LTE SIMULATION ON OPEN SOURCE FRAMEWORK OMNET++

Capstone Project Report

July, 2014

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Abstract

Long Term Evolution (LTE) is the fourth generation of telecommunication technology standardized by 3GPP. It is developed to have high speed data support with low latency, higher data rates and maximum spectral efficiency. This is done by utilizing latest radio access technologies which include OFDM and MIMO. It is an IP based network with support for maximum number of applications.

Performance evaluation of LTE network in real world is costly. For research purposes, testing in a simulation environment is the best choice. There are different commercial packages like OPNET and free source simulators available, which include NS3 and OMNeT++. OMNeT++ is an open source framework for LTE support and is also very user-friendly as compared to NS3. All simulations in this research work are carried out in OMNeT++.

The primary objective of this research work is to evaluate the previous work done in this area and propose new improvements. Multiple scenarios are simulated for voice and video data to achieve these objectives.

Contents

Chapter 1

Introduction

This chapter is a brief introduction to the research objectives and scope of the report. The research methodology is explained later in the chapter to achieve the objectives of solutions to the problems.

1.1 Overview

The evolution of data over wireless communication started getting attention with the increase of internet users. The number of mobile subscribers is expected to increase as far as 3.5 billion by 2015 with the majority being smart phone subscribers [1]. Wireless communication which is high in demand has grown through different stages to support higher data rates and bandwidth-hungry applications. Long Term Evolution (LTE) is standardized by 3GPP in Release 8 [2] to support higher data rates for any type of application with the speed of wire. It uses advance radio access techniques and all IP based communication networks. The network infrastructure of LTE is completely different from the previous generations of wireless networks, thus requiring a complete deployment from scratch and new devices to support LTE communication. LTE is the 4G technology. The main challenge with 3G wireless technology was to support video streaming in real time applications with acceptable performance like IPTV, video-on-demand and voice over IP. LTE promises a peak data rate of 100 Mbps in the downlink using MIMO-OFDM and 50 Mbps in the uplink direction using SC-OFDM to increase the cell throughput and the spectral efficiency with supported bandwidth of 1.4MHz to 20MHz [3]. In this research, the performance of real time applications VOIP and VOD will be analyzed because of their importance to conclude the acceptable performance of a network. The effect of scheduling algorithms and increasing number of VOD and VOIP users will be measured in terms of delay, jitters and MOS values.

1.2 Scope of the Report

This report will cover issues related to factors that can affect the voice and VOD performance. Detailed literature review about the LTE architecture will also be discussed. The technical details of LTE implementation with VOIP and video application in simulation environment of OMNet++ will be provided and different results will be critically evaluated.

Following are the aims and objectives of this report:

- Study the quantitative approach to discuss issues related to LTE architecture for real-time applications;
- Study literature about the LTE architecture and real-time applications;
- Understand the open source OMNet++ structures and simulation environment;
- Understand the open source LTE code developed for OMNet++;
- Deploy LTE network inside OMNet++ with VOIP and VOD applications configured on UEs; and
- Analyze the different performance parameters for real time applications and LTE network

1.3 Research questions

To follow the quantitative approach for solving technical issues in this report, the research questions must be established beforehand. The following research questions will be answered in later parts of the chapters.

- How does the LTE network behave with the increase in number of users?
- How does scheduling algorithms' variation affect on VOIP and VOD performance by keeping the number of users and AMC mode constant?
- How does different modes of AMC affect the VOIP and VOD performance by keeping the scheduling and number of users constant?

1.4 Research methodology

Quantitative research approach will be followed to find solutions and the steps taken to perform the research in this report are explained in [4].

- It is observed that enodeB has limited bandwidth. With the increase in number of UEs in the network, the performance of network degrades.
- The research questions are proposed in the previous section.
- Create a test bed to answer the research questions.
- Analyze the results and make conclusions.

Chapter 2

LTE Architecture

This chapter will explain in detail the LTE architecture. Different parts of LTE network that will be explained in detail are; the radio access layer and the core layer.

2.1 Background

Recently, the growing demand of higher data rate applications such as; multimedia gaming, IPTV, streaming contents, and video on demands, have inspired the 3rd Generation Partnership Project (3GPP) to develop Long Term Evolution (LTE). LTE is the latest standard in the mobile wireless communication technologies hierarchy which formerly released the GSM, EDGE and UMTS cellular network technologies that now cover 85% of all mobile subscribers. LTE will guarantee 3GPP's influence over other mobile network technologies [5]. The first release by 3GPP was in December 1998 and was based on 2nd generation cellular system, i.e. GSM which was then advanced to 3G system and is now known as UMTS. The important steps taken by 3GPP involve: improving standards for UMTS to handle the evolving future requirements such as services advancement, developing the spectrum facilities, decreasing costs, efficiency enhancement and improved integration with other standards. The table below shows the technical specifications of all the releases by 3GPP.

Table 2.1-1 Technical specification of LTE

Release	Specification	Date	Downlink Data Rate	Uplink Data Rate	Round Trip Time
99	WCDMA	March, 2000	384 Kbps	128 Kbps	150 ms
4	TD-SCDMA	March, 2001	384 Kbps	128 Kbps	150 ms
5	HSDPA	March to June, 2002	14 Mbps	5.7 Mbps	<100 ms
6	HSUPA	December, 2004 to March, 2005	14 Mbps	5.7 Mbps	<100 ms
7	HSPA	December, 2007	28 Mbps	11 Mbps	<50 ms
8	LTE	December, 2008	100 Mbps	50 Mbps	10 ms

10	LTE-advance	Published 2012	1 Gbps in low mobility	375 Mbps	

2.2 Performance of LTE

LTE, whose radio access is called Evolved UMTS Terrestrial Radio Access Network (E-UTRAN), has significantly improved the cell capacity and throughput of user equipments and decreased user plane latency. Also, it has resulted in considerable improvement in user experience with full mobility. With the implementation of all IP based traffic, LTE supports scheduling of all IP-based traffic guaranteeing end-to-end Quality of service (QoS). IP-based voice traffic in LTE provides better IP over multimedia integration.

E-UTRAN supports different types of services and applications which include: FTP, HTTP, video streaming, online gaming, real time gaming, push to talk and push to view and all other popular applications, developed for smart phones. Therefore, LTE has been designed to support high data rates and low latency, with specifications provided in the table below. The bandwidth ability of UE in LTE is required to be 20MHZ for both; transmission and reception [6]. However, the service provider can deploy the LTE with any of the spectrum given in the table, which relates the bandwidth cost and the supported data rates. This provides flexibility to the service providers to go for higher frequencies in areas where larger data rate is required. LTE supports power control features using DRX and has low cost and backward compatibility with UMTS. For handover in real time services, the handover time between E-UTRAN and UTRAN/GERAN must be less than 300ms as per ITU requirements for VOIP and interactive videos. On the other hand, in-case of non-real time services, the handover time must be less than 500ms.

Table 2.2-1 LTE network Requirements

Metrics	Requirements
Peak data rate	DL: 100Mbps UL: 50Mbps (for 20MHz Spectrum)
Mobility support	Up to 500kmph but optimized for low speeds from 0 to 15 km/h
Control plane latency (Transition time to active state)	< 100ms (for idle to active)
User plane latency	<5ms
Control plane capacity	> 200 users per cell (for 5MHz spectrum)
Coverage (Cell sizes)	5 – 100km with slight degradation after 30km

Spectrum flexibility	1.4, 3, 5, 10, 15 and20MHz

Ericsson has carried out various simulations of link level and system level of LTE. Figure 2.2-1 shows the simulation spectrum efficiency and user throughput [7].

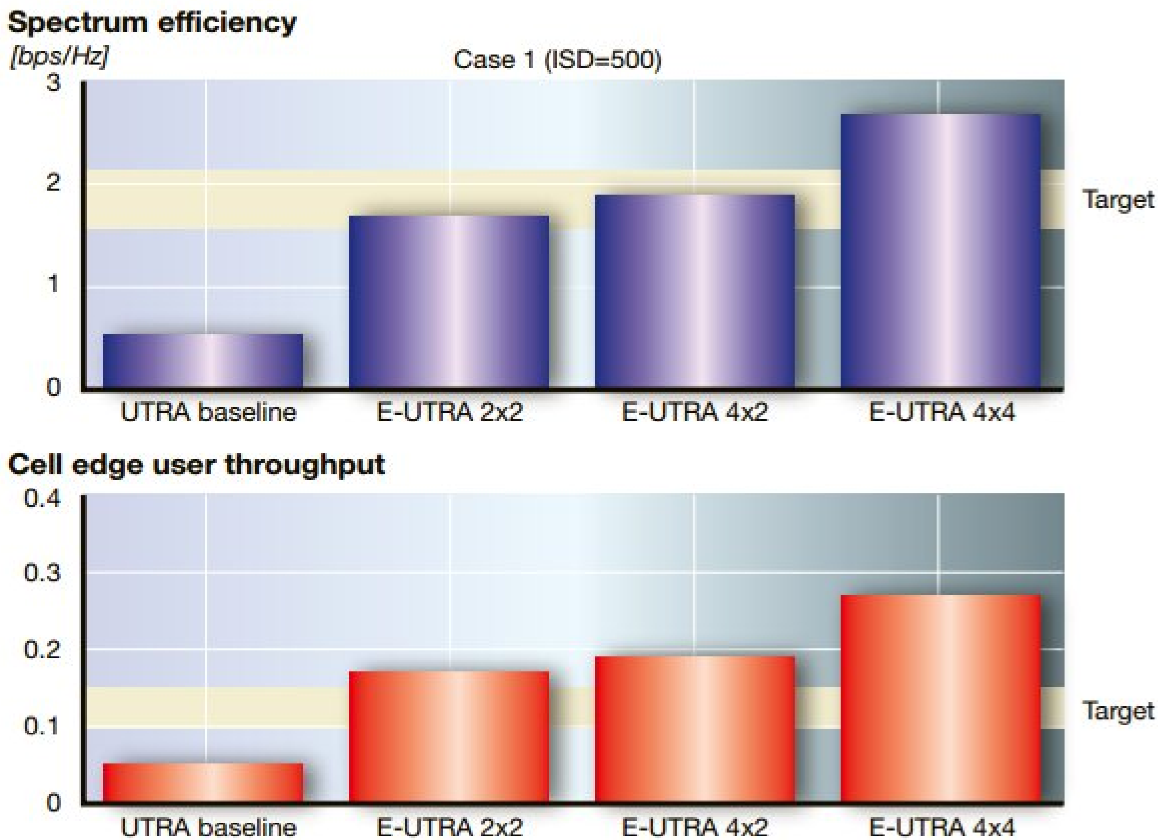


Figure 2.2-1 spectrum efficiency

2.3 Architecture overview and Core network

The LTE System Architecture Evolution (SAE) includes the following main principles.

- A gateway (GW) or common anchor point for all access technologies;
- An enhanced design for the user plane - to reduce the previous four entities of cellular networks to only two network elements (NEs): base station and a gateway;
- All IP based interfaces;
- RAN-CN functionality, split in the same manner as in WCDMA/HSPA;

- Control/user plane, split between the mobility management entity (MME) and the gateway; and
- Integrated non-3GPP access clients, using client and network based mobile IP

A simplified overview of the complete LTE-SAE is provided in Figure 2.3-1. The gateway can perform the role of Packet Data Network (PDN) and serving gateway. It can be configured to serve as both or any of the two. The PDN gateway functions as a common point for any type of access technology, which helps in providing a stable IP point of presence regardless of mobility within or between access networks. The serving gateway is the common point for intra 3GPP mobility. The functionality of MME has been kept isolated from the gateways to serve as the network deployment, independent progress of technology and flexible improvement of capacity. WCDMA/HSPA and GSM systems have been combined into the evolved core through standard interfaces between the SGSN (Serving GPRS Support Node) and the Evolved Packet Core (EPC). It includes MME interfaces for transmitting data and forming bearers when moving in different access technologies. IP connectivity is being established with the gateway using these interfaces. The gateway node is thus acting as the GGSN (Gateway GPRS Support Node) for GSM and WCDMA/HSPA terminals. This architecture also combines GSM, WCDMA/HSPA and LTE into a common packet core network through the combination of SGSN and MME in the same node.

The Home Subscriber Server (HSS) connection interface to packet core will be based on Diameter, not SS7, which will reduce the complexity of the control plane. Also, it will give a simplified solution for control plane IP networking, because the network signaling for charging and policy control are already based on Diameter. The base stations in LTE connect directly with core network through RAN-CN interface. The function of MME is to handle the control signaling e.g. for mobility. User data flow between the BS and gateway is over IP based transport network. To support high speed handover, all the base stations are logically connected with each other and are accessible via an IGP running in the network.

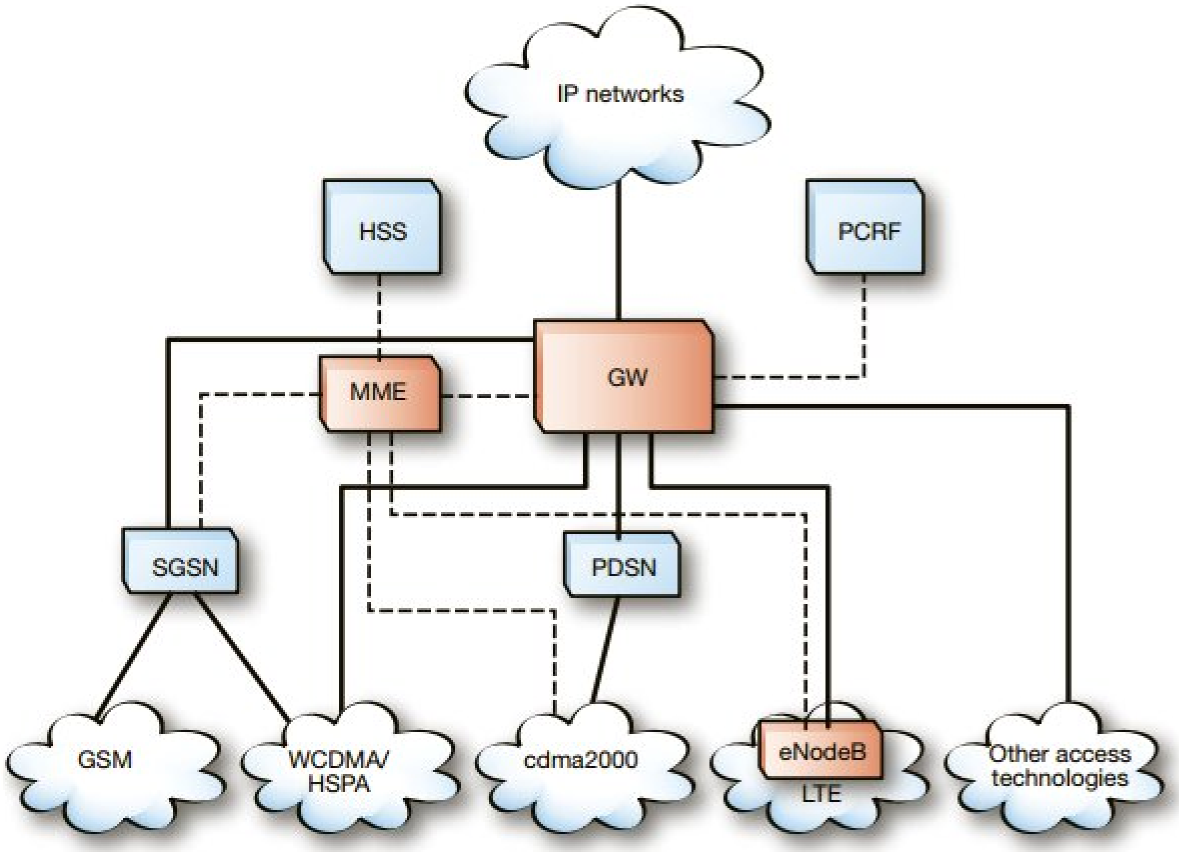


Figure 2.3-1 LTE-SAE [7]

2.4 Access network

3GPP has chosen the Orthogonal Frequency Division Multiplexing (OFDM) for downlink and Single Carrier OFDM (SC-OFDM) for uplink after evaluating different radio access techniques for LTE. Access network of LTE consists of enodeBs, which are connected through S1 interface to EPC and X2 interface with each other. An internal routing protocol is required for connectivity between enodeBs [8].

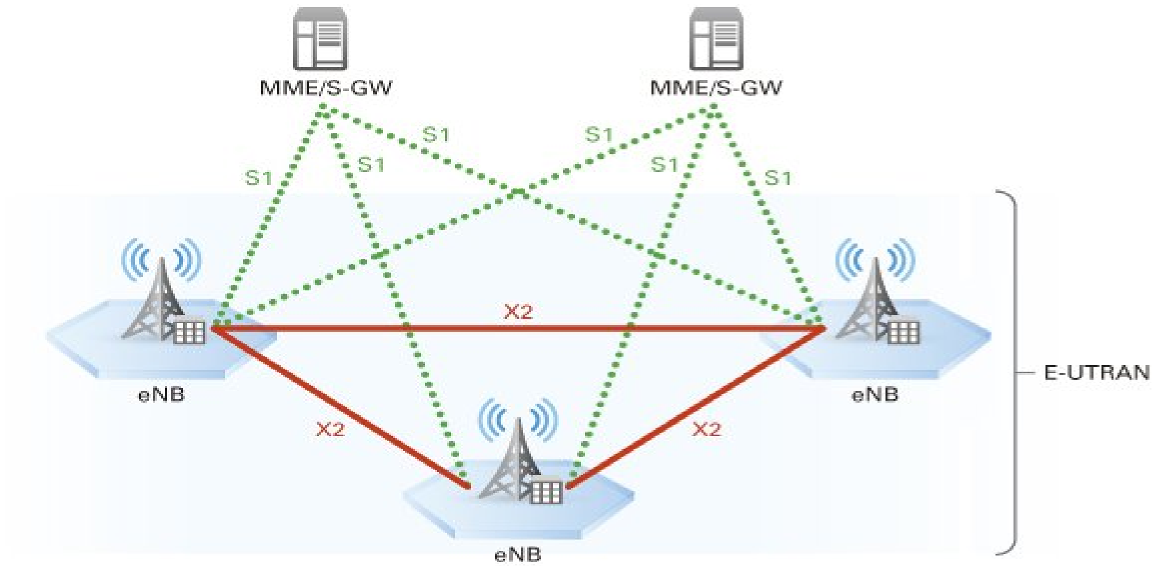


Figure 2.4-1 LTE access network [9]

2.5 Protocol Architecture

The LTE radio is based on high speed packet access (HSPA) architecture. The names and functions of the protocols are similar to HSPA. The only difference lies in the radio access technology protocols. Figure 2.5-1 below shows the LTE user and control planes protocols.

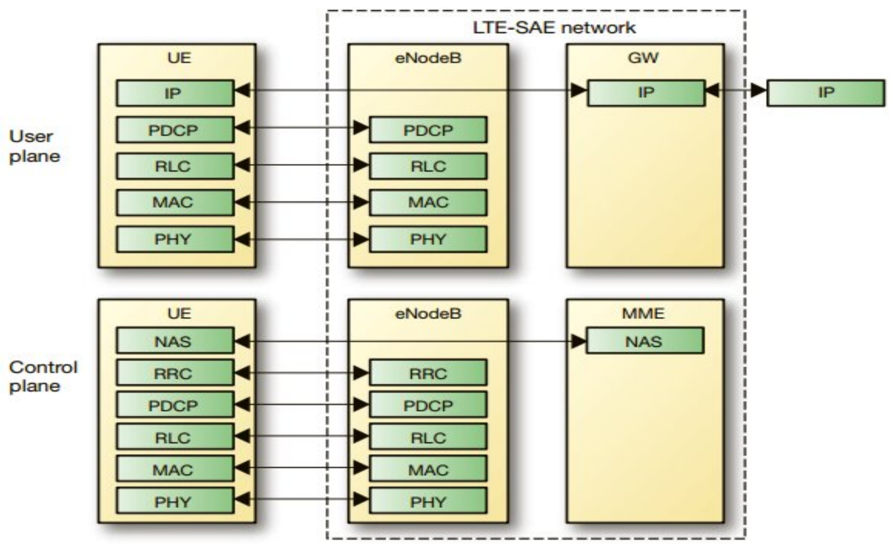


Figure 2.5-1 LTE user and control plane [7]

2.5.1 NAS layer

The Non Access Stratum (NAS) protocol runs in between the Mobility Management Entity (MME) and User Equipment (UE). The NAS is used in the control plane for network attachment, user authentication, mobility management and setting up a bearer. Messages from NAS are protected by UE and MME.

2.5.2 RRC layer

It is also for control purpose and is used for Radio Resource Control (RRC) between the enodeB and the UEB.

2.5.3 PDCP layer

Packet Data Convergence Protocol (PDCP) works in both control and user plane. It is responsible for managing the uplink and downlink.

2.5.4 RLC layer

Radio Link Control is used for the traffic between UE and enodeB. Three different types of reliability modes are provided by RLC, which are; transport mode, acknowledgement mode and acknowledgement mode [10].

2.5.5 MAC layer

Media Access Control layer connects RLC and physical layers, it provides the connectivity between the logical channels of RLC to the physical layer transport channels. MAC layer performs the multiplexing and de-multiplexing, HARQ, control and logical channel prioritization.

2.5.6 Physical Layer

Data is moved to the physical layer after it has been processed by all of the above layers. The lowest time unit of 1ms is allocated for transmission. LTE physical layer performs RF processing, i.e. modulation and demodulation while handling time and frequency synchronization. It has the capability of providing detection against channel errors by using Adaptive Modulation and Coding (AMC) techniques. It performs CQI calculation and sends the report to upper layers.

2.6 Mac scheduling in LTE

The purpose of the scheduler is to maximize the spectral efficiency and maintain different levels of QoS for different applications. The MAC scheduler exists in enodeB and makes fast decisions with TTI of 2ms. There are different types of schedulers whose implementation depends on the vendor. The schedulers that are available in OMNet++ are Deficit Round Robin (DRR), Proportional Fair (PF), and maximum C/I [11]. The enodeB assigns transmission block or slot to UE on the basis of the following criteria.

- CQI report sent by UE about the channel condition;
- Traffic class, Traffic Handling Priority (THP), Allocation Retention Priority (ARP), scheduling priority indicator of the QoS parameters;
- UE queuing report e.g. the amount of data in buffer; and
- Available system resources e.g. transmit power and bandwidth

2.6.1 Round Robin

This is the simplest type of scheduler and provides TTI equally to all the users in the network without considering channel condition and QoS parameters. It does not guarantee maximum spectral efficiency as the UEs having good channel condition cannot maximize their throughput.

2.6.2 MAX C/I

The maximum C/I algorithm maximizes the spectral efficiency of the system by providing resources to users having good channel condition. The instantaneous channel condition received by all users in the network is different. It provides users having good channel condition with ability to transmit data at any time. A system running this algorithm has the best throughput as compared to other scheduling algorithms but it results in unfairness to users at the cell edges.

2.6.3 Proportional fair

The Proportional Fair (PF) also considers the user channel condition but in addition to that, PF also maintains fairness for users under bad channel condition. Thus it has the properties of both RR and Max C/I. PF allocates resources to different users on the basis of following equation.

Pi=(IBi[n])/(ABi[n])

Where

Pi is the priority of user ‘i’ for the current transmission turn, IBi is the instantaneous bit rate of user ‘i’ and ABi is the average throughput of the connection for user ‘i’.

Chapter 3

LTE in OMNeT++

This chapter is a brief introduction of open source network simulator package OMNeT++ and the LTE-SIM available in it for LTE simulation.

3.1 OMNeT++

OMNeT++ has a Discreet Event Simulator (DES) based engine using underlying object oriented language for programming and simulating different types of wired and wireless networks with a rich support of built-in protocol suits. It also provides an interface for modifying and adding new protocols. It is primarily designed for the simulation of communication networks but due to its flexible architecture, it can also be used for simulation of complex IT systems, queuing and hardware theories.

OMNeT++ provides a component based architecture, which can be combined and imported in the project for modeling different designs using a higher level language (NED). It also has a well-developed Graphical User Interface (GUI) and due to its modular architecture, the models can be easily integrated inside the applications. OMNeT++ has the following components [12].

- Simulation Kernel library;
- NED topology description language;
- Eclipse based OMNeT++ IDE;
- GUI support;
- Command line support; and
- Extensive documentation for help

3.2 LTE-SIM

LTE-SIM is an open framework developed for OMNeT++ for performance evaluation of LTE networks and it is compatible with iNet framework of OMNeT++. Various functionalities of the LTE EPC and EUTRAN are entirely developed in LTE-SIM. In EUTRAN, the enodeB is modeled while in EPC, PGW and SGW are implemented. Both user and control planes for LTE are developed. In user plane, RLC, PDCP, MAC and physical layers are modeled. Some functionalities of control plane are also modeled with user PDCP layer.

As discussed in the previous section, LTE access layer is OFDM based, which can work either in frequency mode or time mode. In LTE-SIM the LTE time based resource block allocation is modeled with 0.5ms allocation interval.

3.3 EUTRAN of LTE-SIM

In OMNeT++, nodes are implemented in protocol stack with each protocol working as separate object. The enodeB and UE are implemented with the same components, which are; PDCP_RRC, RLC, MAC, and PHY. LtePdcpRrc.ned consists of modules for implementing PDCP and RRC features for UE and enodeB of LTE network. It has support for VOIP, gaming and video applications. It performs header compression and decompression for the traffic received from upper layers and calculates statistics for end-to-end delay and throughput. Ports (in out TM_Sap, in out AM_Sap, in out UM_Sap) are used for receiving and delivering the packets to different types of RLC units (AM, UM or TM mode). RLC and PDCP layers are connected through relays. When packets arrive from upper layer, compression is performed by header compress class, while the set traffic information class chooses the type of RLC required for transmission of packets in downlink. The LTE-SIM functionality of different layers is provided below.

3.3.1 RLC layer

This layer is implemented in LteRlc.ned compound module. Different applications have different Quality of Service (QoS) requirements to be supported by RLC. TM, UM and AM modes are implemented in RLC as LteRlcTm, LteRlcUm and LteRlcAm to handle different QoS requirements. Packets are fragmented for UM and AM mode after being received from upper layers to the size defined by the MAC layer, and to support this, transmission and reception queuing classes are defined. Reliable transmission is required in AM RLC mode to support the windowing and ACK features, which are implemented under simple AmRxQueue. Different performance values for uplink and downlink, such as RLC delay, RLC throughput, packet loss are modeled to check the network health.

3.3.2 Mac layer

This layer has been designed for both UE and enodeB. The important feature of LTE MAC layer is link adaption which implies that coding of data sent on link changes as per the SNR ratio of the signal received on that link. If SNR is low, higher-order modulation schemes with high spectral efficiency are used to achieve higher data rates, else lower order modulation with low spectral efficiency is used. To support this feature, AMC (Adaptive Modulation and Coding techniques) module is implemented, which can handle different types of coding techniques (auto, piloted, das and multi). LTE access technologies are based on OFDMA; LteMacEnbclass has a parameter “rballocationtype” which is designed to properly allocate resources among different UEs without wasting resources. Power model is implemented in LteMacEnb for different types of frames (paging, syncro, normal etc.). The LteMacBase class is implemented to model the performance of MAC layer parameters like throughput delay, HARQ error rate, buffer overflow, and MAC packet loss.

3.3.3 Physical layer

Physical layer parameters of enodeB and UE are modeled in LtePhy.ned, which assigns physical channels to both, UE and enodeB. LtePhy.ned has properties such as; eNodeBTxPower, relayTXPower, ueTXPower, microTXPower, userpropagationDelay, and channel model. LTERealisticChannelModel.cc class has different supported features to model the physical layer of LTE; height of antenna, height of building, correlation distance, antenna gain, thermal loss, cable loss, noise figure, carrier frequency, (default of 2GHz) and various other features to provide a realistic view of LTE simulation. Interference due to other cells and path loss is also measured.

3.4 Core of LTE-SIM

None of the access network technologies is attractive to user without supporting connectivity to the internet. The EUTRAN internet access is provided through EPC which provides a gateway to transfer packets from LTE domain. There are two types of gateways that are modeled; service gateway and packet gateway. Full functionality of these two gateways is not implemented but the basic function of sending data over the tunnel towards the packet gateway has been modeled. Two compound modules are modeled specifically for Traffic Flow Filter and GTP-U (GPRS Tunneling Protocol- user).

3.4.1 Traffic flow filter (TFT)

TFT is a classifier for matching the inner packets of GTP tunnel to provide different levels of radio bearer performance. Packets can be matched and classified, based on the source address, IP protocol type, type of service and source ports, and destination port range. TFT can be used in static or dynamic form. In SIM-LTE, static mode of TFT filtering has been modeled. Packets of dedicated bearer are created after filtering and traffic is assigned to the bearer. The required QoS is then provided to the traffic class. TFT has a support for IPV4 only. Packets are classified according to the source and destination addresses. Currently, port based filtering is not supported. TFT-IDs are assigned after filtering of the packets, which are then used by the GTP-u to route packets in the tunnels. EnodeB filtering table is indexed initially by the source address, so, either source name or source address must be mentioned. At the P-GW, filtering table is accessed by destination address. TFT preserves a table for mapping traffic as per destination IP address.

3.4.2 GTP-U

It is used for carrying data between radio and the core network. The user data can be of type IPV4, IPV6, and PPP for transportation. GTP header has a 32-bit TEID (Tunnel Endpoint Identifier) field, which is used for the identification of different connections within the same tunnel. Each user may have multiple tunnels between the same endpoint for various applications to have different QoS. IP packets of UEs are being encapsulated in GTP-U packets and tunneled among the P-GW and the enodeB for transmission with respect to UE over S1-U and S5/S8 interfaces. GTP-U keeps two tables, which are used for the mapping of TFT to TEID ID and for

TEID to TEID table. Packets are filtered from different traffic flows and then they are mapped to specific bearers for being tunneled outside the network.

3.4.3 Traffic models

The most important driving force for the development of LTE was the support for bandwidth-hungry, delay-sensitive, real-time applications, which were not supported by previous technologies. LTE-SIM supports three crucial applications to test the implementation of LTE simulation, namely; VOIP, Gaming, and VOD. LTE voice is transported over the IP packets, after which the analog voice is encoded in voice encoder, packetized, and sent over the IP network. Reverse process is performed at the destination. Similarly, VOD is another application of LTE modeled in LTE-SIM. Video and voice packets are treated differently by various layers and different QoS is provided.

3.5 Additional modules

3.5.1 Configurator

LTE is a completely IP based technology and in OMNeT++, IPV4 addresses are assigned using IPv4NetworkConfigurator module, which is required in LTE network. It also checks for duplicate addresses in the network. Users provide parameters (in form of xml files) to this module and assign IP to the network.

3.5.2 LTE Binder

It binds the IP address of the node to its ID and maintains it in a table which is used by sender to find the ID of the destination node during communication.

3.5.3 LTE Deployer

This module is used to change the radio parameters of the LTE access network. Channel numbers per band, OFDMA symbols per slot, signaling symbols per channel; all these parameters can be modified to test the behavior of LTE networks in different scenarios.

Chapter 4

Results and Analysis

This chapter will provide detailed analysis of the network's performance on the basis of research methodology explained in the previous chapter. Network parameters will be discussed and performance will be analyzed in the form of graphs resulting from the simulation of the network. Initially, an insight to the previous research work will be provided and later in this chapter, the modifications implemented in this research work will be explained in detail.

4.1 Performance metrics for real time applications

Real time applications like VOIP and interactive videos are very sensitive by nature thus proper QoS and SLA are provided to these applications. The performance is characterized by end-to-end delay, jitter, and traffic drop rate. For an acceptable range, the end to end delay value should be less than 300ms, the jitter should be less than 30ms, and the traffic drop rate should be less than 1% [13].

4.2 Previous work

In [14] the performance of LTE network is observed by increasing the number of clients in the network which resulted in identifying many flaws in that design. These new simulations aim to remove those flaws. The LTE has two main components as discussed previously; the access network and the core network. In OMNeT++, access network can be evaluated separately or in combination with the core network. The core network of LTE is completely IP based and currently, the metro Ethernet nodes support 10G links. And, in near future, the 100G links will also be supported; so, the main bottleneck still exists in the wireless access network. Following are the main flaws in the previous work:

- The theoretical DL bandwidth per user in LTE network is 100Mbps, while the researchers have taken 10Mbps backhaul links;
- LTE standard is developed for long term and for supporting maximum number of users, while in the simulation, a network of just forty nodes degrades the performance. This is actually not due to the LTE bottleneck, but due to the core network bottleneck;
- The applications running under the single enodeB are not uniform; one voice and 40 VOD users have been considered; and
- The path loss model is URBAN MACROCELL and the movement of nodes has been restricted up to 30m of the enodeB.

4.3 Previous work simulation with 100 Mbps links

The previous work is simulated by taking the 100 Mpbs links instead of 10 Mbps and the same parameters as stated in [14]. The core network parts; SGW and PGW, are not being considered in our design as the main focus is on checking the radio network instead of core network. Also, it has been observed that the unacceptable performance and limitation in less number of nodes was due to 10Mbps links in the core. Under the single enodeB, the number of clients is increased up to forty and the same number of voice and VOD users is considered. The network topology for the design is shown in the Figure 4.3-1.

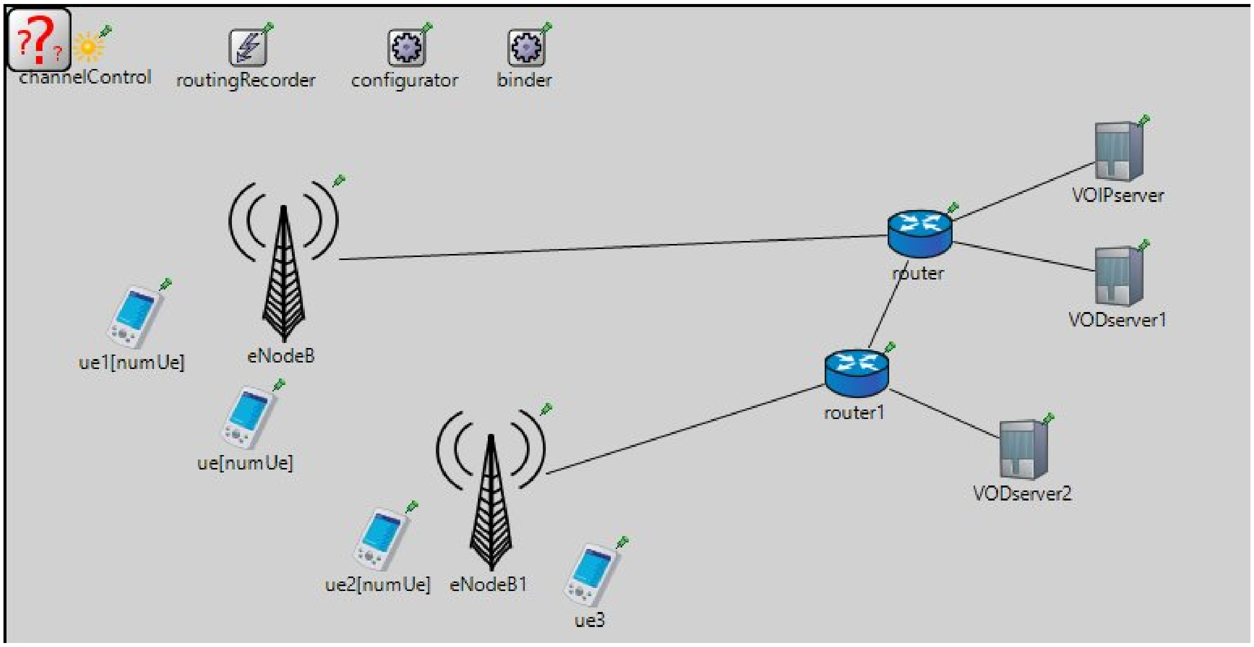


Figure 4.3-1 Network Design

VOD server1 is for enodeB1 and VOD server2 is for enodeB2. The simulation is carried out for 20, 30, and 40 UEs and the results are collected. It is found that by increasing the number of clients up to 40, the LTE network performance is same as for 20, because, for URBAN cell, only 40 clients within the 30m range should have this performance and it has been verified from the graphs shown below.

In Figure 4.3-2, the end-to-end delay for VOD is shown. The red line shows the end to end delay for 40 clients, green for 30 and blue for 20. The red line shows a little increase in the values but all are within the acceptable range as provided in performance metrics and the average is about 15ms.

MOS is shown in Figure 4.3-3. It has been found that, for 20, 30, and 40 the MOS is same and even above 4, which is the best acceptable value for voice call. Moreover, it has been found that the results for increasing number of clients are approximately the same as for less number of nodes. Figure 4.3-4 shows the frame delay, which is less than 300ms and the effect of increasing number of UEs has not been observed again. Figure 4.3-5 shows the throughput and is same for 20, 30, and 40 number of clients.

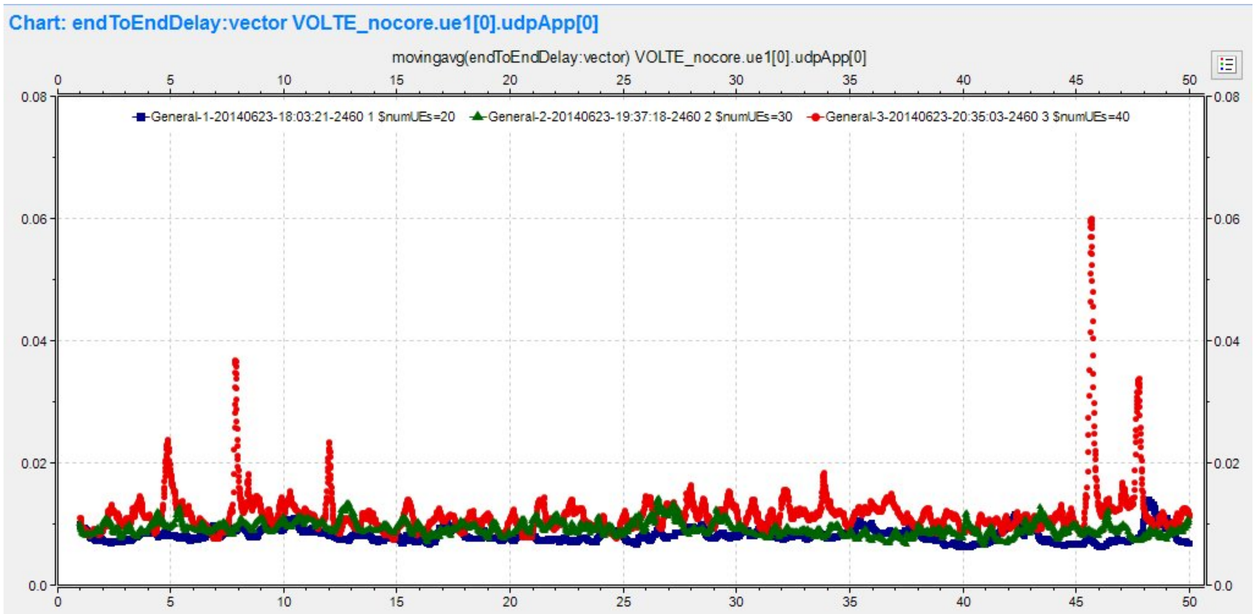


Figure 4.3-2 E2E VOD, time in Second on x-axis and delay in second on y-axis

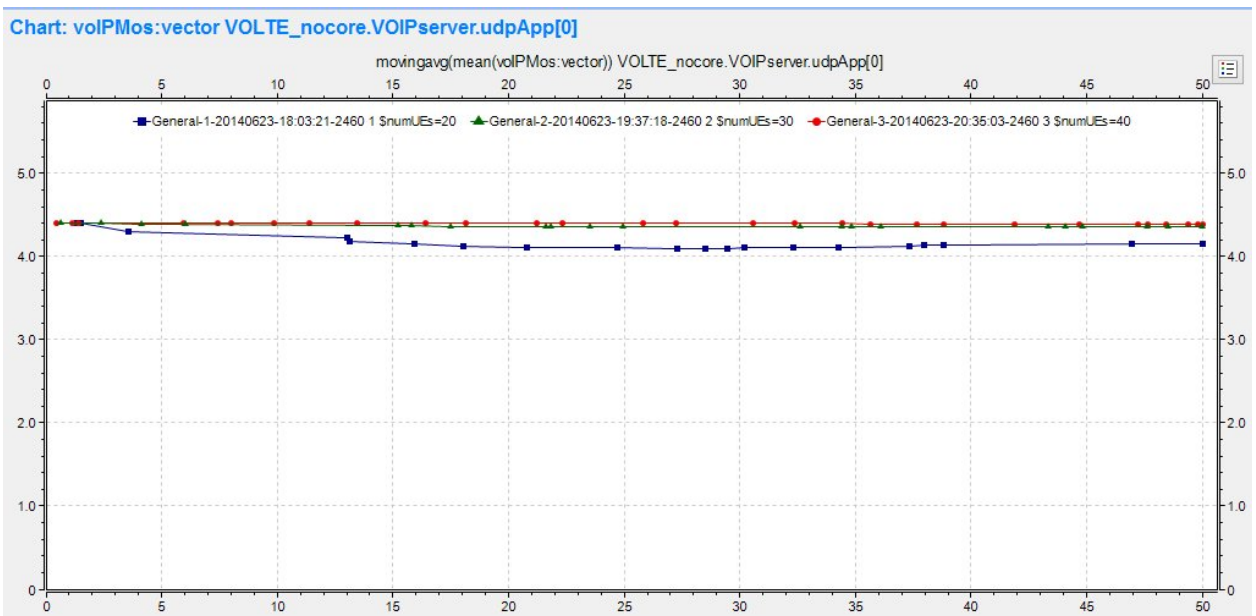


Figure 4.3-3 MOS values, time in second on x-axis and MOS on y-axis

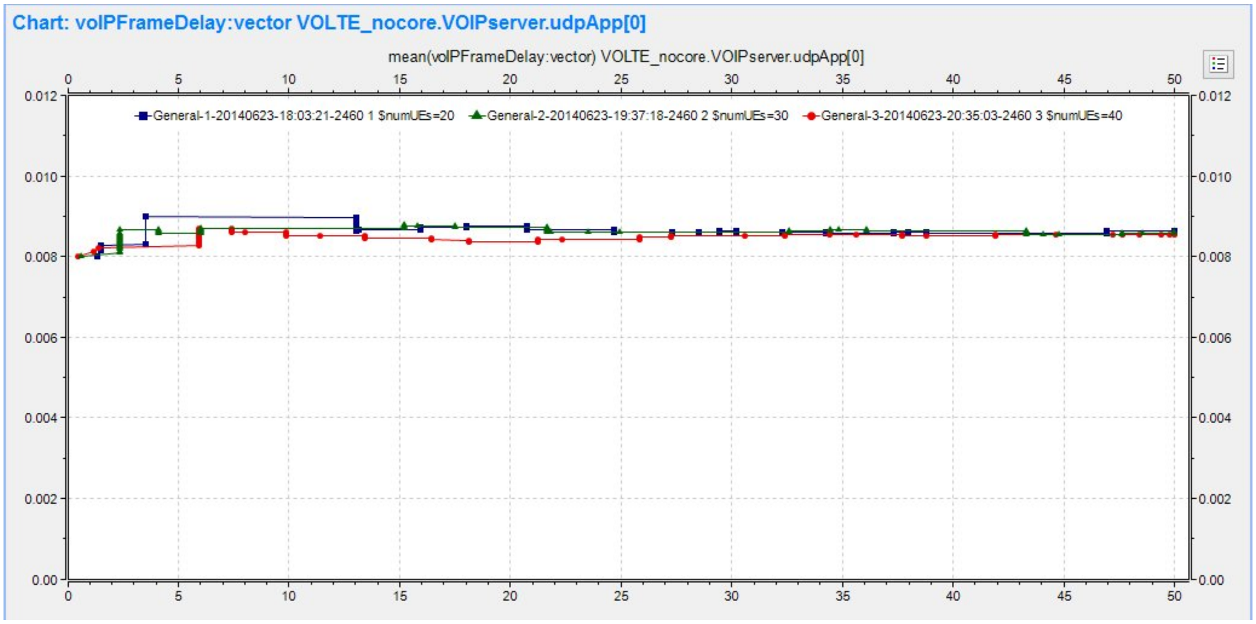


Figure 4.3-4 Frame delay, time in second on x-axis and frame delay in second on y-axis

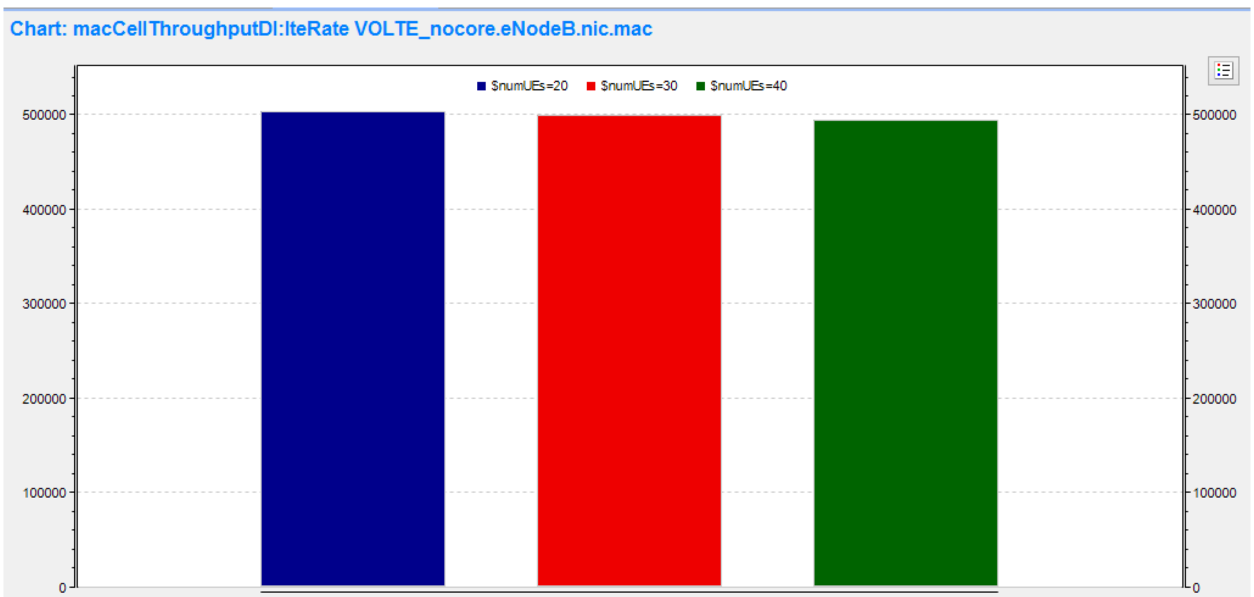


Figure 4.3-5 throughput, y-axis represent scalar throughput in bits/sec

From the graphs above, the flaws in the previous simulations can be verified. The bottleneck was the 10Mbps links, which have badly affected the performance of network with increasing number of clients up to 40.

4.4 Performance of scheduling algorithms

To analyze the network behavior and to check the performance of different scheduling algorithms, a large number of LTE UEs are considered. As explained previously, with the metro Ethernet, the 10G links are easily available, due to which, 10G links are configured in the backhaul. Only one enodeB is considered to reduce the network simulation time as the effect will be the same even if more than one enodeB is considered and the total number of clients are 200 - 100 for VOIP and 100 for VOD. The network topology and configuration details are provided below.

4.4.1 Network design

The network that is simulated is shown in the Figure 4.4-1.

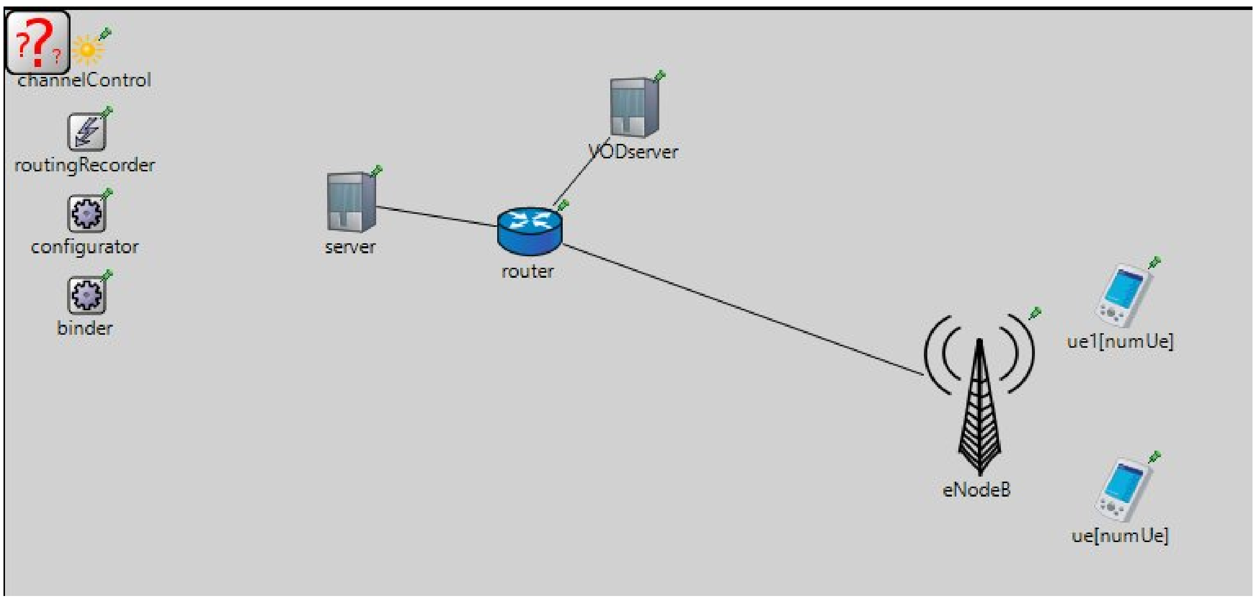


Figure 4.4-1 network design

There are two servers in the network, connected directly to the single enodeB in the network with 10G links. All the 200 UEs are under a single enodeB with the linear mobility of 1mps for VOIP and 2mps for VOD.

4.4.2 Network Parameters

Path loss = Urban Macro cell

Node placement = uniform (50m, 1000m)

EnodeB height = 25

Building height = 20

Carrier frequency= 2GHz

Bandwidth = 10 MHz

Thermal Noise = -104.5

UE noise figure = 7

EnodeB noise figure = 5

Antenna Gain of EnodeB = 18

Antenna Gain of UE = 0

Core link = 10GE

Radio resource allocation type = localized

Max HARQ retransmission = 3

Transmitted power = 40

Scheduling = MAXCI, PF, DRR

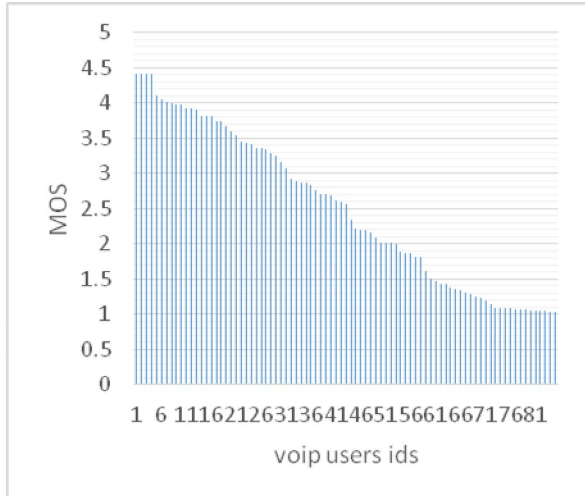
IP addressing and the channel parameters are obtained from the xml files by referring to them from the ini and NED files.

4.4.3 Results and analysis

To check the effect of different scheduling algorithms for 200 UEs per enodeB, different results have been collected. The results are collected for each individual node and all nodes are considered in statistics. Voice and VOD results are collected and evaluated on the basis of scheduling algorithms.

4.4.3.1 MOS

The first important parameter to consider for VoIP is MOS, which represents the quality of voice with 5 being the best and 1 being the worst on scale. The four figures below show the MOS for 3 different scheduling algorithms and a combined graph for all the three. OMNeT++ generates graphs individually for each node, and unlike OPNET, it does not generate average value for all nodes. It also collects results randomly for different nodes, e.g. UE1 will have 4 samples collected at some interval and UE will have 10 samples collected for the same interval. So it becomes difficult with OMNeT++ to choose a single node, e.g. UE1 for all the three different clients and observe its performance. This is because, it is quite possible for UE1 to be placed near enodeB in one simulation, while in other, it may be placed at the edge due to random placement of the clients, thus producing different results. To plot the graphs in a proper assorted manner for all the clients, the results are exported to Microsoft Excel Sheet. First 3 figures show the bar graph for MOS value per user, while the last graph shows the line made by combining scalar value of each user.



There are two important factors that can be observed from the results generated in these simulations; spectral efficiency and the cell edge user performance.

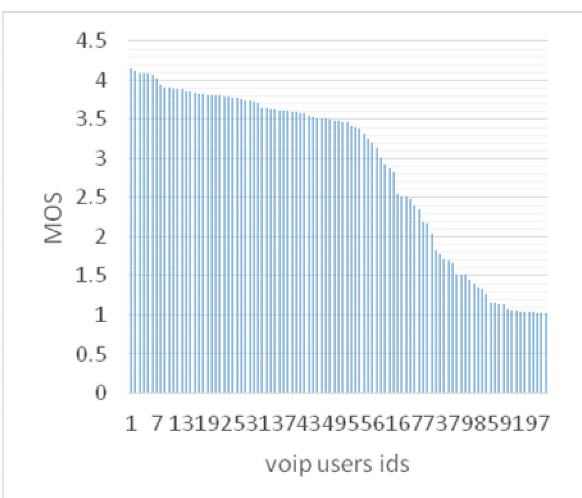


Figure 4.4-2 MOS for PF

Figure 4.4-3 MOS for MAXCI

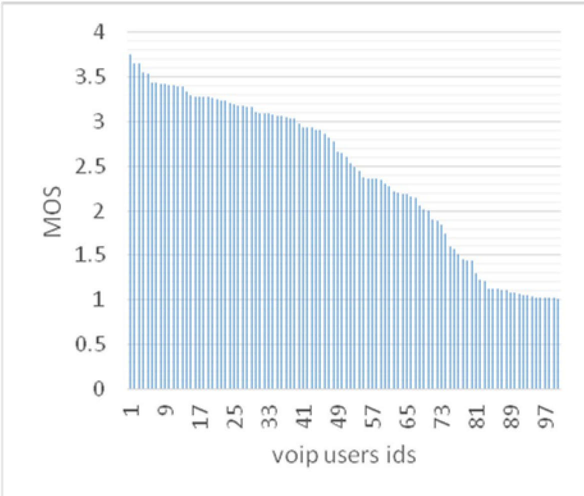
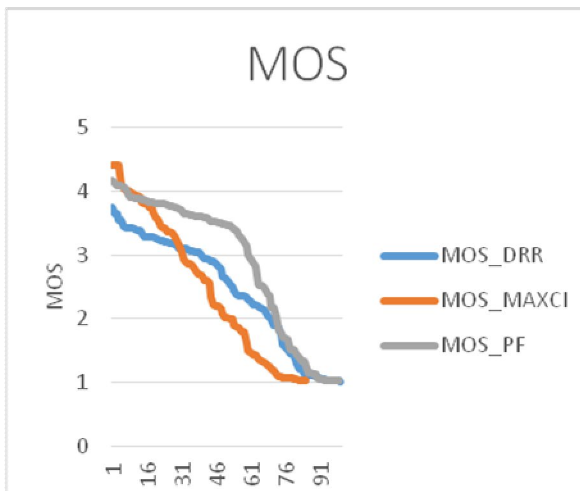


Figure 4.4-4 MOS for DRR user-ids

Figure 4.4-5 MOS for DRR MAXCI PF, x-axis represent



From the above figures collected for MOS values, it has been found that the spectral efficiency is maximum in MAXCI for nodes having good channel conditions and approaching a value of 4.5 as discussed in the theoretical portion. MOS performance degrades quickly for those nodes, which are away from the enodeB or under bad channel condition and maximum number of such nodes are missing from communication. It can be seen from the graph that 15 nodes are missing for MAXCI, and the reason is that, it provides resource allocations for nodes, which are under good channel condition.

The maximum MOS obtained for DRR is 3.6 and all nodes follow the uniform degradation in the graph. The nodes, which are far away from the enodeB are on right side with lower values. No node is missing from communication because in DRR, the resource allocation is provided per client irrespective of the channel condition, which provides fairness. The clients with good channel condition cannot utilize maximum efficiency which can be verified from the graph as well.

The PF is utilizing the intermediate behavior of DRR and MAXCI. From the graph, it can be seen that spectral efficiency is in between DRR and MAXCI. For an acceptable MOS score, about 60 nodes have value more than 3. This is a good score in contrast to MAXCI in which only 30 have a score greater than 3, and DRR in which about 43 nodes have acceptable score of greater than 3.

4.4.3.2 Frame delay and jitter

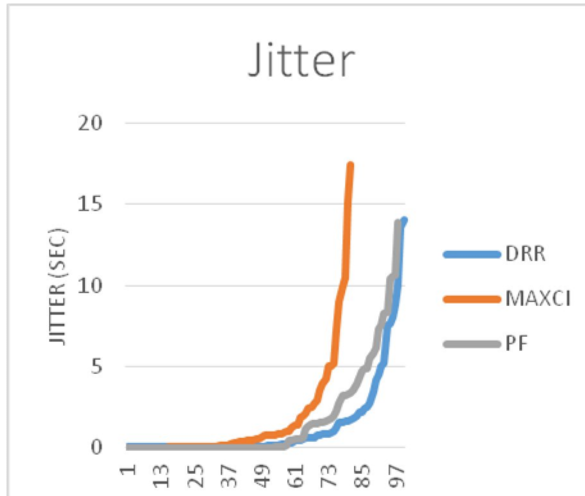


Figure 4.4-6 jitter for DRR MAXCI PF, x-axis=user-ids

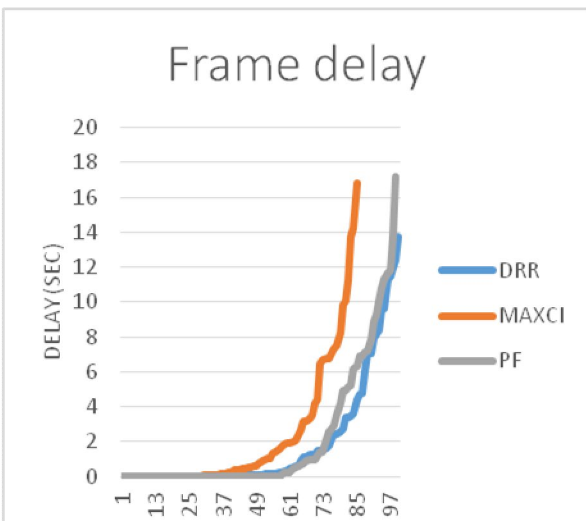


Figure 4.4-7 Frame delay DRR MAXCI and PF, x-axis=user-ids

Figure 4.4-5 and Figure 4.4-6 represent frame delay and jitter respectively. Frame delay represents delay encountered by VoIP frames and jitter is the difference in arrival of the consecutive frames at the destination. It can be seen that jitter is directly related to frame delay; greater frame delay causes greater variance in consecutive frames at the destination. According to the performance metrics, jitter should be less than 30ms for VoIP. In PF and DRR, 60 clients have jitter less than 30ms, while in MAXCI the same jitter is for 30 clients due to which, it becomes unacceptable because the clients getting fewer resources will face greater jitter. In case of PF and DRR, the jitter value becomes unacceptable for nodes at the edges due to bad channel condition and unavailability of another enodeB for handover.

4.4.3.3 VOD end to end delay

The end to end delay of VOD is shown in the Figure 4.4-8. The LTE in OMNeT++ does not provide per service QoS bearer support but it provides better quality to voice than VOD. It can be seen from the graph that in PF and DRR, only 35 nodes are able to communicate for VOD services while in MAXCI only 21 are able to avail the service. The DRR has the maximum delay from the beginning and is providing no acceptable performance for VOD as the network is already congested with VoIP users and it provides resources in a round robin manner to each client. The MAXCI is providing best services for the VOD in a good channel condition with maximum spectral efficiency but it has maximum missing nodes and very high end to end delay for nodes which are in a bad channel condition. PF is providing good performance as compared to others as it is the intermediate of both MAXCI and DRR. For the 10 nodes it is providing acceptable VOD services.

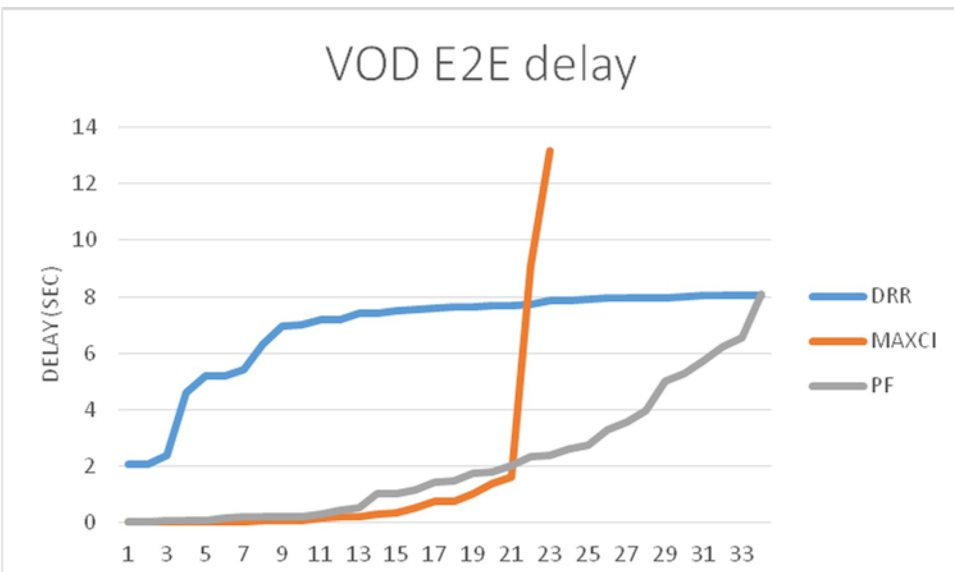
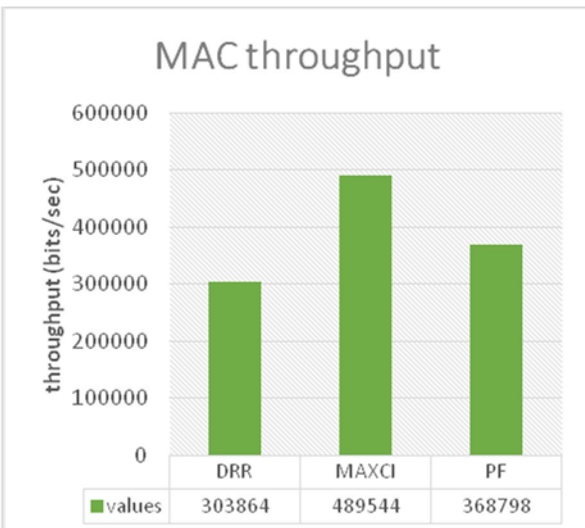


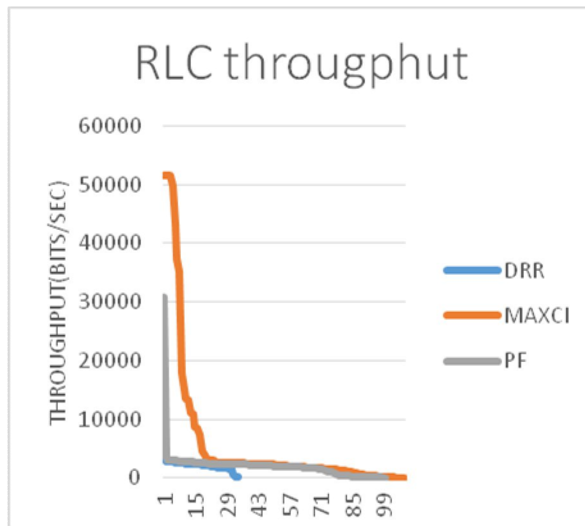
Figure 4.4-8 VOD E2E, x-axis represent VOD user-ids

4.4.3.4 Throughput

As previously discussed in theory and verified from the above results, the MAXCI uses maximum spectral efficiency; thus it has the maximum throughput. The MAC and RLC throughput is shown in the figure 4.4-8 and 4.4-9. Both the throughputs in the graphs below are showing the highest throughput for MAXCI, then PF and then DRR.

Figure 4.4-9 MAC throughput *Figure 4.4-10 RLC throughput, x-axis=VOD users-ids*





Chapter 5

Conclusion

The performance analysis of LTE network was carried out in OMNeT++, which is a free source simulator for modeling and simulation of different types of networking technologies. The LTE in OMNeT++ supports the EUTRAN model, mobility model, and channel modeling. The supported applications are voice and Video On Demand (VOD) for performance evaluation of the network. LTE model has rich support for configuration changes to model different types of real world scenarios and to evaluate the results in terms of end to-end-delay, jitter, MOS, LTE statistics.

By evaluating the results generated, it can be concluded that the core links for an LTE network should be very fast and bandwidth limitations of the core badly degrade the overall network performance. For the service provider to have a high speed LTE network, should have a highest backhaul network. For the same number of voice and video users under congested network, it has been found that voice gets more prioritization than video. The resource utilization at the LTE radio allocates more blocks to voice than to video, as voice is most sensitive application. It has also been observed from the results that most of the video users were unable to communicate and completely denied resources. While evaluating the performance of various scheduling algorithms, it was observed that MAXCI has the highest throughput and maximum spectral efficiency for nodes that are distributed away from the enodeB. It has got worst performance due to bad channel condition and maximum resources are allocated to UEs having good channel condition. In case of voice, maximum number of completely denied users are found in this scheduling method. In Distributed Round Robin (DRR), the throughput and spectral efficiency observed is lower than all the 3 methods, but the resources are allocated uniformly to all the UEs. The best performance was observed for PF scheduling algorithm as it performs the function of both MAXCI and DRR. From the results obtained, it is recommended to use PF for macro cells in rural areas, while MAXCI for micro cells in densely populated areas.

There are various limitations that have been observed in LTE model inside the OMNeT++, which can be eliminated in future releases. The web based surfing applications are not supported in UE. There is no QoS bearer support, which guarantees the end-to-end delivery of application over the LTE network. The evolved packet core network is not properly modeled, which includes charging services, authentication etc.

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