC layers' scheduling algorithm and admission control of LTE network in order to improve the latency and QoS of applications over the network. This thesis is performing an investigation of the effects of mobility on multimedia applications and the study of various techniques of QoS incorporated in LTE as an all IP network most interestingly, when two or more applications are contending the network medium.

OPNET simulator is the primary simulation tool on which this research is based. The Admission Control of LTE is coordinated by the function known as ***lte-as*** found in the eNodeB of the LTE network. The ***lte-as*** contains a feature called ***lte_admit_control_support_radio_bearer_admit()*** **[**5**]**.

Admission control in LTE network starts from the NAS layer of the core network or the UE and this is only applicable to GBR bearers as Non-GBR bearers are only admitted by default as explained in chapter three.

During EPS bearer admission, EPC communicates EPS ID and QoS parameters to the eNodeB. Then, eNodeB S1 translates the EPS record to the radio bearer (EPS_ID-RB_ID) for the eNodeB AS. eNodeB will therefore, estimates based on the available radio resources if the GBR bearer can be admitted or not. If the EPS bearer is admitted, the information about the admission is exchanged with the UE-AS and if otherwise, the NAS at the core network is informed about the rejection of the EPS bearer. Hence, ESM messages will be sent to indicate the EPC that the radio part of the bearer is active and the core network starts sending the traffic mapped to the admitted bearer. The ***lte_admit_control_support_radio_bearer_admit()*** earlier mentioned in this chapter is responsible for the function stated above. The figure 4.1 below illustrate the lte-as found in eNodeB that contains the function ***lte_admit_control_support_radio_bearer_admit().***

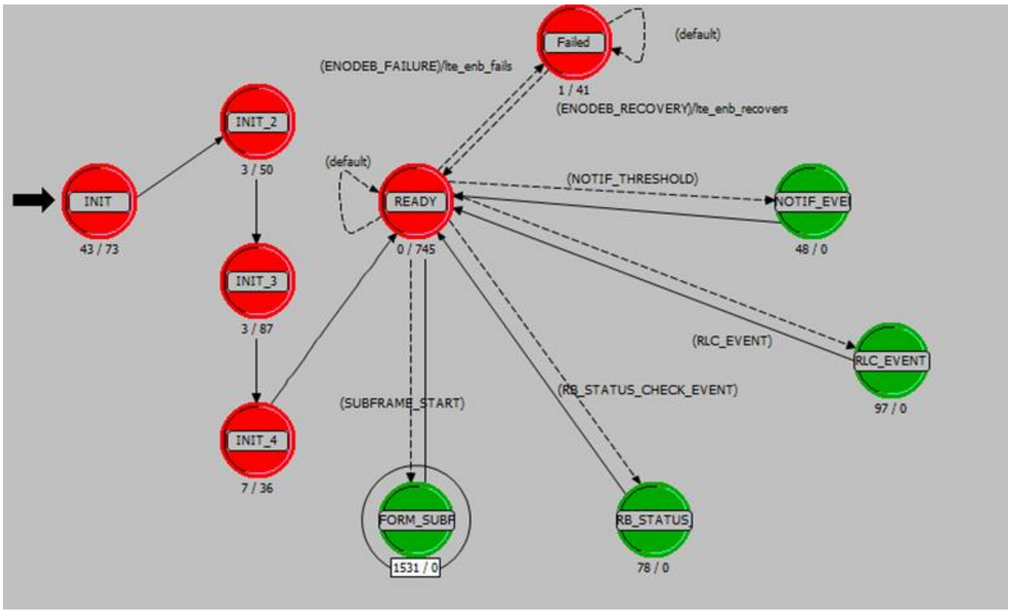

Figure 4.1: LTE-AS. Admission Control

It is also important to note that during EPS bearer establishment, the following procedures are carried out **[**5**]**.

i. Higher layer data packets are queued while the bearers are being created.

ii. Non-GBR EPS are not destroyed or deactivated once they are created

iii. Since certain amount of radio resources are reserved for GBR EPS bearer, they are allowed to go through an admission control process.

iv. Data flow activity through GBR EPS is monitored. In this case, if an EPS Radio bearer becomes inactive for a prolong period of time, its radio resource assigned to it is turned down and released for another bearer and it is reactivated again once it gained another SDF.

v. If GBR EPS bearer request to create radio resource for activation is turned down because of limited resources, queue packets for that bearer are flushed and if any of its SDF becomes active again, the radio bearer creation is reinstated.

vi. During congestion, Admission control preempt GBR radio bearers with higher ARP (Low-Priority) for GBR radio bearer with lower ARP (High-Priority) if the created radio resources are not sufficient to admit the High-Priority GBR bearer.

vii. Sometimes, link adaptation procedure changes the MCS index of the UE due to channel condition. In this case, if the magnitude of the change exceed a set threshold, the required resources by the active GBR radio bearer are reevaluated by the admission control feature of the eNodeB. If the MCS index is lower, the resources required to guarantee QoS for the active GBR bearer may not be available at the eNodeB during this present condition and hence, the eNodeB may release the bearer using the procedure similar to releasing bearer during inactivity or preemption **[**5**]** **[**20**]**.

4.2 Related Work

As earlier mentioned, LTE is designed as an All-IP-Network. This implies that all applications on the network medium share the same channel. Guaranteeing QoS of delay intolerable multimedia applications now become very crucial to fulfill the benefits attached to 4G network in the 3GPP [**4**].

In [**1**], the fairness-based preemption algorithm for LTE-Advanced has been proposed. This algorithm was designed to consider bearer's QoS over-provisioning with respect to their minimum QoS needs into the partial preemption decision. The paper discoursed on the usual preemption of Non Guarantee Bit Rate Bearers by the Guarantee Bit Rate bearers when the available resources to admit incoming GBR bearers is not sufficient. In this case, the paper proposed a fairness based algorithm that will be fair enough to admit the GBR bearers and still keep little resources to maintain the Non GBR bearers already admitted.

The algorithm is an improvement on the traditional preemption algorithm that allows the total preemption of low priority bearers. The proposed algorithm however tries to partially preempt the resources of the active bearers based on their priorities as well as their over-provisioned resources. The result was an improvement on the performance of low priority bearers along LTE network but it never improve on the performance or investigate the QoS of real time applications over LTE channel, which is one of the researches, conducted in this thesis.

In [**21**], comparative performance of different schedulers in LTE Downlink channels were investigated. This is to improve on the QoS of applications on the LTE channels by using the right scheduler that guarantee QoS. Three scheduling algorithms were investigated in order to explore the strengths and weakness of these algorithms. The investigation was conducted using mixed traffics of different multimedia applications.

The result obtained helps to ascertain the importance of prioritization of multimedia traffic in order to achieve better QoS performance in both low and high network load. Although, this is somehow related to this thesis work but it does not mention anything on the effect of mobility on the performance of multimedia application and the amount of GBR bearers that are preempted during mixed traffics of multimedia application as discussed in this thesis.

In [**22**], simulation and modeling was conducted on the downlink subcarrier allocation scheme of LTE network. This was done to investigate the efficient utilization of the available resources in LTE network at the downlink layer. The downlink subcarrier was modeled using opnet modeler to investigate the performance analysis of real-time and non-real time applications when they are allowed to share the same network medium. The research work in [**22**] tried to show that the MAC scheduler in LTE needs to be aware of not only of the condition of the channels but in addition, the different QoS requirements of various applications along the network. The results obtained show an improvement in the performance of VoIP application when each traffic are grouped to different classes over the result obtained when they are allowed to share the same class. The thesis work somewhat tries to investigate these performances in a more elaborate way.

4.3 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.

4.3.1 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-to-use 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).

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 are the reasons for OPNET modeler adoption.

- OPNET provides more simulating features than any other R&D simulating tools presently in practice.

- OPNET provides easy-to-use and friendly graphical interface for simulating events and viewing results
- OPNET contains a dynamic development environment with features that support both distributed systems and modeling of communication networks
- It has user friendly and wide user guide documentation to aide users during simulation.
- The results obtained from OPNET Simulation can easily be exported into spreadsheets and it has a wide tools to interprets and plot the results in different graphical modes.

4.3.2 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.

4.3.2.1 Application Config Utility Object

This is the first step in specifying and configuring standard applications behaviors. It contains names and description tables for different application and their network parameters.

4.3.2.2 Profile Config

This is use to specify different user profiles and their individual nodes in the networks. The application layer traffics produced by the nodes depend on the user application configured in the application config utility object.

4.3.2.3 LTE_Att_Definer_Adv

This nodes is use to configure and keep the EPS bearer explanation and PHY pattern in the entire network which can be modeled in all the nodes in the network.

4.3.2.4 Mobility Config

This node is use to configure and model the position, speed and movement of nodes in the network based on the configuration and the predefined parameters.

4.3.2.5 Lte_access_gw_atm8_ethernet8_slip8_adv node

This node is used for IP-based gateway in LTE. It can support more than 8 Ethernet and 8 serial lines at selectable data.

4.3.2.6 Lte_enodeb_4ethernet_4atm_4slip_adv node

This is the base station of the LTE. It can work conveniently with four Ethernet interfaces and four serial interfaces respectively at selectable data.

4.3.2.7 Lte_wkstn_adv node

This is used for workstation in LTE as sender and receiver over the TCP/IP and UDP/IP. The represents UEs in LTE network.

4.3.2.8 PPP_DS3 link

The link is used for Ethernet connection operation. It has six nodes in running IP with speed 148.61Mbps. This type of link operates in duplex mode.

4.3.3 Problem Formulation

In this thesis, we are interested in the performance analysis of multimedia applications over LTE, the new 4G network interface. QoS is one of the most important areas of research in the study of multimedia system. User satisfaction comes with better network performance and much research need to be carried on in this area. Many researchers have developed some ideas in form of algorithms to enhance better performance of users applications to guarantee their QoS. To guarantee QoS for multimedia application, [**12**] and [**2**] give clear view on the packet delay budget that individual multimedia applications can take. Take for example, Conversation Voice (VoIP) and conversational video will perform better with an end-to-end delay budget of 150ms and less. If the end-to-end delays of these applications rise beyond the stated value, the acceptable performances and the QoS are not guaranteed.

In this case, we try to investigate how VoIP and Video Conferencing react during normal and congested scenarios at varying speed by studying the following QoS parameters.

- End-to-End delay performance
- Packet loss performance
- Packet delay variation
- Total Numbers of Admitted GBR bearers
- Total Numbers of Rejected GBR bearers
- Voice Mean Opinion Score (MOS)

Since both VoIP and Video Conferencing belong to the class of GBR as earlier stated in our study, we further investigate the reactions of these applications during heavy traffic to study the number of GBR bearers that are preempted at congested situation.

To understand how the Admission Control algorithm monitors radio resources and admits/preempts/rejects radio bearers when multiple multimedia applications are contending the network resources with background traffics, we went further to study the effects of HTTP and FTP which belong to NGBR on the VoIP and Video Conferencing applications which are GBR radio bearers. This help us to study the effects of load on the QoS performance of multimedia application and the fairness of the Admission Control algorithms on the NGBR radio bearers in the network medium.

Lastly, according to the admission control policy of the 3GPP, different applications are grouped to different classes based on the type of radio bearers. Radio bearers on LTE network are grouped with QCI and ARP to different classes based on priority. In other word, VoIP radio bearer which has zero tolerance for delay is grouped under a class with QCI of one and ARP of one respectively, while FTP radio bearer that does not really care about latency and as a NGBR bearer can be grouped with QCI seven and ARP seven respectively. We conduct a study to investigate the performance of VoIP and Video Conferencing when they are grouped with FTP and HTTP under the same QCI class and what happened when they are grouped under different class as illustrated in [**22**].

To conclude this section, I will briefly explain some of the aforementioned QoS parameters.

4.3.3.1 End-to-End delay performance

This is sometime refers to as Analog-to-analog or Mouth-to-air delay. It is the time required for a packet to transverse from one node (UE) to another (UE). This delay comprises of Network delay, Encoding delay, Decoding delay and Compression and decompression delay. VoIP application can only experience sender delay, network delay and receiver delay while Video Conferencing may experience all the delays listed

- Network Delay: This is the time at which the sender node give the packet to RTP to the time the receiver got if from RTP

- Encoding delay (Sender delay): This delay occurs at the sender node. It is the time taking by the sender node to encode the packet to be sent. It is computed from the encoding scheme.
- Decoding delay (Receiver delay): This delay occurs at the receiver node. It is the time taking by the receiving node to decode the sent packet. It is assumed to be equal to the Encoding delay.
- Compression and Decompression delay: These delays come from the corresponding attributes in the voice application configuration.

4.3.3.2 Packet Loss Performance

When packet travels from sending node to destination node, there might be some chance for packet loss. This loss can be determined using the formula below

$$\textbf{Packet Loss} = \left(\frac{\textit{Number of Sent Packet} - \textit{Number of Received Packet}}{\textit{Number of Sent Packet}}\right) * 100$$

4.3.3.3 Packet Delay Variation

The variance among end-to-end delays for voice packets. Its sometimes refers to as Jilter, which is the variance of signal with respect to some clock signal or variation of a delay with respect to some reference metric like average delay or minimum delay.

4.3.3.4 Total Numbers of Admitted GBR bearers

This is the total number of GBR bearers that are currently admitted at a particular eNodeB. Recall that GBR bearer does not tolerate jilter or delay, hence, the measure of the total number of admitted GBR bearers help us to study how the LTE network guarantee QoS of multimedia applications along the network.

4.3.3.5 Total Number of Rejected GBR bearers

When the available resources in the network has been utilized. Any GBR bearers that is requesting admittance in the time may be rejected. Although, if there are some NGBR bearers in the network, they could be preempted for the incoming GBR bearer. But in a situation, where by the network is congested with GBR bearers with higher QCI than the incoming GBR and there is no available resource to admit the bearer, such GBR bearer could be rejected.

4.3.3.6 Mean Opinion Score MOS

In voice communication most especially internet telephony, MOS is used to provide a numerical measure of the quality of human speech at the destination end of the circuit [**6**] [**16**]. This method of voice quality test has been in used for decades to obtain human user's view of the quality of the network. The values of MOS are rated from 1-5 Meaning, the illustration is stated as

1- Very Bad, Impossible to Communicate
2- Very Annoying, Nearly impossible to Communicate
3- Annoying, Manage to Communicate
4- Fair imperfection- Sound clear, Good communication though
5- Perfect Communication, like face-to-face or radio reception, Excellent communication

There are nowadays numbers of software tools that carried out MOS automated testing in VoIP deployment. This software try to take into account all the network dependency conditions that could influence the voice quality. Some of these software are ApparaNet voice, Brix Voice Measurement Suite, NetAlly, PsyVoIP and VQmon/EP [**2**] [**6**].

4.3.4 General Parameters for Simulating LTE networks

In all our simulations and the scenarios in this study, we set up our LTE configuration with the parameters as illustrated with the Table 4.1 below and this setting is fixed throughout our study.

Table 4.1: General LTE parameters and Configuration used [**14**] [**16**]

LTE Parameters	Value assigned
VoIP QoS Class Identifier	1GBR
Video Conferencing QoS Class Identifier	2GBR
Ftp QoS Class Identifier	6 NGBR
Http QoS Class Identifier	6NGBR
UL/DL Bandwidth	20MHz
Downlink Guaranteed Bit Rate	1.5Mbps
Uplink Guaranteed Bit Rate	1.5Mbps
Downlink Maximum Bit Rate	1.5Mbps
Uplink Maximum Bit Rate	1.5Mbps

4.4 Simulation Scenarios Design and Results

We set up four different scenarios to investigate the performance of VoIP, Video Conferencing at varying speed and under heavy traffic when they are mixed with background traffic created by HTTP and FTP applications.

4.4.1 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.

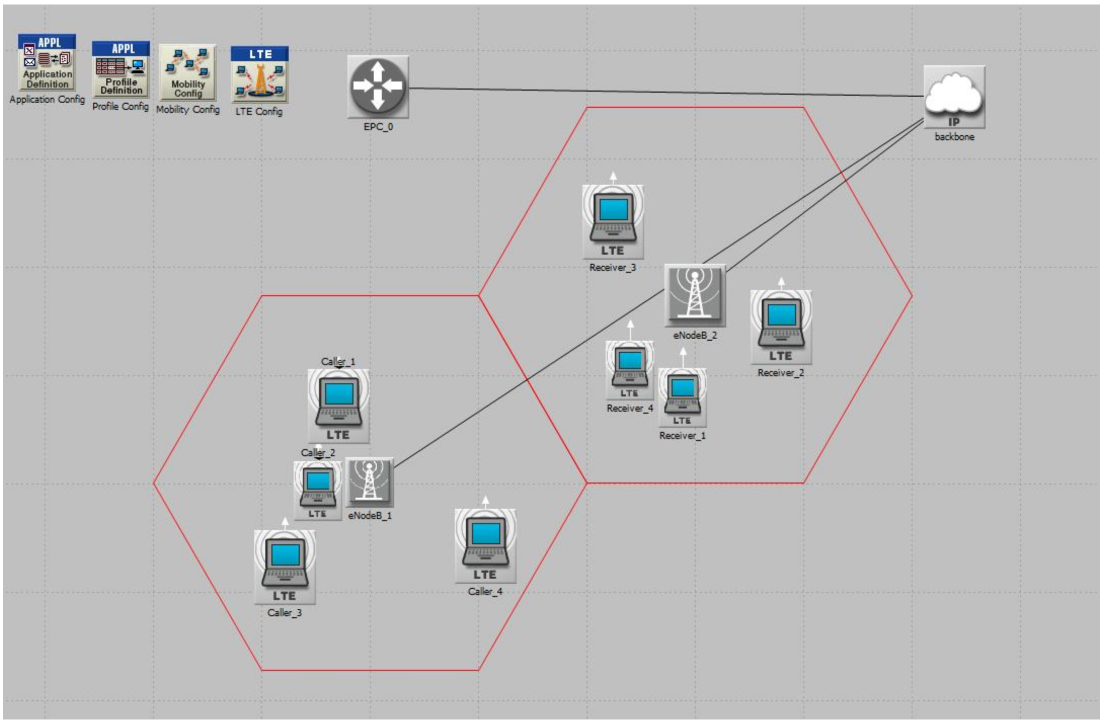

Figure 4.2: VoIP Application at varying speed network scenario setup

Scenario One is a setup of an OPNET Model to simulate the performance of VoIP application at varying speeds. Using all the available models as earlier described in this chapter. Two eNodeB (eNodeB_1 and eNodeB_2) were configured each with four workstations. The four workstations at eNodeB_1 were configured as callers while the four workstations with eNodeB_2 are the receivers respectively as shown in Figure 4.2.

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:

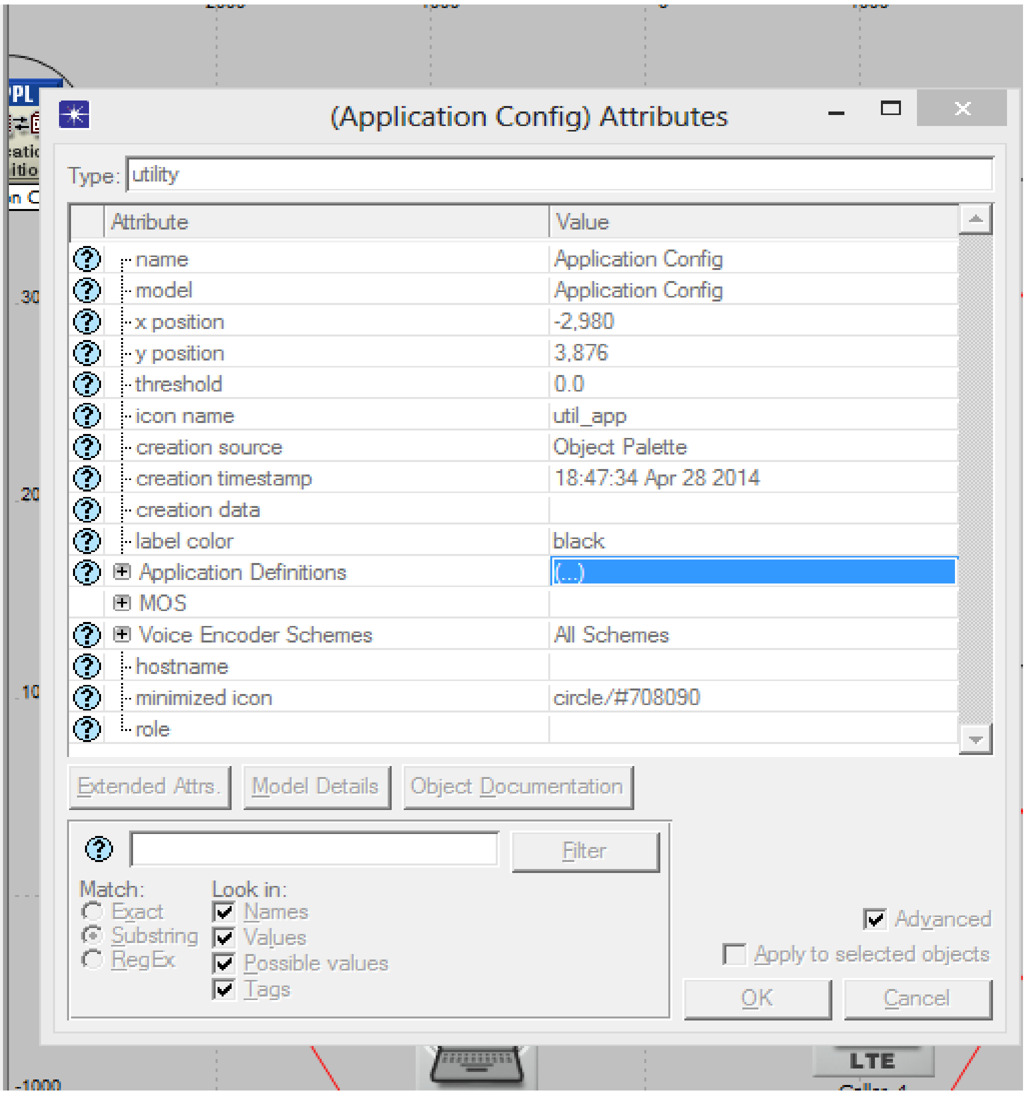

Figure 4.3: Application Configuration Attributes

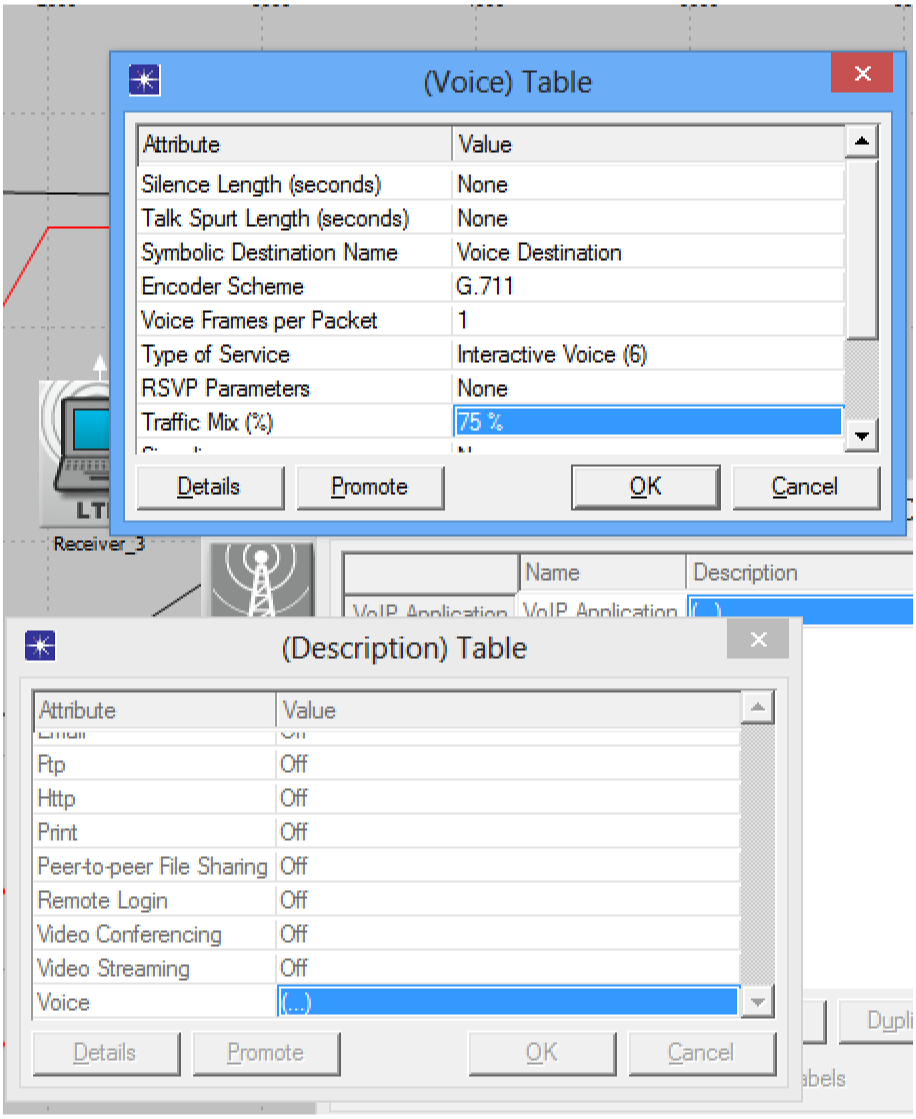

Figure 4.4: Application Configuration VoIP Attributes

Table 4.2: VoIP Configuration Parameters used in Application Configuration

Attributes	**Value**
Encoder Scheme	G.711
Voice Frame per packet	1
Type of Service	Interactive Voice (6)
Traffic Mix	75%
Compression Delay (seconds)	0.02
Decompression Delay (seconds)	0.02

For the Profile Configuration, we configured the application such that VoIP call will be added every second till the end of the simulation. The first VoIP call was established at 105Sec after the simulation started and this continues with addition of VoIP calls every second till the end of the simulation. The Profile Configuration attribute is shown on the structures in Figure 4.5 and 4.6 below.

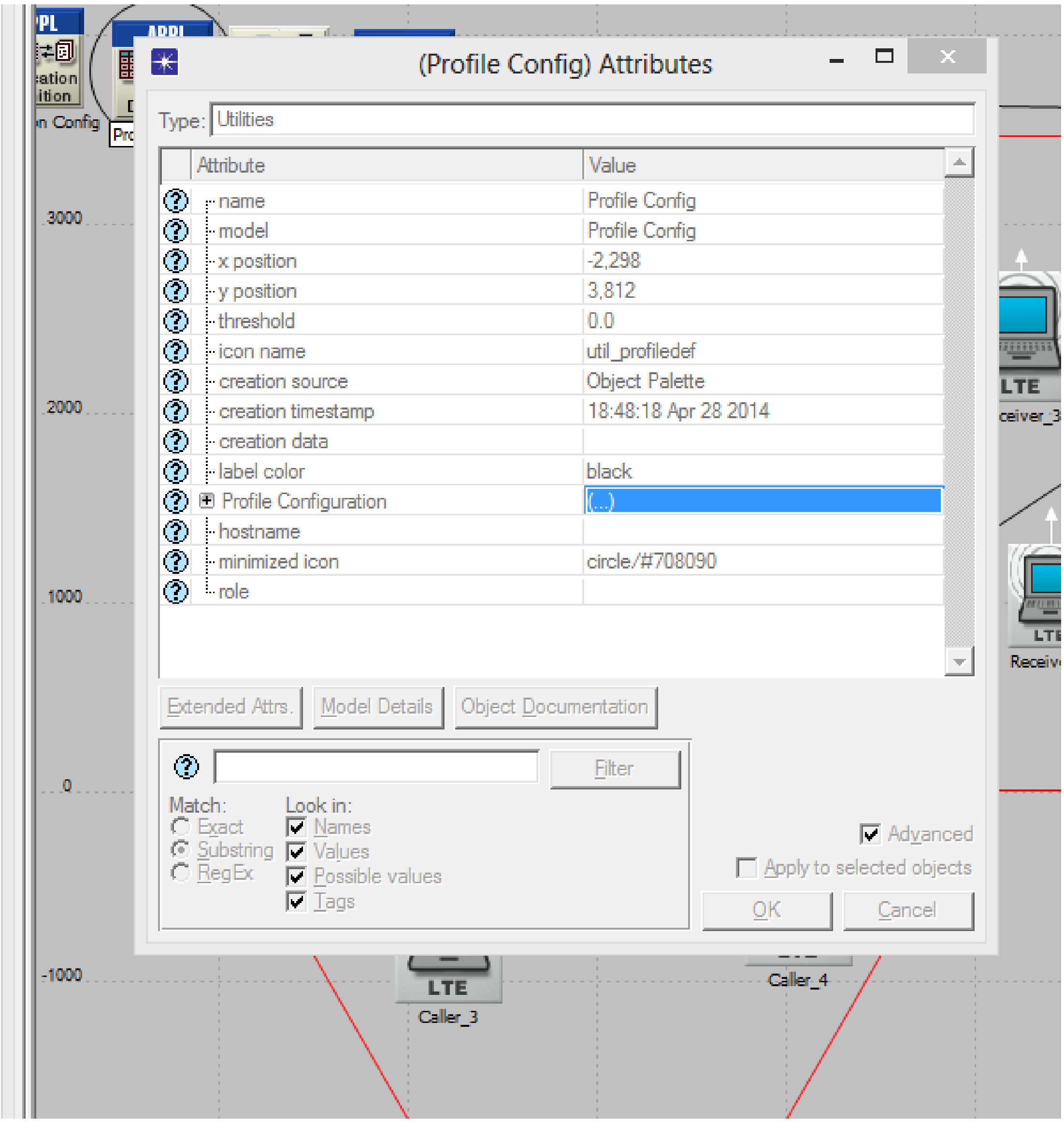

Figure 4.5: Profile Configuration for VoIP Application

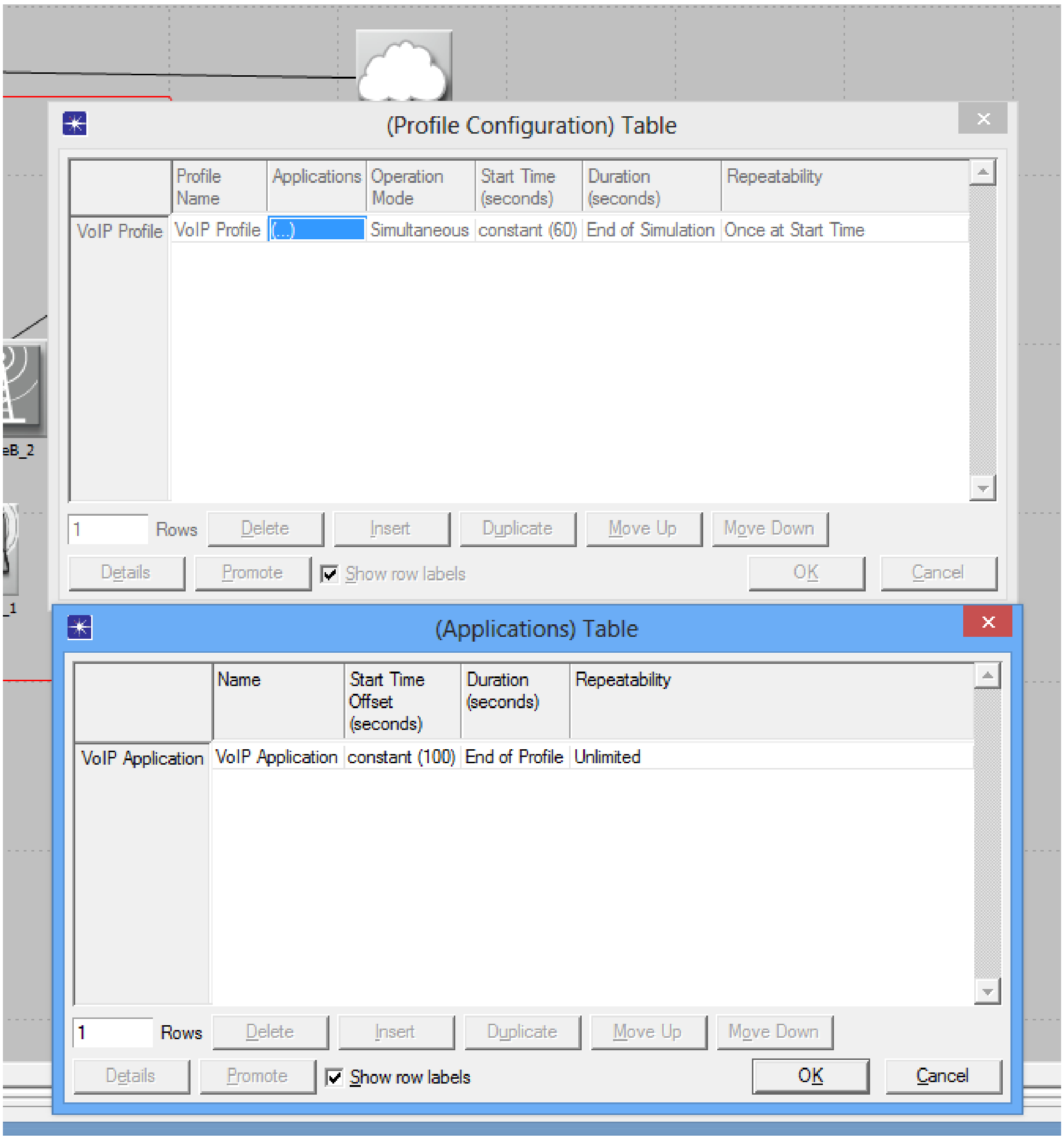

Figure 4.6: Profile Configuration Parameters for VoIP Application

In this scenario, two simulations cases were considered. In Case 1, the speeds of the mobile nodes were set to zero while in Case 2 the speeds of the nodes were set 30 m/s respectively. The simulation was allowed to run for 400seconds. The results of Packet End-to-End delay performance, and Packet Loss Performance and MOS value were collected to investigate the effects of varying speeds on these QoS parameters.

4.4.2 Results of VoIP Application at Varying Speeds in Scenario One

Our interest here is to study the performance of VoIP application at static and varying speeds. Just as illustrated in our Scenario One. We will look at the results generated from the simulated Packet E2E delay performance, VoIP MOS and Packet Loss Performance in this section.

4.4.2.1 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 End to End Delay variation (Second)	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.

Table4.4: Packet E2E Delay of some selected nodes of VoIP Users at Varying Speeds

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

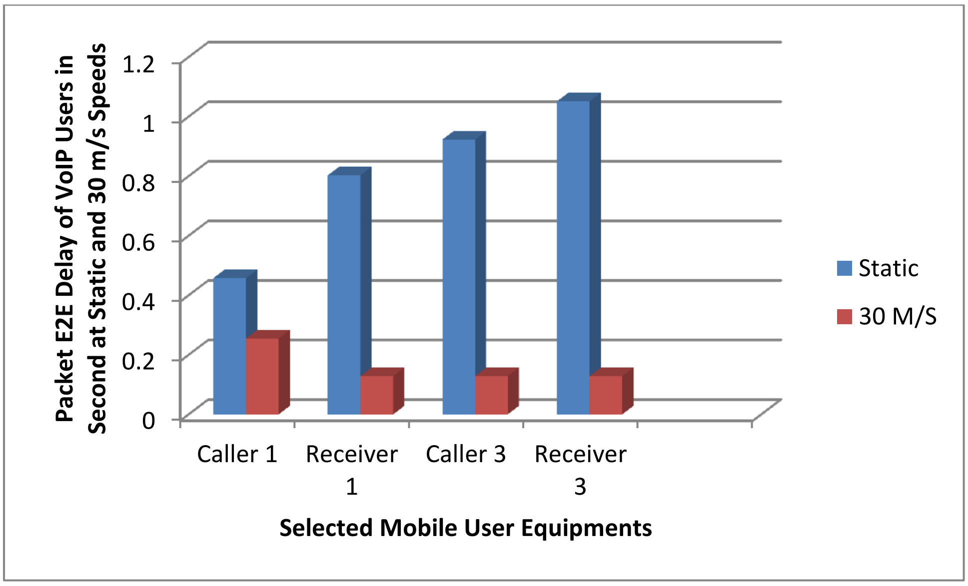

Figure 4.7: Packet E2E Delay of some selected nodes of VoIP Users at 0 m/s and 30 m/s speeds.

From the results obtained in Figure 4.7, its evident that the VoIP application users in the simulation perform better at the mobile nodes than the static nodes and the mobile nodes at different speeds have very little different in their E2E delay performances. This is because, the nodes in motion are experiencing high amount of losses and HARQ retransmission are giving up compared to the static nodes. As a result of this, a less traffic successfully transmits between sender and receiving nodes in the mobile nodes case and they experience less delay.

In the static nodes case, more traffics are making it to the destination nodes but with more delay and due to the retransmission in the wireless MAC, some retransmission that has not reached the maximum allowed are making it to the destination nodes there by causing more delay.

4.4.2.2 VoIP Mean Opinion Score (MOS).

MOS measures subjective call quality for all VoIP calls. It scores VoIP calls range from value 1 for unacceptable (Very Poor) performance to value 5, an acceptable (Excellent) performance. If a VoIP call is scored with a value greater or equal to 4.02, such VoIP call is acceptable and the user is being satisfied with the VoIP call.

Table4.5: Mean Opinion Scores Values of Selected VoIP Nodes at Static and Varying Speeds.

Selected Nodes	**Static**	**30 m/s**
Caller 1	1.4138	4.3323
Receiver 1	1.5451	4.3476
Caller 3	1.1462	4.3340
Receiver 3	1.2341	4.3349

Table 4.5 gives the MOS values of some selected VoIP users from our simulation in Scenario 1 while the range of performance of VoIP users based on the MOS is plotted on a histogram as shown in Figure 4.8 below.

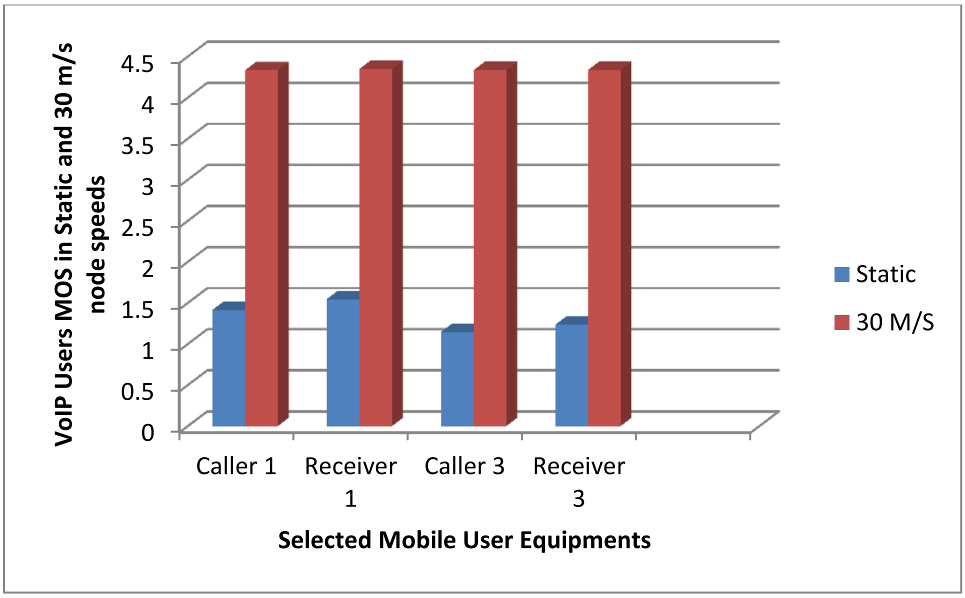

Figure 4.8: Mean Opinion Scores Chat of Selected VoIP Nodes at Static and 30 m/s Speeds.

From the MOS results obtained as illustrated in our chat in Figure 4.8, the Static VoIP nodes selected have poor MOS values owing to the high E2E delay experienced by the users as a result of accumulation of traffics. The VoIP users in the case of 30 m/s scored an acceptable MOS values and these values have very little different as shown in all the mobile nodes selected. This is also as a result of the less traffic these mobile nodes experienced due to high packet dropped in the nodes which reduce the E2E delays in these cases. The MOS of the Mobile Nodes in 30 m/s score more than 4.02 and thereby give satisfaction to the VoIP users.

4.4.2.3 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

$$\textbf{Packet Loss} = \left(\frac{\textit{Number of Sent Packet} - \textit{Number of Received Packet}}{\textit{Number of Sent Packet}}\right)\textbf{*100}$$

We took the results of the VoIP traffic sent and received from our selected nodes in all the four mobility cases simulated and compute the PLP using the formula stated above. The result of the PLP obtained is tabulated in Table 4.6 below. We went further to plot the PLP of these selected nodes from each mobility cases simulated as explained to see the variation in percentage of losses experienced by the VoIP users in the network. The graph in Figure 4.9 which represents the PLP of VoIP users indicated that VoIP users at static nodes experienced high PLP due to the large number of traffics in the network, the mobile nodes in cases with VoIP users speed of 30 m/s experienced very little or no PLP owing to the little traffics that successfully transverse from sender nodes to destination nodes as indicated in the graph in Figure 4.9.

Table 4.6: Packet Loss Performance (%) of some selected Nodes from Scenario One

	Static	30 m/s
Caller 1	35.30	0.12
Receiver 1	32.10	0.30
Caller 3	59.00	0.08
Receiver 3	54.10	0.11

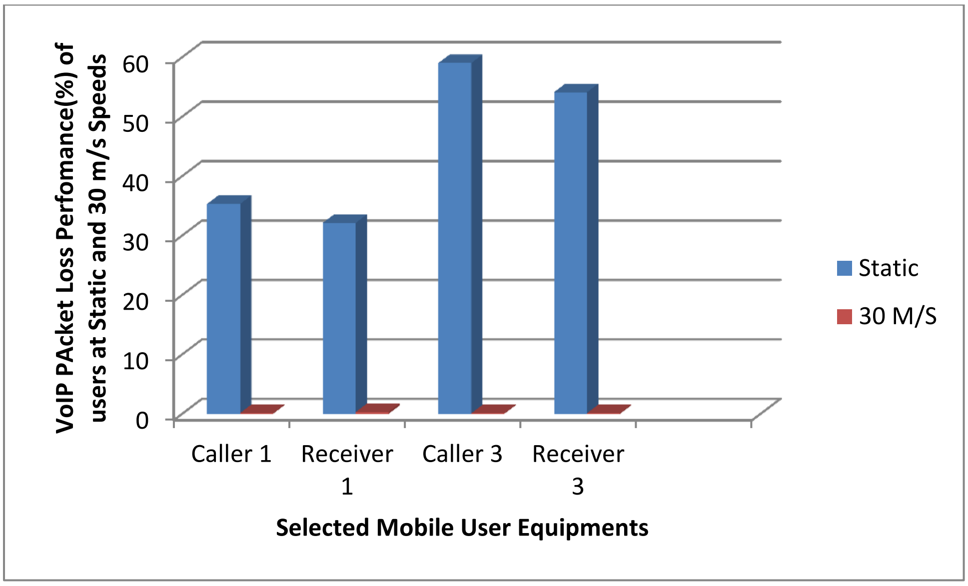

Figure 4.9: Packet Loss Performance (%) graph of some selected Nodes from Scenario One.

From our finding in the simulation, we observed high number of GBR bearer's rejection in case 30 m/s mobility speed. While in the static case, the number of GBR rejection drastically decrease which give rise to admittance of more users and hence, an increase in the overall PDV and E2E delay of users in the network.

The Table 4.7 below gives a summary of the total number of rejection and total number of admission of GBR in the four mobility cases observed in our scenario one. Since there are only two eNodeB in our simulation, the statistics observed showed that Static mobility case has the highest number of admitted GBR bearers with 38.328 and 27.505 bearers in both eNodeB1 and eNodeB2 respectively. The total number of Rejected GBR bearers also in Static Nodes are far less than that of nodes moving at 30 m/s as shown in the Table 4.7 below.

Table4.7: Admission Control Table for Admitted and Rejected GBR bearers of Static and Mobile VoIP users

	Static		**30 m/s**	
	eNodeB1	eNodeB2	eNodeB1	eNodeB2
Total Number of Admitted GBR bearers	38.328	27.505	3.9933	0.0
Total Number of Rejected GBR bearers	1,775.2	1,570.93	22,920.94	0.00

4.4.3 Scenario Two: Video Conferencing Application at varying speeds

The OPNET simulation environment in the Figure 4.10 below represent some mobile and static user's nodes running Video Conferencing application over an LTE-4G network.

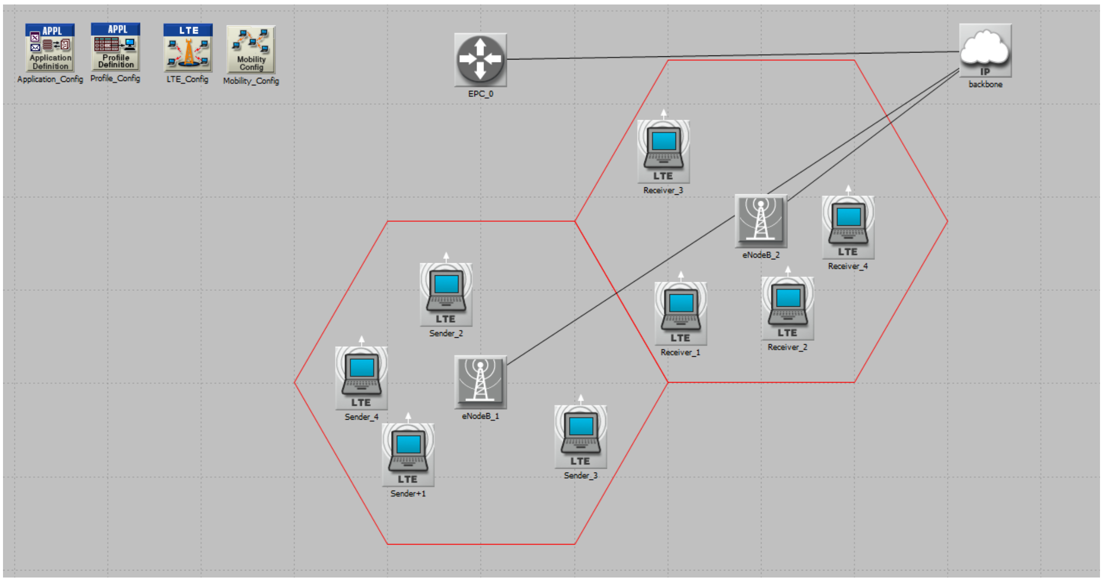

Figure 4.10: Video Conferencing Application under varying speed

In this scenario, the aim is to investigate the QoS performance of Video Conferencing at varying speed just as we observed in Scenario One. The Application Definition Attribute is set as Video Conferencing with 30Frame/sec and traffic mix as 50%. The Type of Service is set to Interactive Multimedia. The speed of the UEs were set to zero in case one while the in case two, the speeds were set to 30 m/s respectively. The simulation was allowed to run for 400sec and the performance analysis based on the packet end-to-end performance and packet delay variation, were collected.

4.4.4 Results of Video Conferencing at Varying Speed Scenario Two

4.4.4.1 Packet Delay Variation Results under Varying Speed of Video Conferencing Users

PDV or jitter is the difference in the E2E one-way delay of some selected packets with any lost packets ignored in the network. As earlier stated in our previous section, we simulated our Video Conferencing application in order to measure the PDV experienced by users as a results of mobility of the users.

From our simulation in Scenario two, we obtained the results of PDV tabulated in Table 4.8 below and plotted the variation in the performance of some selected video conferencing users as simulated in our different mobility cases simulated in Figure 4.11 below.

Table4.8: Packet Delay Variation (millisecond) of Video Conferencing Users at Varying Nodes speed

Selected Nodes	**Static**	**30 m/s**
Sender 1	1.90	0.012
Receiver 1	0.69	0.020
Sender 3	2.3	0.030
Receiver 3	1.0	0.015

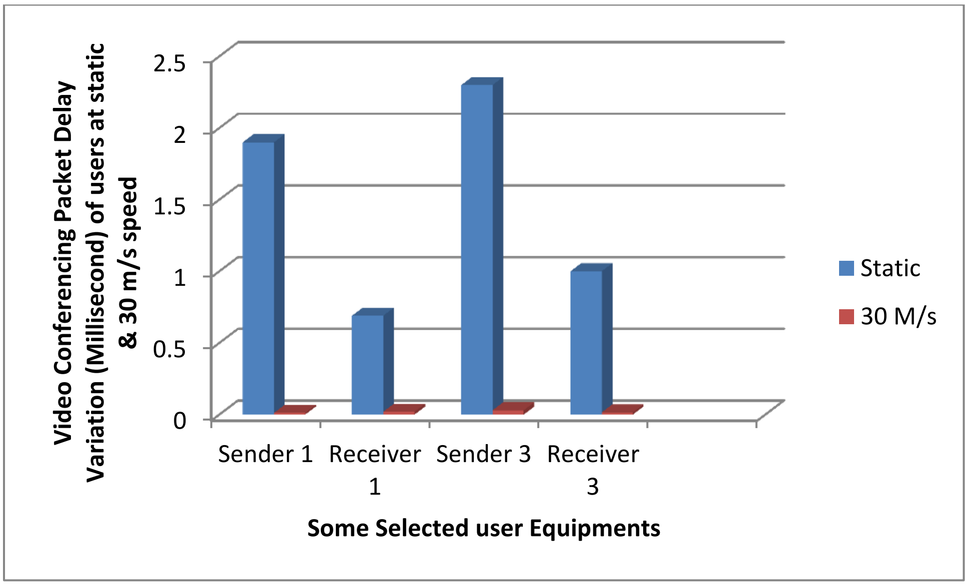

Figure 4.11: Graph of PDV of Video Conferencing Users at Varying Nodes speeds

From the graph in Figure 4.11, we observed that the Video Conferencing Users at static mobility case experienced more PDV than those in motion. The reason for these results could be attributed to high traffic accumulation in the case of static nodes while little or less traffic are allowed to transverse from sender nodes to destination nodes in the cases of mobile nodes in motion.

4.4.4.2 Packet E2E Delay Variation Results under Varying Speed of Video Conferencing Users

End-to-end packet delay which is the time needed for a UE (Sender) to send packet to another UE (Destination). As earlier explained, it comprises of the Sender delay, Receiver delay and the Network delay. From our simulation in Scenario two, the Packet E2E delay of Static Nodes and that of nodes moving at 30 m/s for some selected nodes are tabulated in the Table 4.9.

Table4.9: E2E delay performance of Video Conferencing at Varying speed (Millisecond).

Selected Nodes	**Static**	**30 m/s**
Sender One	40	20.1
Receiver One	42	22.3
Sender Three	56	21.9
Receiver Three	38	21.2

The result of the E2E delay performance of Video Conferencing at static and 30 m/s speeds are plotted in the graph in Figure 4.12 below. From the result, we observed the E2E performance of nodes running at 30 m/s to be better than those nodes at static mode owing to the reason earlier stated that static nodes try to accept as much as possible GBR bearers which result to much delay whereas mobile nodes rather drop the packets and admit less GBR bearers and hence less E2E delays.

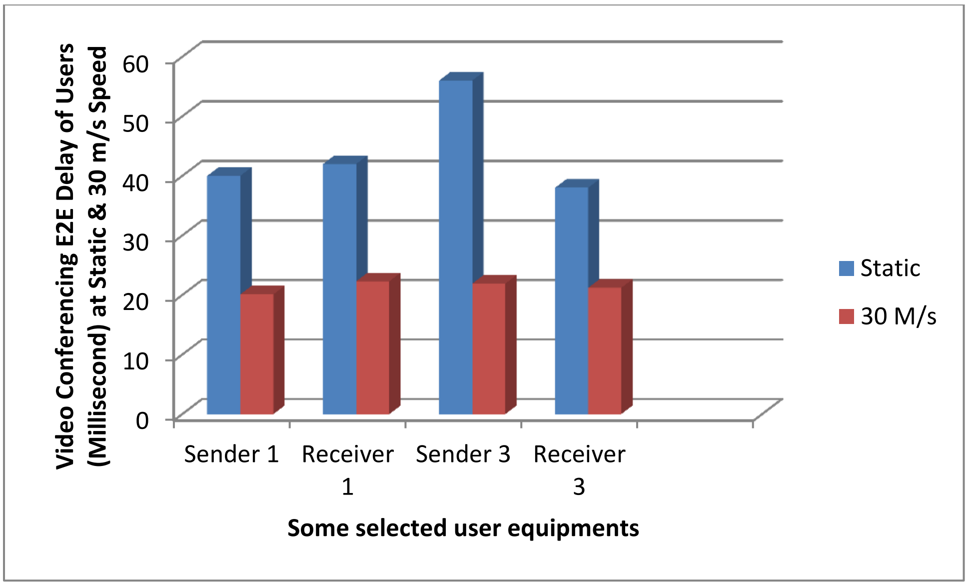

Figure 4.12: Packet E2E performances of Video Conferencing application

4.4.5 Scenario Three: Congested Multimedia Application

We set up the OPNET environment as shown in Figure 4.13 below to study the effect of best efforts traffic over the QoS performances of multimedia applications and users satisfaction.

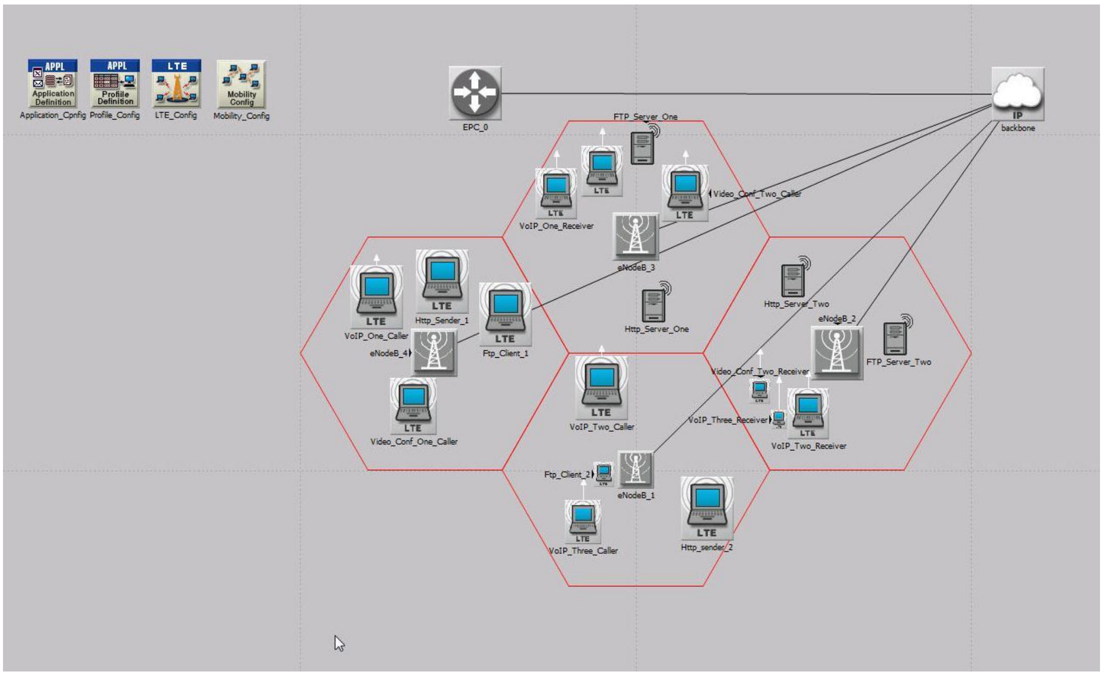

Figure 4.13: Congested Multimedia Application

LTE 4G networks is an IP based system. This implies that all applications on the network share the same narrow band network channel. Although, admittance of applications are done by priority based on the ToS. In this scenario, our interest is to investigate the performance of VoIP and Video Conferencing applications when they are contending the network medium with background traffics created by Ftp and Http applications. In this scenario, we will be able to see how multimedia applications (VoIP and Video Conferencing) as GBR react to network load under congestion. We will also investigate the fairness of GBR application (VoIP and Video Conferencing) with NGBR (Http and Ftp) applications to see the fairness of individual applications either GBR or NGBR in the network in terms of bandwidth greediness of individual applications. The simulation contains four eNodeBs each with four UEs. VoIP calls were generated at 100ms after the simulation stated. We also established Video Conferencing among different UEs. Two cases were investigated in this scenario. In Case 1, we simulated the performance of the VoIP and Video Conferencing to study the traffic received and sent as well as the end-to-end delay performance of the applications when they are allowed to run on the LTE network alone. In Case 2, Ftp and Http Applications were introduced as background traffics and the results of the traffic sent/received and end-to-end performance of VoIP and Video Conferencing applications as well as the average download time of Ftp and page response time of Http applications on the network were collected. The Application and Profile configurations of VoIP and Video Conferencing in this scenario are the same as the configurations in scenarios one and two above. The Ftp and Http are configured as NGBR with QCI 6 while VoIP and Video Conferencing configured as GBR with QCI of 1 and 2 respectively on the LTE attributes model.

4.4.6 Results of Congested Multimedia Applications Scenario Three

Our simulation in scenario three was configured to investigate the performance of VoIP and Video Applications running in a giving network when they are not contending with any traffics and when they are allowed to share the narrow band channel with best efforts traffic created by the introduction of Http and Ftp application. In this section, we will study the performances of VoIP and Video Conferencing application under two cases.

Light Load: this is when VoIP and Video Conferencing applications are running in the network and use the network resources alone.

Heavy Load: this is when VoIP and Video Conferencing applications share the network resources with background traffics created by the introduction of Http and Ftp applications as best effort traffic.

We will now go further to see the performance of VoIP and Video Conferencing applications by studying the results of some QoS parameters obtained from our simulation.

4.4.6.1 VoIP Application performance under Light and Heavy Load

We will compute and analyze the MOS and Packet Loss Performance results of some VoIP users in this section.

4.4.6.1.1 MOS of VoIP at Light and Heavy loads condition

Just as earlier explained in our previous section, MOS value is used to predict how satisfactory a VoIP call is to end users. In this section, our interest is to measure the subjective call quality during the two conditions stated above (Light and Heavy loads). The Table 4.10 below gives the MOS values of some selected VoIP users under heavy and light loads. From the table, the MOS values of selected VoIP users under Light load ranges from 2.0945 to 2.5235. Although, these MOS values are not totally satisfied for quality VoIP call, the reason could be attributed to the present of Video Conferencing traffics in the network. In addition, the results of the MOS values under heavy Load range from 1.1635 to 1.7597. This is a total unacceptable MOS values and this can be attributed to the heavy traffic created by the present of best effort users. The variation in the MOS values of VoIP is plotted in the graph in Figure 4.14 below.

Table4.10: MOS of VoIP Application users under Light and Heavy Loads

Selected VoIP Users	**MOS of VoIP under Light Load**	**MOS of VoIP under Heavy Load**
VoIP One Caller	2.0945	1.3856
VoIP One Receiver	2.1520	1.2098
VoIP Two Caller	2.3506	1.3099
VoIP Two Receiver	2.3021	1.3354
VoIP Caller Three	2.0945	1.7597
VoIP Receiver Three	2.5235	1.1635

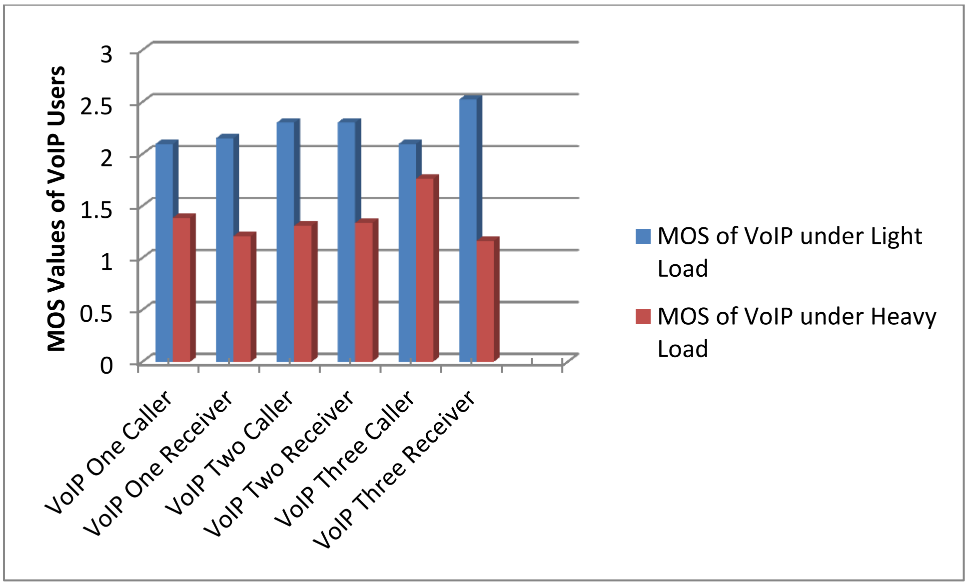

Figure 4.14: MOS of VoIP Users under Light and Heavy Loads

4.4.6.1.2 Packet Loss Performance of VoIP Users at Light and Heavy Load conditions

After studying the MOS values of VoIP users in our Scenario three simulated as explained above, we further want to ascertain our findings by analyzing the packet loss performance of some selected users in the network. The results obtained after calculating the PLP of our VoIP application users are tabulated below in Table 4.11. We went further to plot the graph of PLP against the selected VoIP users to see their PLS variation under the heavy and light load simulated as further explained. The graph is shown in Figure 4.15

Table4.11: Packet Loss Performances value (%) of VoIP Application under Light and Heavy Loads

Selected VoIP Users	**PLP of VoIP under Light Load**	**PLP of VoIP under Heavy Load**
VoIP One Caller	22.98	52.03
VoIP One Receiver	23.02	66.30
VoIP Two Caller	14.13	45.50
VoIP Two Receiver	13.15	49.51
VoIP Caller Three	16.35	29.86
VoIP Receiver Three	10.25	66.17

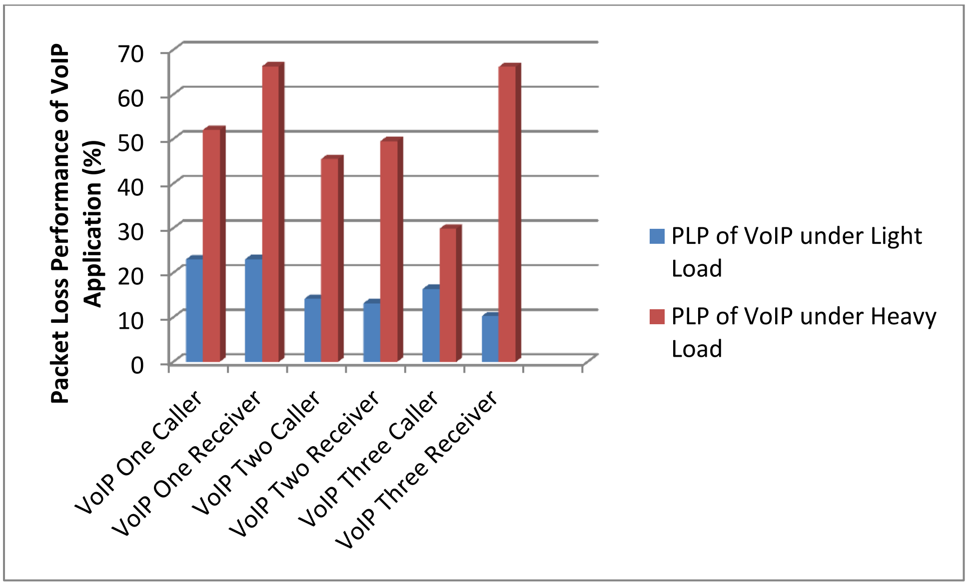

Figure 4.15: Packet Loss Performance of VoIP Application under Light and Heavy Loads

The graph in Figure 4.15 illustrates the result of VoIP PLP of our simulation in Scenario Three where we compared the performance of VoIP and Video Conferencing applications under Light and Heavy Load. The graph show that VoIP application under the Light Load have better PLP performance than that of the Heavy Load owing to the present of Best Efforts traffic in the Heavy Load Case. This result is a confirmation of the VoIP MOS performances in Figure 4.14 as earlier explained. Our conclusion in this case is that, the presence of Best Efforts traffic in the LTE network with multimedia application could cause poor performance of the multimedia applications.

4.4.6.2 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 performance based on the Packet Delay Variation, End-to-End Delay performance and Packet Loss Performances.

4.4.6.2.1 Packet Delay Variation of Video Conferencing under Light and Heavy Loads condition

The Table 4.12 below is a summary of the Video Conferencing PDV obtained from our Scenario Three under Light and Heavy Load.

Table4.12: Packet Delay Variation of Video Conferencing under Light and Heavy Loads Second.

Selected Video Conferencing users	PDV of Video Conferencing under Light Load	PDV of Video Conferencing under Heavy Load
Video_Conf_One_Caller	0.00176	0.1153
Video_Conf-One_Receiver	0.0043	0.2202
Video_Conf_Two_Caller	0.0157	1.8601
Video_Conf_Two_Receiver.	0.0103	1.8370

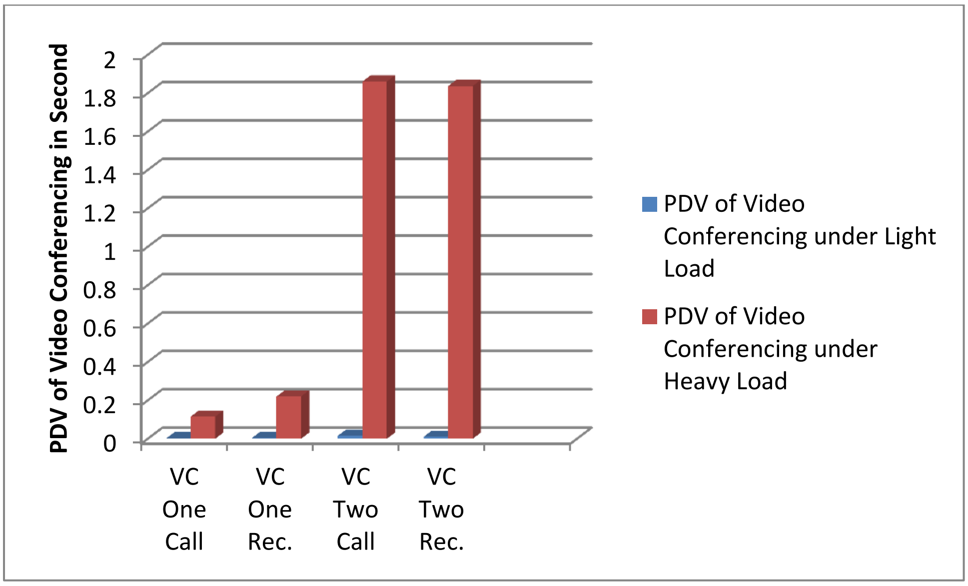

Figure 4.16: Packet Delay Variation graph of Video Conferencing under Light and Heavy Load.

Since our interest is in comparing the performance of Video Conferencing application under Light and Heavy loads, Figure 4.16 gives the PDV performance of Video Conferencing. From this figure, we see clearly that user under heavy load case have very high delay due to the presence of best effort traffic in that case. Considering the four selected UEs, we discovered that in all the four UEs, the PDV under heavy load are higher than the light load and in fact very high in some UEs as shown on the graph. We can conclude here that Video Conferencing performances are highly degraded by the presence of Background efforts in LTE network.

4.4.6.2.2 Video Conferencing Packet End to End Delay Performances under Light and Heavy Loads

We will also investigate the QoS performance of Video Conferencing in this section based on the Packet End-to-End performance of the Video Conferencing application under the light and heavy loads cases simulated in our scenario as earlier explained.

In the Table 4.13 below, we give the results of the Packet E2E delay of some selected nodes from the Light and Heavy Loads cases simulated in our scenarios. The Figure 4.17 gives the graphical relationship of the E2E performance of the selected Video Conferencing (VC) users under light and heavy loads in order to study the variation of their performances in these two cases simulated.

Table 4.13: Packet End-to-End Delay Performance in seconds of Video Conferencing application under Light and Heavy Loads

Some selected Video Conferencing Users	E2E of Video Conferencing under Light Load	E2E of Video Conferencing under Heavy Load
Video_Conf_One_Caller	0.0557	0.5841
Video_Conf-One_Receiver	0.0715	0.8205
Video_Conf_Two_Caller	0.1026	2.6152
Video_Conf_Two_Receiver.	0.1401	1.9949

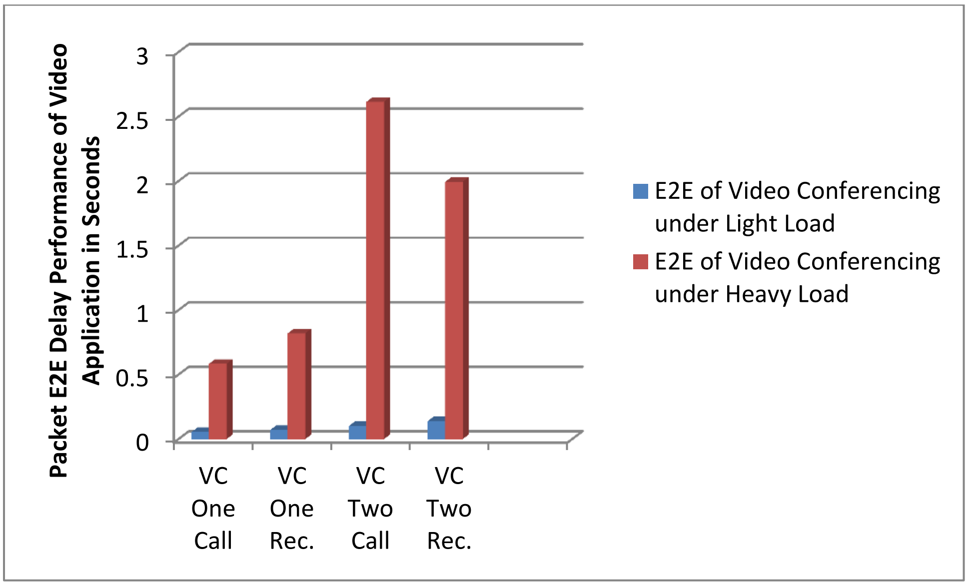

Figure 4.17: Packet End-to-End Delay graph of Video Conferencing under Light and Heavy Load traffics

The graph in Figure 4.17 is the Packet End-to-End delay performance of Video Conferencing application under Light and Heavy Loads traffic. From the graph, the E2E delay of Video Conferencing under heavy load are higher than that of the Light Load. In some of the UE examined i.e. VC Two Call and VC Two Rec, the values of the E2E are not acceptable to guarantee QoS for Video Conferencing users. This is because of heavy traffic present in the network created by Ftp and Http background applications.

4.4.6.2.3 Video Conferencing Packet Loss Performance under Light and Heavy Load traffics

In this section, we will compare the PLP of video conferencing under the Light and Heavy Load traffic cases simulated in our Scenario. The idea is to complement our findings in the earlier sections where we study the PDV and E2E delay performance under these stated traffic classes.

We collected the results from some selected nodes in the simulation and compute the PLP in order to compare the percentage of loss Video Conferencing users experienced in these two traffic classes simulated. The result obtained is shown in the Table 4.14 below.

Table 4.14: Packet Loss Performance (%) of Video Conferencing under Light and Heavy Load traffics.

Some Selected Nodes	**PLP of Video Conferencing under Light Load**	**PLP of Video Conferencing under Heavy Load**
VC One Call	27.11	66.28
VC One Rec.	34.06	36.96
VC Two Call	34.16	70.86
VC Two Rec.	34.54	83.66

The PLP result tabulated above is plotted into graph in order to compared the loss percentage of the two various traffics simulated as shown in Figure 4.18 below. From the result, we observed that the PLP of Video Conferencing under heavy load is higher than that of Light load in all the nodes selected. This gives a better picture of the effects of Load on Video Conferencing application in LTE network, which was the reason for this simulation.

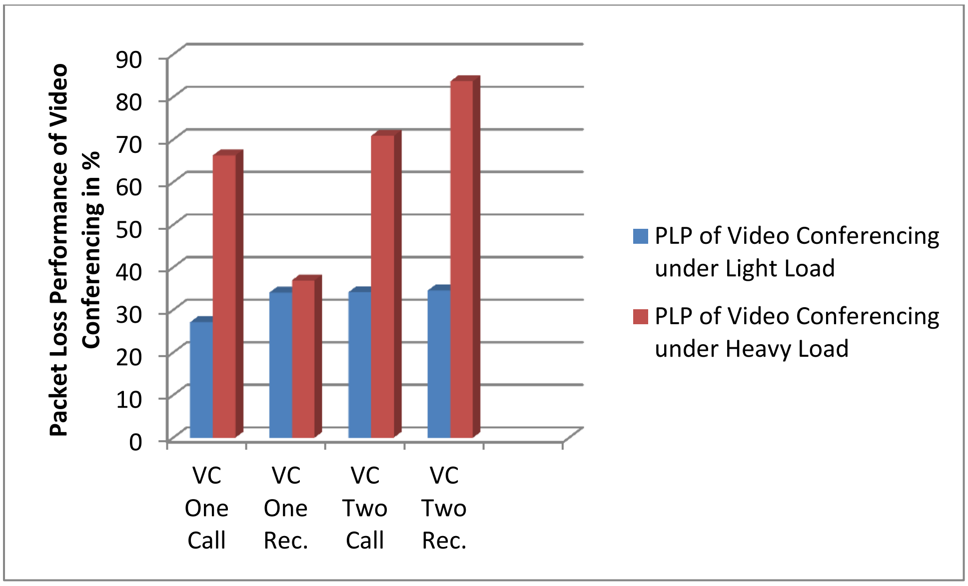

Figure 4.18: Graph of Video Conferencing Packet Loss Performance under Light and Heavy Load traffics

4.4.7 Scenario Four: LTE Traffic Class

The Opnet simulation environment below shows a single cell (1-eNodeB) set-up of multiple applications on an LTE-4G network.

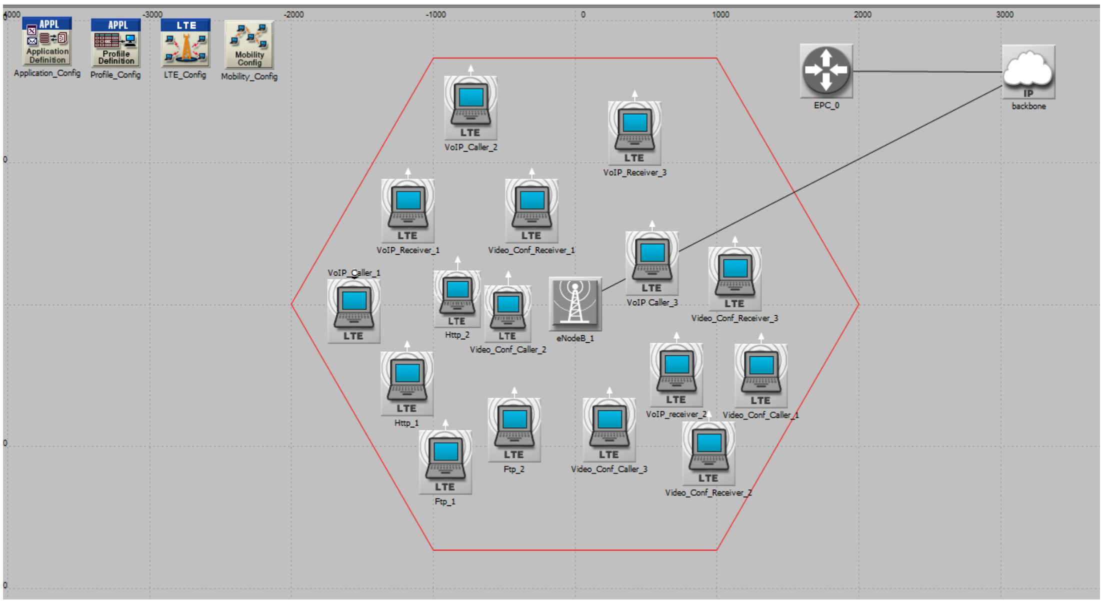

Figure 4.19: Opnet Simulation of LTE Traffic Class

OFDMA is the radio multiple access schemes used in the downlink subcarrier of the LTE network. This radio frequency used both time and frequency domains (TD and FD) to differentiate services along the LTE network according to their QoS constraints thereby guaranteeing QoS of applications based on channel condition and still maintains fairness among all the applications. In this scenario, our major interest is to investigate the importance of these prioritizations according to QoS requirements and ToS. In other word, according to [**22**] and [**18**] and as earlier explained in this chapter on default and dedicated channels of LTE network, two simulation cases were created in this scenario.

In Case1, VoIP, Video Conferencing, Http and Ftp applications are assigned to default channel with QCI 8.

In Case2, VoIP was assigned to dedicated channel with QCI 1; Video Conferencing was assigned to a dedicated channel of QCI 4 while Http and Ftp were both assigned to a default channel with QCI 8. In this case, after investigating the performance of VoIP and Video Conferencing applications under dedicated channel with GBR QCI, we further investigate the effects of Heavy download Ftp user over Light browsing Http application.

The Application and Profile configuration settings used in this scenario are the same with the earlier configuration in the previous scenarios. The mobility configuration is set to use the default random waypoint. The results of the MOS and Packet Loss Performance of VoIP as well as Packet End-to-End Delay Performance, Packet Delay Variation of Video Conferencing were respectively collected.

4.4.8 Results of LTE Traffic Class in Scenario Four

Our interest in this section is to investigate the behaviors of multimedia applications in relation to the LTE traffic class. The 3GPP has classified various applications over the LTE network to classes based on priority. Using ARP and QCI earlier explained in this thesis to group applications based on the TOS. In our scenario Four, we have simulated an LTE traffic class to investigate the performances of VoIP and Video Conferencing when they are grouped with Ftp and Http into a default channel and when they are allocated as GBR bearers into a respective dedicated channels, Ftp, and Http into default channel respectively.

In this section, we analyzed the performance of VoIP and Video Conferencing QoS performances based on the results obtained when they are allocated to default channel with Ftp and Http and when they are allocated to their respective channels based on ARP using ToS.

4.4.8.1 VoIP QoS Performances under LTE Allocation Based on Priority and Allocation Based on Shared Channel

We study the QoS of VoIP application in this section based on the MOS values, Packet Loss Performances and Packet E2E Delay when VoIP is grouped together with other application in a default channel and when it is allocated into the highest prioritized channel in the network.

4.4.8.1.1 MOS of VoIP under LTE Allocation Based on Priority and Allocation Based on Shared Channel

The table 5.13 below gives the results of the MOS performances of VoIP application users when they are allocated to the highest prioritized dedicated channel and when they are allocated into default shared channel with other applications in the network.

Table 4.15: MOS values of VoIP under Prioritized Channel and Shared Channel Allocations.

Selected VoIP Users	Channel Allocation on Priority	Channel Allocation to Shared default Channel
VoIP Caller 1	4.3289	1.1666
VoIP Receiver 1	4.2612	1.2442
VoIP Caller 2	4.1829	1.2388
VoIP Receiver 2	4.3521	1.2401
VoIP Caller 3	4.3070	1.2078
VoIP Receiver 3	4.2757	1.2520

The results of the MOS obtained from our simulation as tabulated in Table 5.13 above is plotted to see the variation in the MOS values of users under the two stated channel allocations system in Figure 4.20. From the graph, we observed that VoIP users allocated to dedicated channel score above 4.02 MOS and outperform those users allocated to shared channel. In fact, based on the MOS subjective call quality, the MOS of all the users in the shared channel are not acceptable for VoIP quality service in this scenario. We can conclude here that the poor MOS values of VoIP observed in shared default channel was because VoIP traffic was mixed with Http and Ftp which are best effort traffics there by degrade the VoIP performance.

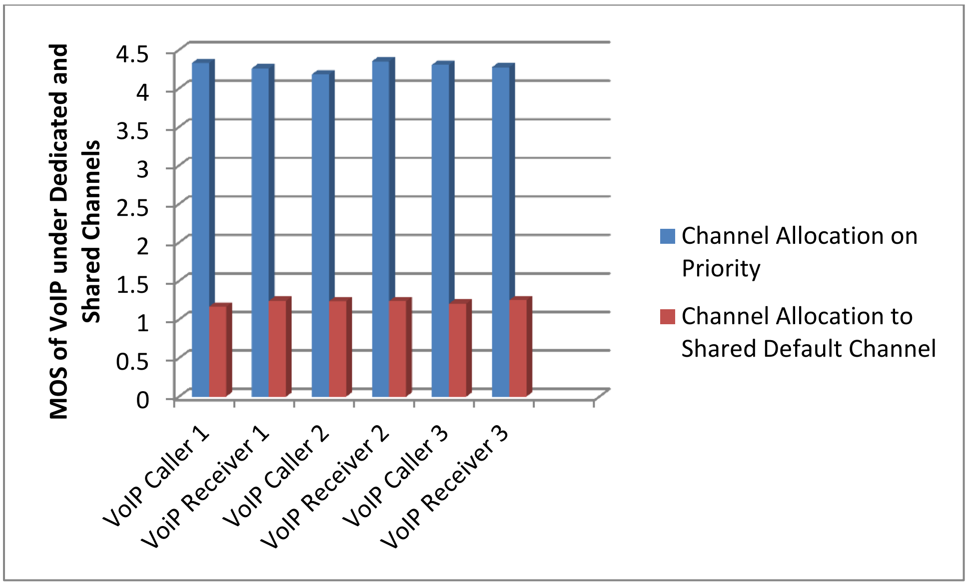

Figure 4.20: MOS values for VoIP under LTE Prioritized and Shared Channels Allocation

4.4.8.1.2 Packet Loss Performance of VoIP under LTE Prioritized and Shared Channels Allocation

In this section, we are interested in comparing the results of the PLP of VoIP users when they are allocated to a dedicated channel based on ARP and when they are mixed with Best efforts traffic in a default channel. We compute the results obtained from the simulation and tabulate the PLP as shown in Table 4.16.

Table4.16: Packet Loss Performances in (%) of VoIP Allocation Based on Priority and Allocation Based on Shared Channel

Some Selected Nodes	Channel Allocation on Priority	Channel Allocation to Shared default Channel
VoIP Caller 1	0.015	0.315
VoIP Receiver 1	0.015	0.043
VoIP Caller 2	0.058	0.976
VoIP Receiver 2	0.058	0.105
VoIP Caller 3	0.010	0.013
VoIP Receiver 3	0.194	0.467

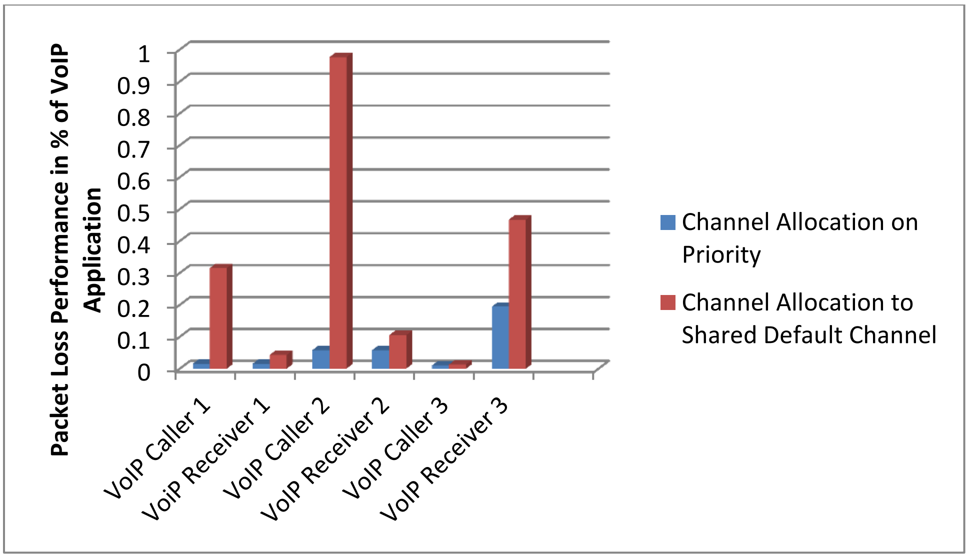

Figure 4.21: Packet Loss Performance of VoIP Allocation based on priority and based on Shared channels

The Packet Loss Performance of VoIP application in the two cases explained and simulated as illustrated above give better results. From the graph in Figure 4.21, the

PLP of allocation based on priority and shared channel does not rise above 1% in all the users in the simulation, although, the case of allocation based on priority still outperforms allocation to shared channel in terms of PLP. We can conclude here that the PLP of both cases are acceptable for VoIP QoS unlike the MOS earlier investigated that give clear different to the results obtained here. It is also interested to note that we cannot conclude on the end users QoS satisfaction based on only PLP alone. All others QoS parameters have to be satisfied according to 3GPP QoS standard.

4.4.8.2 Video Conferencing QoS Performances under LTE Allocation Based on Priority and Allocation Based on Shared Channel

We will study the results obtained from our simulation for Video Conferencing under the two cases simulated based on Packet Delay Variation and Packet End-to-End Delay Performance in this section.

4.4.8.2.1 Video Conferencing Packet Delay Variation Results under LTE Allocation Based on Priority and Allocation Based on Shared channel

The Table 4.17 below gives the summary of the PDV of Video Conferencing obtained from our simulation for allocation based on priority and based on shared channel.

Table4.17: Video Conferencing Packet Delay Variation (Millisecond) for Allocation based on Priority and Allocation based on Shared channel

Selected Video Conferencing Users	**Channel Allocation on Priority**	**Channel Allocation to Shared default Channel**
VC Caller 1	1.18	8.22
VC Rec. 1	2.29	3.01
VC Caller 2	0.90	8.15
VC Rec 2	1.28	4.10
VC Caller 3	1.11	3.78
VC Rec 3	0.79	1.78

Considering the Packet Delay Variation of Video Conferencing result in Figure 4.22, we observed that Video Conferencing performs better in both prioritized and shared channel unlike the VoIP performance earlier studied. The Packet Delay Variation result in the figure is plotted in millisecond which is the jitter observed by the Video Conferencing application in both cases simulated in the Scenario. Although, Prioritized channel shows better performance, but we can conclude here that the results of the Video Conferencing PDV obtained from the simulation are acceptable for Video Conferencing QoS end user satisfaction. This implies that the Video Conferencing response better to Load than VoIP application in terms of user satisfaction.

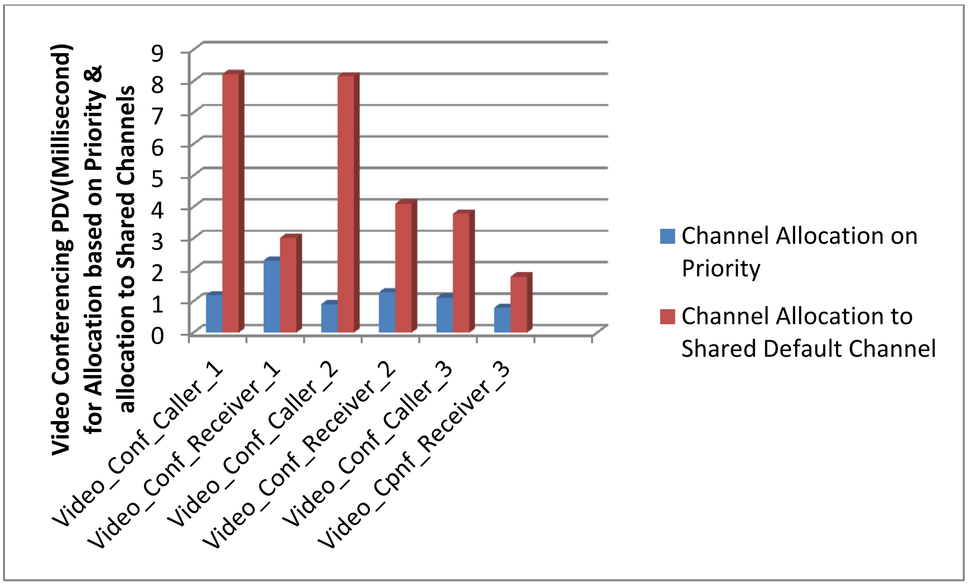

Figure 4.22: Packet Delay Variation of Video Conferencing under Prioritized and Shared Channel cases

4.4.8.2.2 Video Conferencing Packet End-to-End Delay Results under LTE Allocation Based on Priority and Allocation Based on Shared channel

We went further to study the Packet E2E Delay results obtained in our Simulation in Scenario Four. Just as we have used PDV to study Video Conferencing users performances in the two cases simulated as earlier explained, we tabulate the results of the Packet E2E Delay Performance obtained from our simulation under Prioritized and shared channels in Table 4.18 below and we plot the result in a graph as shown in Figure 4.23 in order to study the variation of end users E2E delay performances in both cases.

Table 4.18: Packet E2E Delay Performances in millisecond of Video Conferencing under Prioritized and Share Channel Cases

Some Selected Nodes	Channel Allocation on Priority	Channel Allocation to Shared default Channel
VC Caller 1	29	36
VC Rec. 1	28.9	47.1
VC Caller 2	21.6	32.1
VC Rec 2	26.9	33.5
VC Caller 3	47.1	69.2
VC Rec 3	22.6	50.8

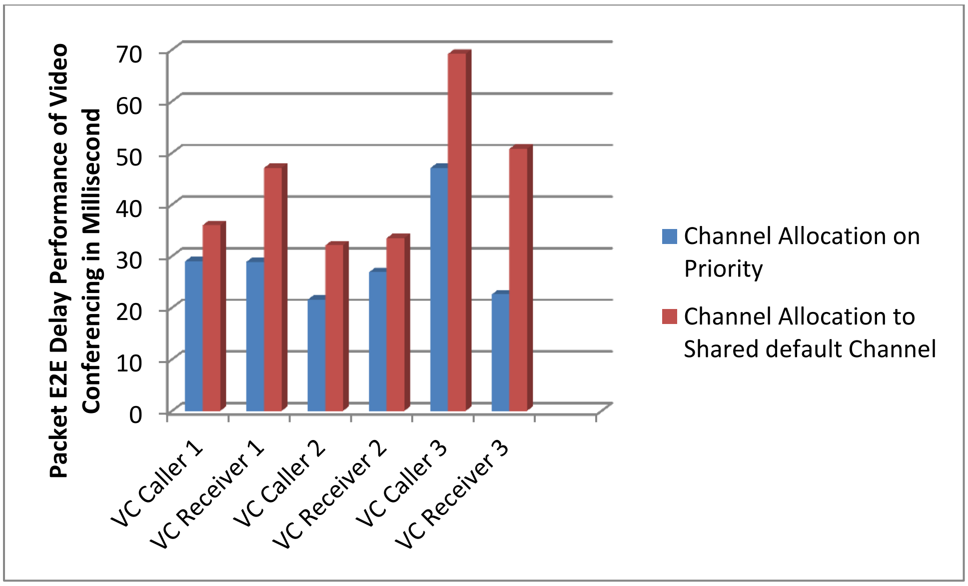

Figure 4.23: Packet End-to-End Delay Performance of Video Conferencing under Prioritized and Shared channels cases

In order to ascertain our PDV results explained in the previous section, we plotted the Packet E2E delay results as shown in Figure 4.23. The result shows an acceptable and user guaranteed QoS in both shared and Prioritized channels for Video Conferencing. The result plotted in millisecond shows that the highest user E2E delay to be 69.2 in both shared and prioritized channel. Although, the performance of Prioritized channel still give better E2E delay results than that of shared channel as shown in the figure. This is an acceptable result in this case, although, we inferred that if the Load in the scenario is further increased, there is likeliness of degradation of Video Conferencing performance.

Chapter 5

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.

From our findings, we were able to clarify that sometimes, PDV and E2E delay performance of mobile nodes in motion could outperform that of static nodes under the same network condition when a VoIP or Video Conferencing application is running on LTE network. The reasons for our findings was as a results of high amount of losses and HARQ retransmissions giving-up which cause less traffic to successfully transmits between sending and receiving nodes at the mobile nodes. While at the static nodes, more traffic are making it to the destination with more delay.

Our researches also clarify the effects of load on the performances of VoIP and Video Conferencing in LTE network. From the results obtained in our simulation when best efforts traffics were allowed to share the network, we observed unsatisfactory QoS performances for both VoIP and Video Conferencing users. VoIP and Video Conferencing applications show far better performances when they are allowed to run on the LTE network alone than while sharing the network with another background application.

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.

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 LTE network.

We set the load on our VoIP and Video Conferencing to a fixed value in this research. For further research studies, we recommend that the load could be varied to study how varying load responds to speeds in LTE. We further recommend that more robust allocation algorithms should be structured to managed background traffics influence on the performance of multimedia application on LTE network.

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