QOS PERFORMANCE ANALYSIS: DESIGN AND DEVELOPMENT OF VOICE AND VIDEO MOBILITY OVER LONG TERM EVALUATION (LTE) MODEL

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THIS THESIS IS PRESENTED AS PART OF DEGREE OF MASTER OF SCIENCE IN ELECTRICAL ENGINEERING WITH EMPHASIS ON SIGNAL PROCESSING

BLEKINGE INSTITUTE OF TECHNOLOGY
SEPTEMBER, 2013

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Abstract

The evolution of 3G systems has contributed to a significant amount of progress towards 4th generation wireless technology, Long Term Evolution (LTE). On the other hand, demand for more bandwidth has been evidenced by the ever-growing usage of real-time applications such as video conference. For instance, users tend to have reliable and efficient connection when they are on the go maintaining the minimum quality of the video conference. In order to meet these challenges, QoS of LTE makes it an ideal solution. A simplified IP-based system architecture and introduction of Orthogonal Frequency Division Multiplexing (OFDM) have made LTE possible to satisfy its promise targets. In addition, LTE capabilities are further improved with enhanced Quality of Service (QoS) support for multiple data services, such as voice and other multimedia applications. LTE packet scheduling plays an essential role as part of LTE’s Radio Resource Management (RRM) to enhance the system's data rate and to support the diverse QoS requirements of mobile services. LTE packet scheduler should intelligently allocate radio resources to mobile User Equipments (UEs) such that the LTE network adheres to its performance requirements. In our thesis work, we conduct a comprehensive performance evaluation of LTE scheduling algorithms for real-time application such as video conferencing traffic. The evaluation is carried out using the OPNET simulator. In order to analyze the performance LTE scheduling algorithm, our analysis involved with LTE Admission Control is twofold. First, 6 scenarios have been modeled in the way that 3 of 6 scenarios deals with no LTE admission control techniques applied in the proposed network models while other 3 scenarios deals with LTE scheduling techniques applied. Secondly, video conferencing sessions are configured between two LTE cells with same number of UEs in which all UEs under each cell in the entire proposed network modeling scenarios. In order to make our evaluation more realistic we have applied various network loads so that we can observe how LTE scheduling techniques work its best in the case of highly loaded network. Our simulation results show that video conferencing node with the highest priority maintains tolerable delay and loss while nodes without scheduling techniques experience worst performance.
Acknowledgement

First of all, we wish to express our deep gratitude to our thesis supervisor Maria Erman, who gave us the ability and courage a lot to complete this thesis work. It is a matter of great privilege to conduct this thesis under her supervision. She has always tried to inspire and guide us in the right direction all the way.

We also would like to confess our appreciation to those who have reviewed our proposal and shared their personal experience and views with us. It should be mentioned that, we are bound to all members of the Blekinge Institute of Technology for their valuable support.

Finally, we thank Almighty Allah. Without their blessing this trip would not have been possible for us.

Md. Sadat Hossain Chowdhury
Shahreer Mahmud
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Chapter 1

Introduction

1.1 Theory overview

The nature of living and working of a human being has completely changed by the mobile broadband technology. The communication among us has improved a lot and still in process of enrichment with greatest speed of broadband and the latest services offered by the broadband providers which are very excited from both on road and at home. For this is to happen LTE[1] that is long term evolution is a key player in the real world. It is a 3.9G which is called 3.9th generation standard of radio access, but where as in its marketing activities mobile carriers are advertising this as a 4G[2] which is 4th generation. This LTE model is designed and introduced with a concept of getting revolution in technology in mobile broadband with some important considerations of rates for data with advanced, effectiveness in power to be enhanced, best in QoS[3] and latency will be low. This is a technology which called access technology and permits communication among the BS (base station) and UE (user equipment). Moreover, it is a network of type fully switched packet and therefore this maintains the data transmission among compliant user equipment’s.

Before introduction of LTE, solutions for broadband services with GSM (global system for mobile communications)[1] and UMTS (universal mobile telecommunications system) [2] given at speed of 20 Mbps for their services like voice and message of short data transfer with files and video calling.

Even though this is a better achievement in broadband era when compared to older technologies, the applications which are developing in current era needs more optimistic in those services and data transmission rate as fast. If any new technology is discovered and proposed it should be adjust with the mobility with lenient in terms of its methods and solutions and it should be in a way that it can operate on any channel conditions irrespective of its speed in changing conditions. The important aspects that should maintain in performance in customer point of view is expecting better QoS, minimizing in errors while transmission, exploring more data transfer rates and with high mobility for solutions to new mobiles. That is the reason why the technology
network of LTE is advent with techniques of radio access is optimized with greatest techniques compared to its older techniques [1].

The E-UTRAN[10, 17] that is called as evolved universal terrestrial radio access network is the new RAN (radio access network)[3, 10] which is developed to latest generation mobiles and its solutions. This E-UTRAN is composed with E-UTRAN Node B[24] which is called as eNodeB. This node will carry out all the services and functionalities involved in radio resource management. LTE belongs to the 3GPP which is 3rd generation partnership project and branded with an air interface which is of highly performing for the technology for mobiles.

The 3GPP[2] group are documented the specifications, standards and releases for LTE. The specifications of LTE provide at least 100 Mbps peak rate for downlink and 50Mbps for an uplink, less than 10 ms times of RAN round trip. Carrier bandwidths which are scalable are supported by the LTE and the range is from 1.4 MHz to 20MHz and accepts both TDD and FDD that is both time and frequency division duplexes. A further step towards 4G of radio technologies specially designed for improving the speed and capacity of mobile telephone networks. The mobile operators like Nokia, Ericsson and TeliaSonera included LTE services in their network. There are eight major focuses which are released by the most of 3GPP for adopting 4th generation technology for mobile communications and includes an architecture of all-IP flat networking.

1.2 Research Aim and Objectives

This thesis focuses and analyzes the QoS (quality of service) effects based on how the radio resources which are available should be allocated to various users for the particular principle in relation to the users precedence basis over LTE. Here schedulers use predefined algorithms based on origin to differentiate the flow of traffic. The additional aims for this research involves in providing proper guidelines and methodologies, which can be addressed in similar type of research in future.

The research objective for this research is to find out the performance of delivering voice and video (video conferencing) traffic over a GBR (guaranteed bit rate) and Non-GBR (non – guaranteed bit rate) carriers with the priority for users. This study will also presents a study of
performance simulation of LTE with focus on both the scenarios of UL (uplink) and DL (downlink) for video conferencing with web traffic.

Following are the stated rules and the major research objectives are reviewed as follows:

- Both qualitative and quantitative methods can be applied for this research to guide this research in appropriate direction
- This study performs replication in OPNET and it can help to be familiar with components from various networks of OPNET software
- Developing, testing and assessing the driven replication in OPNET scenario
- Make use of the strategies of scheduling through the completion of the network model which is proposed in a LTE replication environment
- Here researcher discuss about the various constraints that affects the metrics of performance like packet loss, end-to-end delay and packet delay variation of video conferencing (voice and video) in LTE network and significantly scrutinize different approaches that are recommended in the literature
- Simulation with different loads in the network like low, medium and high with different network scenarios and analysis of the results from the simulation
- Conclusions will be drawn from these outcomes by presenting and interpreting

1.3 Research questions:

1. What is the impact on the packet end-to-end delay for video conferencing when NGBR and GBR bearer are established with mobility in LTE network under varied network load?

2. What is the impact on the packet loss performance for video conferencing when NGBR and GBR bearer are established with mobility in LTE network under varied network load?

3. What is the impact on the packet delay variation performance for video conferencing when NGBR and GBR bearer are established with mobility in LTE network under varied network load?
1.4 Research Methodology

There are two types of research methods for research methodology, those are primary research methodology and secondary research methodology and also called qualitative and quantitative research methods[5].

1.5 Quantitative Research Method

According to the Holland and Brook in 2007, [6] the theoretical approaches are called as quantitative research method. These methods are extensively used in experimental based studies. This quantitative research method includes the opinions and views of different authors and scholars on a selected title through which the essential research data can be gathered and analyzed. Different types of secondary data sources are involved in the quantitative research method such as scholarly articles, case studies, web documents, journals and reviews that are peer reviewed by authors.

1.6 Qualitative Research Method

According to the Collis and Hussey in 2009, human behaviors and actions play a vital role within the qualitative research methodology in which the entire aspects are related to human actions. When compared with the quantitative method, the data sources of the qualitative method are reliable and accurate because it always includes the opinions and views of research participants, which is spontaneous data that does not include any author’s or scholar’s opinions. In qualitative research, there are different types of primary data sources such as research surveys, research interviews, focus groups and semi-structured interviews. In this qualitative research method, the questionnaires play a vital role because the research questions of the proposed study can be fixed and analyzed by using research questionnaires.

1.7 Mixed Research Method

According to the Gibbs in 2002, the mixed research method is the combination of quantitative and qualitative research methods. In the mixed method, both primary data sources like research surveys and research interviews and secondary data sources like published articles, peer
reviewed journals and web documents can be used. The Mixed method is used when there is a necessity of finding huge data quantitatively and qualitatively that needs the involvement of the author’s opinions as well as human opinions. The data sources can be used as the research techniques within this mixed research method as follows:

1.8 Research Techniques

There are two different types of research techniques such as primary research techniques and secondary research techniques that involve many sub research techniques:

Primary research techniques:

- Research interviews
- Research surveys
- Research questionnaires
- Focus groups

Secondary research techniques:

- Scholarly articles
- Individual and company case studies
- Peer reviewed journals
- Expert reviewed web documents

1.9 Methodology

To assess video quality in terms of end-to-end delay, packet loss and packet delay variation in an LTE environment, the key method to analyze data by computer simulation. As most practitioners and engineers advocate, this method has been widely used as an effective method to adjust, troubleshoot and optimize network infrastructure[4]. With the wide range of simulation programs, flexibility is greatly affected in terms of model development while minimizing the hardware cost.
In the beginning, the most important factors affecting the video quality in terms of packet delay variation, end-to-end delay and packet loss in an LTE environment, will be identified by taking the help of existing research and knowledge. After this, a detailed review of existing literature related to the current field of research is to be conducted. The required data will then be collected for the assessment. After the literature study, experiments and simulations will be undertaken to produce statistical data that will be analyzed.

For the current study, the LTE network models were developed in the OPNET simulator, using a variety of networking devices where several attempts were used with different network loads to investigate the video quality with packet delay variation, end-to-end delay and packet loss. Quantitative data such as packet loss and end-to-end delay is collected for analysis, which validates our experiment in this context. The simulation results are subsequently discussed in various statistical plots and tables.

1.10 Thesis contribution

The performance analysis of QoS measurements is one of the main focuses in our thesis paper where we focus on packet delay variation, packet loss and end to end delay for voice and video conferencing in the LTE network under a congested network. We also focus on the issue of LTE protocol responses where we measure how well the LTE protocol responds under different network scenarios. We also validate our network models and analyses our results using the OPNET modeler 17.1 while video conferencing is happening with real time applications in three different network scenarios.

1.11 Related research work

Various publications and contributions discuss the QoS evaluation of the LTE. The authors of [6] have described the LTE air interface with (the link from the base station to the user equipment) video and audio capability. They discuss how the system external quality of video and audio can affect air interface both video and audio capability. They also provide different explanation of quality with respect to various statements. In [7] the authors discuss the effects of different QoS type. It is connected with the scheduling strategies and providing services over LTE varied result. Authors of [8] have presented a new scheduling algorithm to deliver wireless RT video in the case of downlink LTE and they have achieved the best video quality. The authors in [9]
(equalization of video rates transmission types) have developed a method called semi-optimal video smoothing strategy. In LTE environment considering Quality of Service they managed to transfer MPEG-4 and H.264 with high data rate.

1.12 Thesis Outline

In this thesis, the background of the LTE technology, the basic coverage of the area described. Study of LTE network architecture of this study is given in Chapter 3. In Chapter 4, the simulation architecture and implementation is described whereas, simulation results and analyzes are described in Chapter 5. Chapter 6 concludes the whole thesis.
Chapter 2

LTE Background

This section will include the complete background about LTE and in this section we present an overview of the 3GPP LTE and LTE goals. The most significant information about the OFDMA and SC-FDMA will be available for reference.

2.1 Overview of 3GPP LTE

The LTE is the evolution drawn from the RAN which is Radio Access Network [3]. LTE is also known as the E-UTRAN which is called evolved universal terrestrial radio access network [10, 17]. The targets of 3GPP LTE are to provide more efficiency and the enhancement in data rates[3]. This also supports enhancement in signal range with greatest user response time[2]. LTE assists in interoperability and this will come with circuit switch network with inheritance contrasted to systems in present era. Long term evolution maintains a variety of bandwidths like 1.4, 3.0, 5, 10, 15 and 20 MHz (TECHNICAL WHITE PAPER, 2007)[10]. Long term evolution uses OFDMA [10](Orthogonal Frequency Multiple Access) for DL (downlink) and UL (uplink) SC-FDMA [11](FDMA with single carrier) (J. Zyren and D. W. McCoy, 2007). The DL peak rates for LTE is over 150 Mbps and the round trip time for radio access network is less than 30ms which is 3 times more compared to HSPA called as high speed packet access in spectral efficiency and is depicted in 3GPP release 6[12] (UTRA-UTRAN, 2006). The figure 2.1 shows the overview of 3GPP LTE.

![Figure 2.1: overview of 3GPP LTE](imageURL)
2.1.1 The evolution in LTE (LTE-Advanced)

The concept of invention behind the LTE is to provide the broadband services to mobile platform with service quality and speed in the service. The LTE is introduced with most efficacies in its service and this technology delivers the services with quality and speed. With the air interface introduced in latest and the innovation in technology used is OFDMA or SC-FDMA[11] was the cause for improved service compared to network of HSPA with offerings and art combination state. This LTE uses the greater suppleness for having the spectrum such as hold up of TD-LTE and 20 MHz[10] bands to use the unpaired spectrum and it gives a toolbox for holding up the future implementations such as Higher Order Modulation and Multiple Input Multiple Output(MIMO)[11]. To say in reality, for assisting a flawless growth to HSPA+ this toolbox is useful for the HSPA[16].

Maintaining the speed and quality in broadband services for mobile platform in service delivery for prospected customers for development in efficacy for better performance overall, carriers of mobile networks should observe for exponential growth like customers on mobile broadband traffic continuously. The future developments done on LTE are called as LTEA and the major aspect involved in LTEA as conceptual is to deal with the demands discussed above[16].

The evolution in LTE is the basis for LTEA and in LTEA deployment of air interface should occur continuously. These deployments will provide the complete efficacy for the technologies which are exist and the smooth evolution towards LTE –A is in its ecosystem. LTE-A allows more than 20 MHz bandwidths for carrier aggregation and this enables to provide supplementary service like MIMO as toolbox for innovative features like relaying. LTE-A allows the previous releases by LTE as it works as backward direction and inherits the required tools. It is not available in another technologies besides LTE-A. We are very thankful for these features with advanced technology and a true technology of 4G is transitioned by LTE according to its needs of ITU for IMT advanced.

2.1.2 Status of LTE-A (as of January 2011)

The standards of LTE are constant and they are in compatibility from the 2008 march, where it is the time of ample networks that are giving the proof for commercial networks that shows the starting trials. The truth behind this is to deliver the services of broadband services for mobiles
using LTE with the user experience in the real world and the actual deployment. The LTE got succeeded in commercial way instead and growing in an adoption rate and it has over limited on every other technology which network used. At the end of the year 2010, over 120 operators are committed for roll out for the network of LTE. The releases of 8 and 9 from 3GPP specifies about LTE.

The key drivers for the improvement in technical aspects and LTE-A schedule growth got advanced as IMT. The processor for IMT Advanced is started by the ITU for describing the needs for next generation of RIT abbreviated as radio interface technologies which are released in a letter in 2008 early periods. The 3GPP has to achieve the goals of advanced requirements of IMT to meet and for its progress in standardization got well in 3GPP. The initial specifications of LTE Advanced are released in earlier period of 2011 and the progresses are accomplished by 3GPP and external bodies given revelation for LTE-A to meet the IMT requirements which are advanced. The initial LTE-A 3GPP standards services and products which are complained are entered in 2012.

2.2 The LTE-A toolbox

As indicated in releases of 8 and 9 of 3GPP the LTE has been optimized according to the broad vicinity operation as predictable, on stations based on macro basis and for the receiver of dual type and transmit used is single antenna including terminals use the single band. While talking about this use case which is basic particularly, the LTE-A does not offer any important performances with enhancements because of no latest technologies are depicted in this to make it feasible. As a substitute for this, the main focus has been moved for improving the latest technologies and features for enlarging the capacities of LTE and assisting the latest trends in deploying and executing services to ensure the optimal distribution.

The technologies which are newly depicted in LTE-A will provide advantages for the CSP [13](communication service providers) community for enabling the enhancements in data rates with peak in terms of its performance, efficiency of spectrum with average, performance of cell edge, coverage, latest trends in reduction of cost in process for arranging and the networks for operating with base stations of small in size, and eliminating the connections for fixed transport.
The latest technologies introduced in LTE-A will contain the improvements in UL and DL multi antenna which we called as MIMO in general. This will co-ordinate with CoMP that is multi cell transmission as well as reception. This also has extensions for bandwidth with CA (carrier aggregation), RN (relay nodes) and Hetnet (heterogeneous network) deployments.

Figure 2.2: Depicts that the LTE macro network and enabling the efficient use of small cells both are supported by LTE-Advanced (LTE-A) [16]

### 2.3 LTE Goals

The goals or objectives for the LTE are as follows[11]:

- Downlink rates of 100 Mbps using 64QAM SISO on 2x20 MHz. 326.4 Mbps using 4x4 MIMO
- Uplink rates of 50-86.4 Mbps using SISO
- Scalable channel bandwidths of 1.4, 3.0, 5, 10, 20 MHz X2
- Multiple frequency bands: 700 MHz, 900 MHz, 1800…
- Sub 5ms latency for IP packets
- 0-15 km/hr optimized, 15 to 120 km/hr high performance, 120 to 350 km/hr supported
- Compatibility with other cellular standards (using multimode devices)

The key goals of LTE are to reduce the complexities involved in system equipment and UE (user equipment). To permit deployment of flexible spectrum which is existing or can be frequency
spectrum which introduced newly and for enabling coexistence with the remaining 3GPP RATs that is radio access technologies[13][14].

The whole objective of LTE is for giving the excessive performance as RAT and so that provide complete solution for enhancement in speed in mobility and it can be prepared for co-existing with HSPA also in the previous networks. The operators can able to change their customers from one network to another because of the scalable bandwidth from HSPA to LTE over time.

LTE is expresses as the main concert is to be a system with latency as low as given away in table 2.1. File transport protocol (FTP), online gaming, streaming of videos, Voice over IP, Push to view, videos of real time and push to talk is forecasted to maintain various kinds of services with browsing through web in E-UTRA. For the both of operations like transmission of data and reception using UE are forecasted to be as of 20MHz. The service provider can have the chance to modify the existing spectrum amount. The spectrum amount which is limited can have the additional capability for lowering the sincere pricing and development (TECHNICAL WHITE PAPER, 2007).

Table 2.1 Requirements for performance of LTE[10].

<table>
<thead>
<tr>
<th>Measured</th>
<th>Requirements</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet data rates</td>
<td>Downlink: 100 Mbps, Uplink: 50 Mbps For 20MHz spectrum</td>
</tr>
<tr>
<td>Mobility Supports</td>
<td>Up to 500 km/h but performed for low speeds from 0 to 15 km/h</td>
</tr>
<tr>
<td>Control plane latency (Transition time to active state)</td>
<td>Less than 100ms both in idle and active.</td>
</tr>
<tr>
<td>User plane latency</td>
<td>Less than 5ms</td>
</tr>
<tr>
<td>Control plane capability</td>
<td>More than 200 users per cell for 5MHz spectrum.</td>
</tr>
<tr>
<td>Cell size (Coverage)</td>
<td>5-100km with minor degradation following 30km.</td>
</tr>
<tr>
<td>Range flexibility</td>
<td>1.4, 3, 5, 10, 15 and 20MHz.</td>
</tr>
</tbody>
</table>
2.4 OFDMA for DL

Orthogonal frequency division multiplexing (OFDM) [32] is a technique of transmission with multi carrier which is accomplished of assisting services with high speed at the same time as still being proficient bandwidth. This can be achieved by forcing the several subcarriers mutually in that way for lessening the required bandwidth in association with methods of most traditional frequency division multiplexing (FDM). Though, for ensuring these adjoining subcarriers it do not be a reason for the extreme interface, and these are must be in orthogonal or it can be 90 degrees to one with the another.

2.4.1 FDM

However, OFDM is based on the FDM and in this it will allow the multiple frequencies with concurrent transmission with multiple signals in a parallel way. The figure 2.3 will deeply illustrate about the system of FDM and with 3 sub carriers, each of these are separated by the guard band for reducing the obstruction. At the receiver end we use filters for separating the individual sub carriers that are demodulated consecutively.

This research included about FDM for enabling effective demodulation layer of dual type beam to form LTE system and it is based on reference signal for demodulation DRS pattern for sub frame with CP (cyclic prefix). DRS pattern and location will be analyzed in detail in view of multiplexing method. The link level simulation results showed that the pattern designed with this method provided the best performance of throughput in most of occasions among the candidate patterns.

![Figure 2.3: View of frequency division multiplexing](image_url)
2.4.2 FDM Vs OFDM

The limitation we can observe in the FDM is that FDM needs to have a guard band among the carriers in it. Whereas in OFDM it uses the same conceptualization which FDM uses but the major difference occur between these two at the enhancement in spectrum efficiency and it will be facilitated by using the spacing among the sub carriers to be importantly lessened till they are in result with overlapping. It is happening for OFDM because of its radio signals are in orthogonal where as 90 degrees in mathematical expressions.

This can be illustrated very clear in the figure 2.4. Here the subcarrier C peak power frequency which is shown as blue is also the sub carrier C-1 frequency shown in red and C+1 shown in green and will pass through zero. This will be reference case for primary harmonics. The signals in the figure which are radiated by adjacent carriers that are C-1 and C+1 will be the reason for low interference for the C that is sub carrier. Guard bands are also used in the OFDM but here these can be appear at top and bottom ends in the channel allocation for lowering the interference among the adjacent channels.

![Figure 2.4: The variations of modulation in FDM and OFDM][20]
The real OFDM system uses the fast Fourier transform (FFT) which is a mathematical function for getting result as the parallel data streams. The FFT system and its operational descriptions is shown in the figure 2.5, figure 2.6.

The process of FFT is as follows

Figure 2.5: FFT system and its operational description[25]

Figure 2.6: the process of FFT[25]
2.5 SC-FDMA for UL

The requirements of LTE UL (uplink) will vary with the LTE DL (downlink) requirements in some ways. The key considerations for the terminals of UE are unsurprisingly power consumption. The major concerns that we have to consider here are the association between the OFDM signaling and the related loss in high PAPR (peak to average power ratio) in its efficiency. Consequently, the option for the OFDM was required using the LTE in uplink.

Single carrier FDMA that is SC-FDMA[31] is very much appropriate for the requirements of LTE UL (uplink). If we clearly observe the basic architecture of SC-FDMA for the transmitter and receiver, it is much identical to the OFDMA[32] and it provides equivalent level of protection in multipath. More significantly the PAPR is considering being lower because of the vital single carrier waveform which is fundamental.

2.6 The process of SC-FDMA

The figure 2.7 shows illustrates the process of converting bits into multiplexing in physical layer.

![Diagram of the process of converting bits into multiplexing in physical layer](image)

Figure 2.7: Process of converting bits into multiplexing in physical layer[33]
2.7 LTE Release and features

In this research and particularly in this section we discuss the various LTE (long term evolution) releases and their significant features which are described as follows.

Table 2.2 shows the 3GPP group publication about Technical specifications (Agilent Technologies, 2011)[16]

<table>
<thead>
<tr>
<th>Release</th>
<th>Specification</th>
<th>Date</th>
<th>Downlink Data Rate</th>
<th>Uplink Data Rate</th>
<th>Round Trip Time</th>
</tr>
</thead>
<tbody>
<tr>
<td>Release 99</td>
<td>WCDMA</td>
<td>March, 2000</td>
<td>384 kbps</td>
<td>128 kbps</td>
<td>150 ms</td>
</tr>
<tr>
<td>Release 4</td>
<td>TD-SCDMA</td>
<td>March, 2001</td>
<td>384 kbps</td>
<td>128 kbps</td>
<td>150 ms</td>
</tr>
<tr>
<td>Release 5</td>
<td>HSDPA</td>
<td>March to June, 2002</td>
<td>14 Mbps</td>
<td>5.7 Mbps</td>
<td>&lt;100ms</td>
</tr>
<tr>
<td>Release 6</td>
<td>HSUPA</td>
<td>December, 2004 to March, 2005</td>
<td>14 Mbps</td>
<td>5.7 Mbps</td>
<td>&lt;100ms</td>
</tr>
<tr>
<td>Release 7</td>
<td>HSPA</td>
<td>December, 2007</td>
<td>28 Mbps</td>
<td>11 Mbps</td>
<td>&lt; 50 ms</td>
</tr>
<tr>
<td>Release 8</td>
<td>LTE</td>
<td>December, 2008</td>
<td>100 Mbps</td>
<td>50 Mbps</td>
<td>10 ms</td>
</tr>
<tr>
<td>Release 10</td>
<td>LTE-Advanced</td>
<td>Published 2012</td>
<td>1 Gbps in a low mobility</td>
<td>375 Mbps</td>
<td>5ms</td>
</tr>
</tbody>
</table>

Table 2.2: The 3GPP group publication about Technical specifications

2.8 Quality of service

The major challenges facing in the service based on the current IP is the providing the Qos which is quality of service. It is mandatory that to maintain the quality of service (QoS) because of the rising number of multimedia applications across the internet. For instance, voice services and video services can have the wide-ranging bandwidth and these require maintaining the QoS as less delay and one more point here is that it is not probable for all requirements to maintaining the quality always.
To sustain the demand of quality of service (QoS) [3] there are some models, different mechanisms, policies and schemes recommended by internet engineering task force (IETF). In this research, researcher used algorithm for scheduling and this will prioritize based on the requirements of users. For this situation, the prioritization for the IP packets will be based on importance.

This provides many bandwidths to more important traffic while delays those, which have less priority. Moreover, in some cases these less important traffics are ignored. There is a concept of dividing packets using the priorities as well as associating quality of service identifier which is called QCI by the TFP acronym as traffic forwarding policy is done by the trick of introduction from the QCI. The description for TFP as a set of parameters to the forwarding of traffic will flow through all nodes in network from source user to destination user. There will be various TFP’s allocated for various traffics which are defined by QCI and it can give the guaranteed bandwidth that is required.

In majority situations user will classify the QoS by the priority settings. For getting the services on demand for the users, the end-user will say that how the classification of packets should be done as well as related with TFP (R. Ludwig, H. Ekstrom, 2006).
The figure 2.8 showing the QoS in PSN (packet switched network)

Figure 2.8: Quality of service in packet switched network[17].
The diagram 2.9 shows the QoS in session network

Figure 2.9: Quality of service in Session network
3.1 LTE Architecture

Unlike the previous cellular systems, which are circuit-switched model, the intent of creating this LTE mode is to support the packet-switched services. The gamut of having such a system built is to have an seamless provision of Internet Protocol for the connectivity of User Equipment (UE) and the PDN (packet data network), however the objective of all this designing is to ensure that there is no kind of disruption to the end user’s in mobility for their applications.

The Evolved Packet System (EPS) which is combination of LTE and SAE is the basic attribute of this architecture. However, Universal Mobile Telecommunication System (UMTS)[22] is encompassed to LTE, by radio access thru the Evolved UTRAN (E-UTRAN), it is also integrated by evolution of non-radio aspects which are technically specified as “System Architecture Evolution” (SAE)[22]. This also includes the EPC network which specifies the support systems for the mobility of the application.

Figure 3.1: The EPS Network Elements.[17]
EPS bearers are the concept which is predominantly in place for routing the traffic to UE[22] from the PDN in the gateway. Whenever there is any data transfer between the Gateway and UE, a bearer with a IP packet flow under the specific QoS is used. As per the requirement analyzed by the system, the bearers are released together by E-UTRAN and EPC. This chapter elucidates the comprehensive insight of the overall LTE, EPS network architecture, by reviewing the concepts and functionalities of Core Network (CN) and E-UTRAN.

The protocol stack which is vividly established among various interfaces and the varied protocol layers which are part of the scenario is considered for the review to have a better understanding. The bearer paths which are end-end, in specific with the QoS have also been reviewed, including the procedural methods for establishing a bearer. This chapter focuses on the network interfaces in specific along with the focus on E-UTRAN and the other procedures which are used to support user mobility.

The Services such as VoIP and the basic attributes like PDN for the usage of Internet, is provided with IP connectivity by EPS. EPS bearer is integrated with defined QoS. Multiple bearers shall be put to work for ensuring varied streams of QoS or the connectivity levels to varied PDNs. To
illustrate the scenario, we can presume that the user might be using a web browser for downloading some stuff, and also might end up in a VoIP simultaneously. The bearer which is handling VoIP will ensure QoS for the voice calls, whereas the session of the browser will be handled by a suitable bearer with requisite standards.

3.2 The core Network

The combination of EPC and SAE together is called the Core Network, which is in over control of establishing the bearers and the control of UE. The logical nodes which are key to the EPC network are:

P-GW (PDN Gateway)

S-GW (Serving Gateway)

MME (Mobility Management Entity)

In categorization to these nodes, there are few other nodes and functions of EPC, like HSS (Home Subscriber Server) and the PCRF (policy control and charging rules function). As the fact of matter, only for a certain QoS, the path bearers are provided by EPS, IP multimedia systems (IMS), works on the provision of bearers for the VoIP applications, which are in trend considered as the outside of EPS.

The Logical Nodes are in detail discussed below.

PCRF- This function is accountable for the policy controls and also for the control of the flow-based functions pertaining to the PCEF (Policy Control Enforcement Function). This PCEF is an integral of the PDN gateway. The key functionality of this PCRF is to decide upon the data flow treated in the PCEF and this is in specifications to the user’s subscription activity.

HSS- This contains the SAE data of the user’s, the data comprising of subscribed EPS – QoS profile and the restrictions towards access upon roaming. The relevant information pertaining to the PDN of the user is also available for the user to connect; however that data is maintained in the form of APN (Access Point Name) or a PDN Address denoting the IP address. In addition to
this, this protocol holds the dynamic functions like Authentication Center Integration (AUC), which is primarily accountable for authentication and generation of security keys.

P-GW – The gateway of PDN does the act of allocating the IP address for UE, and also ensure the enforcement of QoS. It also takes care of the flow-based charges according to the specifications from the PCRF. P-GW is also responsible for the TFTs (Traffic Flow Templates), and the enforcement guarantees the bit rate of the data flow by GBR (Bit rate bearer).

The other functionality which it acts is as mobility anchor to interworking when working on non-3GPP technologies as CDMA and networks as WiMAX.

S-GW – The user IP packets are all transferred with the usage of Serving Gateway, as it works as the local mobility anchor, for the UE move amidst of eNodeBs as the data bearers. Retaining of information pertaining to the bearers also takes place in this node, when the UE is in idle an state which is terminally mentioned as ECM-IDLE. It holds the information also about temporarily buffered downlink data, whilst MME is working on paging the UE to bearers for reestablishment. Few of the administrative functions like the records of information collection for charging and the lawful interception[23].

S-GW also handles the function of the being a mobility anchor with the other 3GPP technologies interworking, and for GPRS and UMTS

MME – This node is the control node which is responsible for processing the signals between the CN and UE. The protocols running between them are considered as the NAS (Non Access Stratum) protocols.

Security[23] functionalities are the primary account of the MME for both user data and the signaling. When any network gets tangled with UE, there is authentication which is performed mutually between MME and the UE. This authentication is established with security keys which are used for the encryption of the bearers of such data.
3.3 The access network

The access network related to E-UTRAN of LTE is comprised of the eNodeBs network. For the regular user traffic there is controller which is centralized with E-UTRAN, and this makes it to be a flat network.

In a usual scenario, the eNodeBs are interconnected to each other with an interface called[24] “X2” and also it gets connected to the EPC thru S1 Interface categorically. They get connected to the MME with S1-MME interface, whilst S-GW is connected by S1-U interface. The protocols which are running between the eNodeBs and the UE are technically termed as “A S Protocols”

Figure 3.3: Overall E-UTRAN Architecture[17]
The E-UTRAN is primarily accountable for functions which radio-related, and the ones which are to considered as functions are

RRM (Radio Resource Management) – The functions of radio bearers such as controlling of radio bearer, admission control of radio signals, scheduling and also dynamic allocation of the resources to UE’s in the frequencies of uplink and downlinks too.

Header Compression – This function supports in effective usage of radio interface by ensuring to compress the IP packet headers, which in other way might turn out to be a significant overhead for small packets.

Security – The functionality of ensuring that all the data sent over the radio interface is also encrypted

EPC Connectivity – Signaling towards MME and the S-GW bearer path is also the key functionality.

When we dwell in to the network side, many of these functions stay put in the eNodeBs, and they are responsible independently for handling multiple cells. In the earlier generation technologies, there use to be a very complex process, whereas here LTE integrates the controller functions of radio in to eNodeB. This facilitates a uninterrupted interaction between varied protocol layers or the RAN (Radio Access Network), thus avoiding any kind of latency and raising the efficiency.

Whenever such distributed control is in place to truncate the need for high-need, processing of intensive controllers, is bound to cut down the costs and also avoid the failure rates.

In furtherance to all it, LTE doesn’t handle any kind of soft handover[24], and there is no need for any kind of centralized data mining functions for the network. One of the repercussions of lack of holding a centralized controller node is that the when the UE moves across, the network is bound to transfer all the information pertinent to UE, from one eNodeB to the other respective node transfers. Hence forth, there is a need to administer lot of mechanisms to avoid any kind of loss in the data during the transition.
3.4 Roaming Architecture

PLMN (Public Land Mobile Network) is a network which is run by a telecom operator in any country. The mode of allowing the subscribers of one service provider to access the network of other PLMN’s thru the roaming facilitation, is a very effective tool for mobile networks. This facility stands no exception to LTE networks too. Whereas the implications are the LTE/SAE factors are allows P-GW of home network or the visited network of current location, and facilitating the usage of home network even on the roaming network[19]. The significance is that when the P-GW is in the visited network, a indicator of “Local breakout” is specified to the internet of the visited network.

Figure 3.4: Roaming Architecture for 3GPP Access with P-GW in Home Network.[18]
3.5 Other Networks and Interworking

The support of Interworking and the usage of mobility with the corresponding networks, like RAT’s (Radio Access Technologies), which is supposed to be the GSM, WiMAX, CDMA2000, UMTS. The architecture of networking the internetworks with varied mobile networking streams.

The S-GW, is the protocol [27] node which acts as the interface for the anchoring the mobility of interworking with the other technologies pertaining to 3GPP, which is as UMTS and GSM. However when the P-GW acts as a anchoring unit, allowing seamless mobility towards 3GPP networks. The P-GW also supports PMIP (Proxy Mobile Internet Protocol) kind of interface.

![Figure 3.5: Architecture for 3G UMTS][38]

3.6 Protocol Architecture

The radio protocol architecture of E-UTRAN insight is provided with functionalities[25].

3.6.1 User Plane:

The entangling of IP Packet for UE is congregated in a EPC specific protocol and the P-GW and the eNodeB for the purpose of transmission to the UE. Varied tunneling protocols are put to use
in different interfaces. A 3GPP specific kind of tunneling protocol technically termed as GTP (GPRS Tunneling Protocol) over the CN interfaces and interfaces like S1 and S5/S8.1

The E-UTRAN kind of user plane, protocol stack which will consist of the PDCP (Packet Data Convergence Protocol), MAC (Medium Access Control) and RLC (Radio Link Control) sub layers which gets terminated in the eNodeB of the other sides of network.

![Figure 3.6: The E-UTRAN user plane protocol stack[20].](image)

### 3.6.2 Handover of Data Handling

When there is an absence of any controller node which is centralized, handover of data buffering, due to the mobility of user in the E-UTRAN which is to be performed in the eNodeB itself. The protection of data whilst of handover[25] is the accountability of PDCP layer. The MAC and RLC [28] layers approaches start a new session in a new cell once the handover[25].
3.6.3 Control plane

The stack which indicates the AS protocols in the blue region of the figure 3.7 is the protocol stack which is for the plane control between MME and UE. The functions of the lower layers, that there is no header compression function of the control plane.

The RRC[26] (Radio Resource Control) protocol is considered as “Layer 3” in the stacks of AS protocol. This is the base for the functions of AS, which shall be counter of the radio bearers and configuration of the lower layers pertaining to RRC signaling amid of the eNodeB and the UE.

![Figure 3.7: Control Plane protocol stack][20]

3.7 EPS bearers and QoS

In a conventional case, there shall be many applications which are running in a UE at a given sequential time, and also the likely hood that each of them is with a specific QoS factors. For instance, UE of a particular application engaged in VoIP shall be in need on stringent terms for its QoS, as is compared to any other setups which are for the other browser applications. In such cases as the need is to control the damage of packet loss rates the specificity changes accordingly with the each associated QoS terms.
In a broad classification, the bearers can be categorized into varied categories depending upon the nature of the QoS they are associated with.

**GBR** : This kind of bearers which are used for applications like VoIP, have an associated value of GBR which shall ensure a dedicated resources of transmission and are permanently dedicated with a bearer establishment :

When we consider few facts, Bit rates are quite higher than the usual case of the GBR resources when available. In a case of exemptions, MBR parameter, which is to be integrated to the bearers of GBR, do not implicit a particular bit rate.

Table 1: Standardized QCI's for LTE[21]

<table>
<thead>
<tr>
<th>QC</th>
<th>RESOURCE TYPE</th>
<th>PRIORITY</th>
<th>PACKET DELAY BUDGET (MS)</th>
<th>PACKET ERROR LOSS RATE</th>
<th>EXAMPLE SERVICES</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>GBR</td>
<td>2</td>
<td>100</td>
<td>$10^2$</td>
<td>Conversational voice</td>
</tr>
<tr>
<td>2</td>
<td>GBR</td>
<td>4</td>
<td>150</td>
<td>$10^4$</td>
<td>Conversational video (live streaming)</td>
</tr>
<tr>
<td>3</td>
<td>GBR</td>
<td>5</td>
<td>300</td>
<td>$10^4$</td>
<td>Non-conversational video (buffered streaming)</td>
</tr>
<tr>
<td>4</td>
<td>GBR</td>
<td>3</td>
<td>50</td>
<td>$10^4$</td>
<td>Real-time gaming</td>
</tr>
<tr>
<td>5</td>
<td>Non-GBR</td>
<td>1</td>
<td>100</td>
<td>$10^4$</td>
<td>IMS signaling</td>
</tr>
<tr>
<td>6</td>
<td>Non-GBR</td>
<td>7</td>
<td>100</td>
<td>$10^4$</td>
<td>Voice, video (live streaming), interactive gaming</td>
</tr>
<tr>
<td>7</td>
<td>Non-GBR</td>
<td>6</td>
<td>300</td>
<td>$10^4$</td>
<td>Video (buffered streaming)</td>
</tr>
<tr>
<td>8</td>
<td>Non-GBR</td>
<td>8</td>
<td>300</td>
<td>$10^4$</td>
<td>TCP-based (for example, WWW, e-mail), chat, FTP, p2p file sharing, progressive video and others</td>
</tr>
<tr>
<td>9</td>
<td>Non-GBR</td>
<td>9</td>
<td>300</td>
<td>$10^4$</td>
<td></td>
</tr>
</tbody>
</table>

The bearers of Non-GBR do not signify any specific bit rate. This can be resourceful for applications like transfer protocols of files or for web browsing. There is no need of any permanent allocation of bandwidth of resources to these bearers.

An EPS bearer must cross many interfaces as shown in figure 8 - the S5/S8 interface from P-GW to S-GW, the S1 interface, from S-GW to the eNodeB and the radio interface.
(also known as "LTE-Uu interface") from the eNodeB to the UE. Over each interface is EPS bearer mapped onto a lower layer carriers, each with its own carrier identity. Each node must keep track of the bond between the bearer IDs to its different interfaces [29] [30].

A S5/S8 bearer carrying packages of an EPS bearer between a P-GW and S-GW[38]. The S-GW stores a one-to-one mapping between an S1 bearer and a S5/S8 bearer. The holder is identified by the GTP tunnel IDs over both interfaces.

![Figure 3.8: LTE/SAE bearer across the different Interfaces][21][38].

The packets of an EPS bearer are transported by an S1 bearer between an S-GW and an eNodeB, and by a radio bearer between a UE and an eNodeB. An eNodeB stores a one-to-one mapping between a radio bearer ID and an S1 bearer to create the mapping between the two.

IP packets mapped to the same EPS bearer receive the same bearer-level packet forwarding treatment. In order to provide different bearer-level QoS, a separate EPS bearer must therefore be established for each QoS flow. User IP packets must then be filtered into the appropriate EPS bearers.

Packet filtering into different bearers is based on TFTs. The TFTs use IP header information such as source and destination IP addresses and Transmission Control Protocol (TCP) port
numbers to filter packets such as VoIP from web-browsing traffic, so that each can be sent down the respective bearers with appropriate QoS. An Uplink TFT (UL TFT) associated with each bearer in the UE filters IP packets to EPS bearers in the uplink direction. A Downlink TFT (DL TFT) in the P-GW is a similar set of downlink packet filters.
Chapter 4

Network Model and Implementation

This chapter discusses the network modeling, implementation and their configuration.

4.1 Network Model Assumption and Their Configuration

In this work as the purpose of our subject study involved, we have considered OPNET Modeler 17.1 as a choice of simulators for the simulation analysis. This section illustrates the network model used and their configuration in this study. In the consideration of 6 network scenarios are assumed to be modeled, which will be elaborately demonstrated in the up-coming sections. Scenarios 2 and 3 are modeled followed by baseline scenario 1 in which scenario 2 has been modeled to demonstrate the imposition of medium network load while scenario 3 is modeled with high network load. It is noted that Scenarios-1, 2 and 3 are configured without LTE scheduling techniques having implemented whereas Scenarios-4, 5 and 6 are modeled to demonstrate LTE scheduling techniques implementation followed by the respective scenarios 1, 2, and 3.

Figure 4.1: Baseline Topology
4.2 Baseline Scenario Description

In the Baseline network topology depicted in Figure 4.1, two eNodeBs (eNodeB_1 and eNodeB_3) act as the base stations. The interface between eNodeB and UEs is covered in E-UTRAN entire network. In eNodeB, in terms of EPS bear configuration, there is one call admission control which is distinct by a position of actions to determine if the call request can be accepted or rejected [36]. In eNodeB, Inactive Bearer Timeout set at 20 seconds that implies with GBR bearers if the GBR bearer goes through the admission control protocol. LTE Physical Profile is applied in both eNodeBs which allocates physical layer resources for both uplink and downlink shared channels. The physical profile is set to 1.4 MHz FDD. To identify the different eNodeB, it is important to set the eNodeBs’ ID. ID for eNodeB_1 is set to 1 while eNodeB_3 is set to 3. SONET/OC3 link operating at the rate of 148.61 Mbps is used to establish the connection between eNodeBs and EPC (Evolved Packet Core). In all of 6 network scenarios, 6 UEs under eNodeB_1 are considered as source nodes for video conferencing traffic while 6 UEs under eNodeB_3 are acting as designated destination source nodes. Application, LTE and Profile configuration and their parameters used in this baseline network scenario are discussed in the following section.

4.2.1 Network Components

Table 4.1 shows different components of the objects related to various network configuration devices that are used in order to model the LTE network.

<table>
<thead>
<tr>
<th>Object Name</th>
<th>Network Components</th>
<th>Objects’ name referred to the Network Topology</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ite_configuration node</td>
<td>LTE</td>
<td>LTE_Configuration</td>
</tr>
<tr>
<td>Application Config</td>
<td>Application</td>
<td>Application Definition</td>
</tr>
<tr>
<td>Profile Config</td>
<td>Profile</td>
<td>Profile Definition</td>
</tr>
<tr>
<td>LTE Workstation</td>
<td></td>
<td>UE (User Equipment)</td>
</tr>
<tr>
<td>Ite_e_node_b_slip4_adv</td>
<td></td>
<td>eNodeB_1 and eNodeB_3</td>
</tr>
<tr>
<td>EPC (Evolved Packet Core)</td>
<td></td>
<td>EPC</td>
</tr>
</tbody>
</table>

Table 4.1: Used components in the network
4.2.2 LTE Configuration Parameters

We have configured LTE physical profile 1.4 MHz FDD using LTE configuration object (lte_configuration). Store Profiles for both LTE Physical (PHY) configurations and EPS Bearer definitions has been set out using Lte_configuration, which is referenced by all LTE nodes (UEs) in the network. The parameters for LTE PHY profile are configured followed by Table 4.2.

Table 4.2: LTE FDD Profiles

<table>
<thead>
<tr>
<th>Index</th>
<th>Name</th>
<th>UL SC-FDMA Channel Configuration</th>
<th>DL OFDMA Channel Configuration</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>Default UL 1.4 Mhz</td>
<td></td>
<td>Default DL 1.4 MHz</td>
</tr>
<tr>
<td>0</td>
<td>LTE1.4MHz FDD</td>
<td>BaseFrequency 1.92 Ghz</td>
<td>BaseFrequency 2.11 Ghz</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Bandwidth 1.4 MHz</td>
<td>Bandwidth 1.4 MHz</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Cyclic Prefix Type Normal(7Symbols per Slot)</td>
<td>Cyclic Prefix Type Normal(7Symbols per Slot)</td>
</tr>
</tbody>
</table>

4.2.3 Traffic Generation

To generate the video conferencing traffic, two configuration objects, application definition and profile definition, are used and described in the following sections.

4.2.3.1 Video Conferencing Application Configuration Parameters

In our case, the "Application Configuration" node is used for generating video conferencing traffic. For example, "video conferencing traffic (videoconference (BE), videoconference(Exclnt Effort and videoconference(IntracMultimedia))" profiles refer to the video application. VideoConference(BE, Excellnt Effort and Interacvmultimedia) profiles are assigned in the "Profile Config" object. These profiles consist of different video properties which are configured as shown Table 4.3. Video traffic configuration parameters depicted in Table 4.3 and 4-4 are important to be noted here. For above mentioned 3 different video application profiles consist of Frame Inter-arrival Time Information, Frame Size Information and Type of Service (ToS). Frame Inter-arrival Time Information is defined as 10 frame/sec while Frame Size Information (incoming and outgoing stream frame sizes) in both directions is considered to be 1000 bytes. ToS has been configured for example, ToS for traffic in the videoconference(BE) profile has been set out to Best Effort while ToS for video traffic in the videoconference(Exclnt Effort)
profile is defined as Excellent Effort. ToS for traffic configured in the 3rd application profile referring to videoconference(IntractvMultimedia) is assigned as Interactive Multimedia. ToS will be used during the configuration of LTE Quality of Service (QoS).

Table 4.3: Video Traffic Configuration Parameters (Application Attributes)

Table 4.4: Video Traffic Configuration Parameters (Application Attributes)
Table 4.5: Network Load according to Video Traffic Configuration Parameters (Application Attributes)

<table>
<thead>
<tr>
<th>Scenarios</th>
<th>Profile Name</th>
<th>Frame Size Information</th>
<th>Frame Inter-arrival Time Info</th>
</tr>
</thead>
<tbody>
<tr>
<td>Low Load</td>
<td>videoconference (BE), videoconference(Excellent Effort and videoconference(Intr activMultimedia))</td>
<td>Frame Size Info</td>
<td>10 frames/sec</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Frame Inter-arrival Time Info</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>Incoming Stream Frame Size (Bytes)</td>
<td>constant (1000)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Outgoing Stream Frame Size (Bytes)</td>
<td>constant (1000)</td>
</tr>
<tr>
<td>Medium Load</td>
<td>videoconference (BE), videoconference(Excellent Effort and videoconference(Intr activMultimedia))</td>
<td>Frame Size Info</td>
<td>10 frames/sec</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Frame Inter-arrival Time Info</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>Incoming Stream Frame Size (Bytes)</td>
<td>constant (1000)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Outgoing Stream Frame Size (Bytes)</td>
<td>constant (1000)</td>
</tr>
<tr>
<td>High Load</td>
<td>videoconference (BE), videoconference(Excellent Effort and videoconference(Intr activMultimedia))</td>
<td>Frame Size Info</td>
<td>30 frames/sec</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Frame Inter-arrival Time Info</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>Incoming Stream Frame Size (Bytes)</td>
<td>constant (1000)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Outgoing Stream Frame Size (Bytes)</td>
<td>constant (1000)</td>
</tr>
</tbody>
</table>

4.2.3.2 Profile Configuration

With the help of Application Profiles assigned and demonstrated in section 4.2.3.1, profiles’ corresponding attributes are configured in way that the video conference traffic generation can be controlled as what time the traffic generation supposed to be started, ended and in how many
times the application is repeated during the simulation period. Looking at profile configuration shown in Table 4.6, video conference starts at 120 seconds after the simulation is started and ended at the end of simulation. Repeatability is Once at Start Time that means in our study we have kept it as the application will start as the simulation run and ends at the end of simulation without repeating the application generation.

Table 4.6: Profile Configuration (Video Conference Traffic)

4.2.4 Mobility Configuration by using trajectory settings

4.2.4.1 Trajectory Control in OPNET Modeler

OPNET Modeler has several ways to control the nodes’ mobility. Trajectory refers to the path specification for mobile nodes’ movement over the course of simulation. Mobility can be changed based on predefined trajectories or randomly selected path. There are two ways on assigning the node mobility [6] in what follows.

- Segment-Based Trajectories define movement using a series of pre-defined points.
- Vector-Based Trajectories define movement in terms of a bearing, ground speed, and ascent Rate.

For control the mobility of the UEs, segment-based variable-interval trajectory is used in our analysis. Segment-based trajectory consists of fixed interval and variable interval trajectories. As objective of our thesis work is concerned, variable-interval trajectory has been taken into
consideration for controlling the mobility of UEs in the proposed network scenario. Variable-interval based trajectory is stored in ASCII text file with a .trj extension [6] [7].

The structures of variable-interval based trajectory are given below. The *.trj file for variable interval trajectory has following structure:

Version: 5
Position_Unit: <position_unit>
Altitude_Unit: <altitude_unit>
Coordinate_Method: <coordinate_method>
locale: <locale>
Calendar_Start: <start_time>
Coordinate_Count: <coordinate_count>

# X Position, Y Position, Altitude, Traverse Time, Wait Time, Pitch, Yaw, Roll
<x_coord_0>, <y_coord_0>, <alt_0>, <trav_time_0>, <wait_time_0>, <pitch_0>, <yaw_0>, <roll_0>
<x_coord_1>, <y_coord_1>, <alt_1>, <trav_time_1>, <wait_time_1>, <pitch_1>, <yaw_1>, <roll_1>
...
<x_coord_n>, <y_coord_n>, <alt_n>, <trav_time_n>, <wait_time_n>, <pitch_n>, <yaw_n>, <roll_n>

4.2.4.2 Relative movement

Relative movement can be done in two ways, for example, sub-network itself can have movement in a certain direction and relatively child nodes under that sub-network can move as well. In our case, parent sub-network is kept stationary while child nodes’ respective positions (UEs connected eNodeB 3) are varied from one point to another point based on the assigned trajectory [6].

4.2.4.3 Defining segment-based trajectories

As far as objective of our analysis is concerned, we have considered worst case scenario by applying the mobility speed high in which segment-based trajectory is defined. Single segment for trajectory is moved forward towards in a particular direction. In OPNET, there is trajectory
file format demonstrated in section 4.2.4.1.1. Table 4.7 represents the predetermined trajectory configuration parameters and their corresponding values. Mobility for UEs under eNodeB_3 can be observed from Network Topology presented in Figure 4.1 and Table 4.7.

According to the predetermined Trajectory configuration parameters shown in Table 4.7, 2 coordinates are considered and their corresponding position-based locations have been defined. UEs’ under eNodeB_3 initial x position (0.000 km) and y position (0.000 km) are specified in the parent sub-network. ‘Distance’, ‘Traverse Time’ and ‘Ground Speed’ for the first position (0.00 km, 0.00 km) of UEs is not applicable. UEs altitude is defined as 0m. The ‘Wait Time’ appeared in Table 4.7 is used to keep UEs on hold for 2 minutes 20 seconds. The “Pitch” value is set to ‘Autocomputed’ while “Roll” Value is assigned to ‘unspecified’ referring to 0 degrees in which 0 degrees refer to the parallel to the ground. “Yaw” value is set to ‘autocomputed’ which will set such that UE_mobile orientation matches the motion vector for each trajectory. After the simulation started, UEs begin to move followed by predefined trajectory where UEs wait for 2 minutes 20 seconds at (0.00 km, 0.00 km). After 2 minutes and 20 seconds simulation time, UEs moves 7 km towards the north from (0.0 km, 0.0 km) where traverse time is 2 minute 20 seconds attaining the ground speed of UE_mobile, 111.85 kilometer per hour (km/hr) [6] [7].

Table 4.7: Trajectory Configuration

<table>
<thead>
<tr>
<th>Position_Unit:</th>
<th>kilometers (Specifies how the x_coord_n and y_coord_n values of positions are interpreted)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Altitude_Unit:</td>
<td>Meters</td>
</tr>
<tr>
<td>Coordinate_Method:</td>
<td>relative</td>
</tr>
<tr>
<td>Altitude_Method:</td>
<td>absolute</td>
</tr>
<tr>
<td>locale:</td>
<td>C (i.e., Reserved for future use. The only valid value is &quot;C&quot;.)</td>
</tr>
<tr>
<td>Calendar_Start:</td>
<td>unused</td>
</tr>
<tr>
<td>Coordinate_Count:</td>
<td>4</td>
</tr>
<tr>
<td>X Pos (n)</td>
<td>Distance (m)</td>
</tr>
<tr>
<td>0.000</td>
<td>0.000</td>
</tr>
<tr>
<td>0.000</td>
<td>7000</td>
</tr>
</tbody>
</table>

4.3 Simulation Scenarios

Network topology depicted in Figure 4-1 represents a baseline scenario which is used to create 6 scenarios (1, 2, 3, 4, 5 and 6) considering individual configuration parameters that regards to LTE scheduling techniques. Our proposed six network scenarios as follows and their corresponding configuration are discussed under the following sections.
4.3.1 Network Load Configuration

This section illustrates about 3 different networks. One of the key factors of this experiment is that it considers 3 different network loads corresponded to 6 proposed network scenarios. Varied network loads (i.e., low, medium and high) have been realized and configured in our simulation to better understand how the performance of Quality of Service (QoS) metrics is impacted when network load is changed along with varied inter-frequency interference and nodes mobility. Network load has been configured followed by the configuration parameters shown in Table 4.8.

<table>
<thead>
<tr>
<th>Scenarios</th>
<th>Uplink/Down Link Load</th>
<th>Up/Down Link Capacity</th>
<th>Network Load</th>
</tr>
</thead>
<tbody>
<tr>
<td>Scenario1_SchedulingNotApplied_LowLoad</td>
<td>480 Kbps</td>
<td>1 Mbps</td>
<td>48%</td>
</tr>
<tr>
<td>Scenario2_SchedulingNotApplied_MediumLoad</td>
<td>900 Kbps</td>
<td>1 Mbps</td>
<td>90%</td>
</tr>
<tr>
<td>Scenario2_SchedulingNotApplied_HighLoad</td>
<td>1899 Kbps</td>
<td>1 Mbps</td>
<td>180%</td>
</tr>
<tr>
<td>Scenario4_SchedulingApplied_LowLoad</td>
<td>480 Kbps</td>
<td>1 Mbps</td>
<td>48%</td>
</tr>
<tr>
<td>Scenario5_SchedulingApplied_MediumLoad</td>
<td>900 Kbps</td>
<td>1 Mbps</td>
<td>90%</td>
</tr>
<tr>
<td>Scenario6_SchedulingApplied_HighLoad</td>
<td>1899 Kbps</td>
<td>1 Mbps</td>
<td>180%</td>
</tr>
</tbody>
</table>

4.3.2 Scenario1_SchedulingNotApplied_LowLoad

Baseline scenario and its relevant configuration have been illustrated in the previous section. This scenario i.e., Scenario1_SchedulingNotApplied_LowLoad, is replicated from the baseline.
scenario presented in Figure 4.1. That means the entire network configuration applied to the baseline scenario has been kept similar in this scenario and in addition to that additionally network load is injected according to network load Table 4.8. In ‘low’ load network scenario, we have not imposed the background traffic in which only the explicit video traffic, 480 Kbps generated at the designated source workstations (that belongs to eNodeB_3). All the UEs () are connected and considered to source workstations. All 6 UEs (eNodeB 3) are having video conference with designated destination workstations, belonging to the eNodeB1, is contributed to the total network load injected at eNodeB level as about 48% when Uplink/Down link base frequency is 1.4 MHz.

4.3.3 Scenario2_SchedulingNotApplied_MediumLoad

This scenario_2 (i.e., Scenario2_SchedulingNotApplied_MediumLoad) is replicated from the scenario_1. All the configuration parameters used in scenario_1 are kept same except the degree of network load injected in the network changing from low to medium network load. Detail of network for this can be seen from Table 4.8. In ‘medium’ network load scenario2, 90% explicit traffic generated at source workstations (UEs) which belong the corresponding eNodeB_3.

4.3.4 Scenario3_SchedulingNotApplied_HighLoad

This scenario_3 (i.e., Scenario3_SchedulingNotApplied_HighLoad) is replicated from the scenario_1. All the configuration parameters used in scenario_1 is kept same except the network load changed from low load to high (see detail parameters in Table-4-8 that configured for High network load. In ‘high’ network load scenario (3), 180% explicit traffic generated at the source workstations (UEs).

4.3.5 Scenario4, 5, 6_SchedulingApplied (Low, Medium and High, respectively)

This section deals with the configuration of LTE QoS implementation and illustration with the help of other previously explained scenarios (1, 2 and 3). Scenarios (4, 5 and 6) are configured followed by scenarios 1, 2, 3 and similar with all the configuration parameters except LTE admission control systems. 1.4 MHz bandwidth is used in this simulation. Evolved Packet System (EPS) consists of Evolved UTRAN (E-UTRAN) and Evolved packet core (EPC).

EPS bearer presented in Table 4.9,10,11 is used to specify which bearer or channel users (UEs) want to use. For scenario_4, 5 and 6, 3 EPS bears are used corresponded with that we used four EPS bearer UE_1_1, UE_1_2, and UE_1_3 which corresponds with Excellent Effort, Multimedia Interactive and BE, respectively. The same sets of EPS bears are also used in all cases. QCI (QoS Class Identifier) is a scalar value which refers to a set of the parameters to determine packet forwarding characteristics.
The Traffic Forwarding Policy (TFP) defines a set of parameter at every node along the path between the end users. In QCI, the allowed value range is 1 to 9 where QCI value 1 to 4 represent GBR bearer and others are non-GBR bearers [11]. QCI values for Scenario_4, 5 and 6 that assigned to 6, 4, and 2 is corresponded with BE, Excellent Effort and Interactive multimedia bearer, respectively.

Table 4.9: EPS Bearer Definitions (Scenario 4)

Table 4.10: EPS Bearer Definitions (Scenario 5)

Table 4.11: EPS Bearer Definitions (Scenario 6)
4.3.6 Simulation Run

All the simulations run for 480 seconds, and all applications that generate the traffic (i.e. video conferencing) start simultaneously at 120 seconds of the simulated time, that is, every event has the same probability to occur at every value at 120 seconds. The simulation is implemented in OPNET Modeler 17.1 running on a HP laptop with Windows 7, Pentium IV 1.7 GHz with 2GB of RAM. These information regarding 9 different scenarios have been collected from OPNET after the simulation run is finished.

Table 4.12: Simulation Run
Chapter 5
Simulation Results Analysis

This chapter discusses about the statistical data corresponded to the obtained results from OPNET simulation. QoS performance metrics (i.e., Packet Delay Variation, Packet End-to-End Delay and Packet Sent/Received) are, of interest, presented and analyzed in this chapter.

It is noted that in this chapter, the following explicit name for the considered scenarios refers to the long form of scenarios’ names assigned left and their corresponding short name of the scenarios are given to the right after the symbol (→).

- Scenario1_SchedulingNotApplied_LowLoad  → Scenario1
- Scenario2_SchedulingNotApplied_MediumLoad  → Scenario2
- Scenario2_SchedulingNotApplied_HighLoad  → Scenario3
- Scenario4_SchedulingApplied_LowLoad  → Scenario4
- Scenario5_SchedulingApplied_MediumLoad  → Scenario5
- Scenario6_SchedulingApplied_HighLoad  → Scenario6

5.1 Performance of packet delay variation (PDV)

This section and subsequent subsections outline an evaluation on the performance of Packet Delay Variation, a QoS performance metric for the proposed network model presented in chapter 4. In computer networking, a performance metric, packet delay variation (PDV) is the difference in One-Way-Delay (OWD) between selected packets in a flow with any lost packets being ignored as defined in a draft of the IETF IPPM working group [1] [2].
5.1.1 Packet Delay Variation for Low Load Scenarios (1 and 4)

Figure 5.1: Video Conferencing Packet Delay Variation for Low Load Scenarios (1 and 4)

Figure 5.1 shows performances of the UE during a video conference under low load. In Scenario 1 and Scenario 4, the packet delay variations of the UE are shown. The X-axis represents the time and the Y-axis represents the corresponding packet delay variation. For all the equipment in Scenario 4, scheduling is applied whereas, for the equipment in Scenario 1, no scheduling is applied. The UE showing best effort in Scenario 4 has a considerably higher packet delay variation in comparison to that of Scenario 1. For the UE with excellent effort, we see that around the two minutes mark the UE which is scheduled (i.e. the red curve) has a higher packet delay variation (PDV) but at around the four minutes mark it starts rising and continues to have a higher PDV than the one which is not scheduled.

Table 5.1: Video Conferencing Packet Delay Variation for Low Load Scenarios (1 and 4)

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Object Name</th>
<th>Minimum</th>
<th>Average</th>
<th>Maximum</th>
<th>StdDev</th>
</tr>
</thead>
<tbody>
<tr>
<td>Scenario1</td>
<td>UE_1_1_GBR1(ExclntEffect)</td>
<td>0.00010050</td>
<td>0.00010053</td>
<td>0.00017774</td>
<td>0.00001628</td>
</tr>
<tr>
<td>Scenario1</td>
<td>UE_1_3_nonGBR1(BE)</td>
<td>0.00005150</td>
<td>0.00010045</td>
<td>0.00012853</td>
<td>0.00001452</td>
</tr>
<tr>
<td>Scenario1</td>
<td>UE_1_2_GBR2(inter-actvMultimedia)</td>
<td>0.00006339</td>
<td>0.00009421</td>
<td>0.00010808</td>
<td>0.00000785</td>
</tr>
<tr>
<td>Scenario4</td>
<td>UE_1_3_nonGBR1(BE)</td>
<td>0.00006341</td>
<td>0.00014859</td>
<td>0.00009555</td>
<td>0.00005031</td>
</tr>
<tr>
<td>Scenario4</td>
<td>UE_1_1_GBR1(ExclntEffect)</td>
<td>0.00010049</td>
<td>0.00011867</td>
<td>0.00014310</td>
<td>0.00000734</td>
</tr>
<tr>
<td>Scenario4</td>
<td>UE_1_2_GBR2(inter-actvMultimedia)</td>
<td>0.00004746</td>
<td>0.00010708</td>
<td>0.00012537</td>
<td>0.00001891</td>
</tr>
</tbody>
</table>
5.1.2 Packet Delay Variation for Medium Load Scenarios (2 and 5)

In Figure 5.2 shows the packet delay variations of different UE are shown under medium load. In Scenario 2 scheduling is not applied to any of the equipment but to all the UE under Scenario 5, scheduling has been applied. In case of best effort, the equipment which is scheduled has a higher packet delay variation than the one where no schedule is applied. For excellent effort, the UE which is not scheduled has a higher packet delay variation than the one which is scheduled. However, after the nine minutes mark, the packet delay variation for both the UE seem to be equivalent. In case of interactive multimedia, the scheduled equipment tends to have a higher packet delay variation in the start but it gradually lessens and eventually has a lower packet delay variation than the equipment which is not scheduled.
Table 5.2: Video Conferencing Packet Delay Variation for Medium Load Scenarios (2 and 5)

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Object Name</th>
<th>Minimum</th>
<th>Average</th>
<th>Maximum</th>
<th>StdDev</th>
</tr>
</thead>
<tbody>
<tr>
<td>Scenario2</td>
<td>UE_1_2_GBR2(Inter-actvMultimedia)</td>
<td>0.00009803</td>
<td>0.00015403</td>
<td>0.0001673</td>
<td>0.00001456</td>
</tr>
<tr>
<td>Scenario2</td>
<td>UE_1_1_GBR1(ExclntEffort)</td>
<td>0.00010593</td>
<td>0.00014592</td>
<td>0.0001932</td>
<td>0.00001014</td>
</tr>
<tr>
<td>Scenario5</td>
<td>UE_1_3_nonGBR1(BE)</td>
<td>0.00018610</td>
<td>0.00027217</td>
<td>0.0003303</td>
<td>0.00002209</td>
</tr>
<tr>
<td>Scenario5</td>
<td>UE_1_2_GBR2(Inter-actvMultimedia)</td>
<td>0.00011717</td>
<td>0.00014699</td>
<td>0.0001701</td>
<td>0.00001763</td>
</tr>
<tr>
<td>Scenario5</td>
<td>UE_1_1_GBR1(ExclntEffort)</td>
<td>0.00010869</td>
<td>0.00014577</td>
<td>0.0001614</td>
<td>0.00001427</td>
</tr>
</tbody>
</table>

5.1.3 Packet Delay Variation for High Load Scenarios (3 and 6)

In Figure 5.3 views the packet delay variation of different equipment under high load. The UE in Scenario 6 are scheduled, whereas, the UE in Scenario 3 are not. In case of best effort, the non-scheduled UE has a greater packet delay variation than that of the scheduled one. For excellent effort, the packet delay variation of the non-scheduled UE is much greater whereas, that of the scheduled UE is negligible. In case of interactive multimedia, the UE which is not scheduled has a substantially greater packet delay variation than that of the scheduled UE.
Table 5.3: Video Conferencing Packet Delay Variation for High Load Scenarios (3 and 6)

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Object Name</th>
<th>Min.</th>
<th>Avg.</th>
<th>Max.</th>
<th>StdDev</th>
</tr>
</thead>
<tbody>
<tr>
<td>Scenario3</td>
<td>UE_1_2_GBR2(Inter-actvMultimedia)</td>
<td>0.18372</td>
<td>11.803</td>
<td>12.397</td>
<td>3.7713</td>
</tr>
<tr>
<td>Scenario3</td>
<td>UE_1_1_GBR1(ExclntEffort)</td>
<td>0.07520</td>
<td>3.144</td>
<td>3.785</td>
<td>1.1070</td>
</tr>
<tr>
<td>Scenario3</td>
<td>UE_1_3_nonGBR1(BE)</td>
<td>0.01429</td>
<td>0.438</td>
<td>0.623</td>
<td>0.2264</td>
</tr>
<tr>
<td>Scenario6</td>
<td>UE_1_3_nonGBR1(BE)</td>
<td>0.013493</td>
<td>0.02686</td>
<td>0.06356</td>
<td>0.012095</td>
</tr>
<tr>
<td>Scenario6</td>
<td>UE_1_1_GBR1(ExclntEffort)</td>
<td>0.000178</td>
<td>0.00100</td>
<td>0.00180</td>
<td>0.000335</td>
</tr>
<tr>
<td>Scenario6</td>
<td>UE_1_2_GBR2(Inter-actvMultimedia)</td>
<td>0.000195</td>
<td>0.00066</td>
<td>0.00089</td>
<td>0.000166</td>
</tr>
</tbody>
</table>

5.2 Performance of packet end-to-end (e2e) delay

This section presents the performance of End-to-end (e2e) delay. The e2e delay, a QoS performance metric is defined as follows. Difference between the time of sending the packet by sender workstation and the time of receiving the packet by the receiver workstation for a successfully received packet is defined as one way end-to-end delay. That means the amount of time is taken by one packet to be successfully received by the receiving working station is called One-Way-Delay (OWD). The statistical data in terms of e2e delay performance for the proposed LTE network analysis have been discussed.
5.2.1 Packet End-to-End (e2e) Delay for Low Load Scenarios (1 and 4)

Figure 5.4 displays the end to end delay of equipment under a low load in a video conference. Scheduling has been applied in Scenario 4 but not in Scenario 1. When best effort is prevailing, the non-scheduled UE has a greater end to end delay than that of the scheduled equipment as the latter is prioritized. In case of excellent effort, the UE which is not scheduled, again, has a greater packet delay than the one which is scheduled. Circumstances differ a little with interactive multimedia, in which case it is evident that UE which is scheduled has a higher packet delay in comparison to the one which is not scheduled.

Table 5.4: Video Conferencing Packet End-to-End Delay for LowLoad Scenarios (1 and 4)

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Object Name</th>
<th>Minimum</th>
<th>Average</th>
<th>Maximum</th>
<th>StdDev</th>
</tr>
</thead>
<tbody>
<tr>
<td>Scenario1</td>
<td>UE_1_1_GBR1(ExclntEffort)</td>
<td>0.033150</td>
<td>0.033863</td>
<td>0.035831</td>
<td>0.0017077</td>
</tr>
<tr>
<td>Scenario1</td>
<td>UE_1_2_GBR2(Inter-actvMultimedia)</td>
<td>0.031662</td>
<td>0.032284</td>
<td>0.033257</td>
<td>0.0024141</td>
</tr>
<tr>
<td>Scenario1</td>
<td>UE_1_3_nonGBR1(BE)</td>
<td>0.028370</td>
<td>0.029076</td>
<td>0.029621</td>
<td>0.0022780</td>
</tr>
<tr>
<td>Scenario4</td>
<td>UE_1_2_GBR2(Inter-actvMultimedia)</td>
<td>0.033048</td>
<td>0.034518</td>
<td>0.037621</td>
<td>0.0026243</td>
</tr>
<tr>
<td>Scenario4</td>
<td>UE_1_1_GBR1(ExclntEffort)</td>
<td>0.030623</td>
<td>0.031085</td>
<td>0.034870</td>
<td>0.0017377</td>
</tr>
<tr>
<td>Scenario4</td>
<td>UE_1_3_nonGBR1(BE)</td>
<td>0.023237</td>
<td>0.023541</td>
<td>0.025598</td>
<td>0.0018206</td>
</tr>
</tbody>
</table>
5.2.2 Packet End-to-End (e2e) Delay for Medium Load Scenarios (2 and 5)

Figure 5.5 and looking at Table 5-5, we can see that the packet delay suffered by the user equipment in video conference under medium load. In Scenario 5, scheduling is applied and in scenario 2, no scheduling is applied. In case of best effort, the UE, which has no scheduling applied to it, has a much greater packet delay than the one which is scheduled and is thus prioritized. Where there is excellent effort, the situation is similar as the non-scheduled UE has a higher packet delay. However, in case of interactive multimedia, the UE being scheduled has a higher end-to-end delay than that of the non-scheduled one.

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Object Name</th>
<th>Minimum</th>
<th>Average</th>
<th>Maximum</th>
<th>StdDev</th>
</tr>
</thead>
<tbody>
<tr>
<td>Scenario2</td>
<td>UE_1_2_GBR2(Inter-actvMultimedia)</td>
<td>0.032194</td>
<td>0.034425</td>
<td>0.034540</td>
<td>0.0026983</td>
</tr>
<tr>
<td>Scenario2</td>
<td>UE_1_1_GBR1(ExclntEffort)</td>
<td>0.03154</td>
<td>0.033727</td>
<td>0.033746</td>
<td>0.0021474</td>
</tr>
<tr>
<td>Scenario2</td>
<td>UE_1_3_nonGBR1(BE)</td>
<td>0.031755</td>
<td>0.033079</td>
<td>0.036803</td>
<td>0.0030179</td>
</tr>
<tr>
<td>Scenario5</td>
<td>UE_1_2_GBR2(Inter-actvMultimedia)</td>
<td>0.033081</td>
<td>0.035199</td>
<td>0.038000</td>
<td>0.0034106</td>
</tr>
<tr>
<td>Scenario5</td>
<td>UE_1_1_GBR1(ExclntEffort)</td>
<td>0.028900</td>
<td>0.031626</td>
<td>0.031630</td>
<td>0.0024236</td>
</tr>
<tr>
<td>Scenario5</td>
<td>UE_1_3_nonGBR1(BE)</td>
<td>0.029192</td>
<td>0.029187</td>
<td>0.032649</td>
<td>0.0040351</td>
</tr>
</tbody>
</table>
5.2.3 Packet End-to-End (e2e) Delay for High Load Scenarios (3 and 6)

Figure 5.6 shows the end to end delay of equipment under a high load. Scheduling is applied in Scenario 6 and in Scenario 3, no scheduling is applied. For best effort, the UE which is not scheduled has a much higher end to end delay than the UE which is scheduled. In case of excellent effort, the UE which is scheduled, on an average, nearly has a 12 times lower packet delay than the UE which is not scheduled. While in case of interactive multimedia, the packet delay suffered by the non-scheduled UE is, on an average, almost 29 times greater than that of the scheduled equipment.

Table 5.6: Video Conferencing Packet End-to-End Delay for High Load Scenarios (3 and 6)

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Object Name</th>
<th>Min.</th>
<th>Avg.</th>
<th>Max.</th>
<th>StdDev</th>
</tr>
</thead>
<tbody>
<tr>
<td>Scenario3</td>
<td>UE_1_2_GBR2(Inter-activMultimedia)</td>
<td>0.9148</td>
<td>1.5879</td>
<td>6.8632</td>
<td>2.9831</td>
</tr>
<tr>
<td>Scenario3</td>
<td>UE_1_1_GBR1(ExclntEffort)</td>
<td>0.4096</td>
<td>0.5719</td>
<td>3.3912</td>
<td>1.3534</td>
</tr>
<tr>
<td>Scenario3</td>
<td>UE_1_3_nonGBR1(BE)</td>
<td>0.2064</td>
<td>0.2064</td>
<td>1.2122</td>
<td>0.4450</td>
</tr>
<tr>
<td>Scenario6</td>
<td>UE_1_3_nonGBR1(BE)</td>
<td>0.09222</td>
<td>0.09216</td>
<td>0.31091</td>
<td>0.08883</td>
</tr>
<tr>
<td>Scenario6</td>
<td>UE_1_2_GBR2(Inter-activMultimedia)</td>
<td>0.05421</td>
<td>0.05418</td>
<td>0.07361</td>
<td>0.01415</td>
</tr>
<tr>
<td>Scenario6</td>
<td>UE_1_1_GBR1(ExclntEffort)</td>
<td>0.04832</td>
<td>0.04828</td>
<td>0.06261</td>
<td>0.01662</td>
</tr>
</tbody>
</table>
5.3 Packet Sent and Received/Packet Loss Performance

Packet loss is one of the important QoS performance metrics for real-time application such as video conference as of a case in our analysis. To determine packet loss, the number of received packets is recorded and deducted from the number of sent packets giving the number of lost packet in a particular network. Under this section, we present the obtained statistical results and discuss how 3 different types of network load affect the performance of packet loss while video conferencing traffic is passing through the LTE network. Statistical data are collected and plotted accordingly. A comparative analysis in terms of packet loss for video conferencing traffic under 3 different types of network load (Low, Medium and High) is investigated and presented.

5.3.1 Packet Sent and Received for Low Load Scenarios (1 and 4)

The graphs 5.7 (a) and (b) show the sent and received traffic during a video conference under low load. For equipment of both Scenario 1 and Scenario 4, the average number of sent traffic is equal, i.e. 10,000. The least traffic loss is suffered by the Interactive Multimedia equipment in Scenario 4. As it is prioritized and as the load is low, the Interactive Multimedia equipment suffers a minimal packet loss of an average of 19 Bytes/Sec. In case of best effort, both the UE have received almost the same amount of packet with the scheduled one receiving a little more (70 Bytes/Sec). For excellent effort, packet loss for the non-scheduled UE has suffered an average of 31 Bytes/Sec higher than that of the scheduled one.
Table 5.7: Video Conferencing Packet Sent and Received in (Bytes/sec) for Low Load Scenarios (1 and 4) (in Bytes)

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Object Name</th>
<th>(ReceivedAndSent(Bytes/Sec))</th>
<th>Min.</th>
<th>Avg.</th>
<th>Max.</th>
<th>StdDev</th>
</tr>
</thead>
<tbody>
<tr>
<td>Scenario 1</td>
<td>UE_1_2_GBR2(InteractiveMultimedia)</td>
<td>9500</td>
<td>9908</td>
<td>10,000</td>
<td>129</td>
<td></td>
</tr>
<tr>
<td>Scenario 1</td>
<td>UE_1_1_GBR1(ExclntEffort)</td>
<td>9167</td>
<td>9863</td>
<td>10,000</td>
<td>168</td>
<td></td>
</tr>
<tr>
<td>Scenario 1</td>
<td>UE_1_3_nonGBR1(IE)</td>
<td>8833</td>
<td>9525</td>
<td>10,000</td>
<td>349</td>
<td></td>
</tr>
<tr>
<td>Scenario 4</td>
<td>UE_1_2_GBR2(InteractiveMultimedia)</td>
<td>9333</td>
<td>9941</td>
<td>10,167</td>
<td>118</td>
<td></td>
</tr>
<tr>
<td>Scenario 4</td>
<td>UE_1_1_GBR1(ExclntEffort)</td>
<td>9500</td>
<td>9894</td>
<td>10,167</td>
<td>124</td>
<td></td>
</tr>
<tr>
<td>Scenario 4</td>
<td>UE_1_3_nonGBR1(IE)</td>
<td>8667</td>
<td>9600</td>
<td>10167</td>
<td>351</td>
<td></td>
</tr>
</tbody>
</table>

Same amount of Video Config Traffic Sent for Scenario 1&4 (Bytes/sec) 10000

5.3.2 Packet Sent and Received for Medium Load Scenarios (2 and 5)

The graphs 5.8 (a) and (b) show the differences between traffic sent and traffic received by equipment under a medium load. All equipment under Scenario 5 is scheduled and those under Scenario 2 are non-scheduled. Observing Table 5.8, we can see that in case of best effort, the scheduled UE has suffered a loss of 25 more Bytes than the one which is not scheduled. In case of excellent effort, the non-scheduled equipment has a loss of 189 more Bytes than the one which is scheduled. For interactive multimedia, the non-scheduled UE has received higher packet than the scheduled one.
Table 5.8: Video Conferencing Packet Sent and Received in (Bytes/sec) for Medium Load Scenarios (2 and 5)

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Object Name (Sent and Received)</th>
<th>Min.</th>
<th>Avg.</th>
<th>Max.</th>
<th>StdDev</th>
</tr>
</thead>
<tbody>
<tr>
<td>Scenario2</td>
<td>UE_1_2_GBR2 (Inter-actvMultimedia)</td>
<td>13333</td>
<td>14554</td>
<td>15,000</td>
<td>363</td>
</tr>
<tr>
<td>Scenario2</td>
<td>UE_1_1_GBR1 (ExclntEffort)</td>
<td>12833</td>
<td>14440</td>
<td>15,167</td>
<td>420</td>
</tr>
<tr>
<td>Scenario2</td>
<td>UE_1_3_nonGBR1 (BE)</td>
<td>11833</td>
<td>13860</td>
<td>14,833</td>
<td>583</td>
</tr>
<tr>
<td>Scenario5</td>
<td>UE_1_1_GBR1 (ExclntEffort)</td>
<td>13667</td>
<td>14629</td>
<td>15,167</td>
<td>299</td>
</tr>
<tr>
<td>Scenario5</td>
<td>UE_1_2_GBR2 (Inter-actvMultimedia)</td>
<td>12833</td>
<td>14527</td>
<td>15,000</td>
<td>434</td>
</tr>
<tr>
<td>Scenario5</td>
<td>UE_1_3_nonGBR1 (BE)</td>
<td>11500</td>
<td>13835</td>
<td>15,000</td>
<td>702</td>
</tr>
</tbody>
</table>

Traffic Sent for Scenarios 2 & 5 (Average) 14991 Bytes/Sec

5.3.3 Packet Sent and Received for High Load Scenarios (3 and 6)

![Figure 5.9: (a) Packet Received and (b) Sent for High Load Scenarios (3 and 6)](image)

The graphs 5.9 (a) and (b) represent the packet sent and received and their differences in a video conference under a high load. In each scenario the average sent traffic is identical to each other. Scheduling is applied in Scenario 6 and not applied in Scenario 3. The highest loss of packet is suffered when there is best effort is persisting. The non-scheduled UE under best effort, however, has a higher packet loss attaining 700 Bytes/Sec than the scheduled one. In case of excellent effort, the scheduled UE is assigned to higher priority and thus has received 1902 Bytes/Sec higher packet than the non-scheduled UE. For interactive multimedia, the non-scheduled UE, on an average, has suffered a loss of 636 Bytes/sec than the scheduled UE.
Table 5.9: Video Conferencing Packet Received (Bytes/sec) for High Load Scenarios (3 and 6)

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Object Name (-Received (Bytes/sec))</th>
<th>Min.</th>
<th>Avg.</th>
<th>Max.</th>
<th>StdDev</th>
</tr>
</thead>
<tbody>
<tr>
<td>Scenario3</td>
<td>UE_1_1_GBR1(ExclIntEffort)</td>
<td>17667</td>
<td>25063</td>
<td>39,833</td>
<td>3425</td>
</tr>
<tr>
<td>Scenario3</td>
<td>UE_1_2_GBR2(Inter-actvMultimedia)</td>
<td>13167</td>
<td>25060</td>
<td>40,500</td>
<td>3797</td>
</tr>
<tr>
<td>Scenario3</td>
<td>UE_1_3_nonGBR1(BE)</td>
<td>18000</td>
<td>24256</td>
<td>31,833</td>
<td>2066</td>
</tr>
<tr>
<td>Scenario6</td>
<td>UE_1_1_GBR1(ExclIntEffort)</td>
<td>20167</td>
<td>26965</td>
<td>29500</td>
<td>1511</td>
</tr>
<tr>
<td>Scenario6</td>
<td>UE_1_2_GBR2(Inter-actvMultimedia)</td>
<td>15500</td>
<td>25896</td>
<td>29,000</td>
<td>2033</td>
</tr>
<tr>
<td>Scenario6</td>
<td>UE_1_3_nonGBR1(BE)</td>
<td>20000</td>
<td>24956</td>
<td>30,667</td>
<td>1844</td>
</tr>
</tbody>
</table>

Table 5.10: Video Conferencing Packet Sent (Bytes/sec) for High Load Scenarios (3 and 6)

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Object Name (-Sent (Bytes/sec))</th>
<th>Min.</th>
<th>Avg.</th>
<th>Max.</th>
<th>StdDev</th>
</tr>
</thead>
<tbody>
<tr>
<td>Scenario3</td>
<td>UE_1_1_GBR1(ExclIntEffort)</td>
<td>30000</td>
<td>30029</td>
<td>30167</td>
<td>63</td>
</tr>
<tr>
<td>Scenario3</td>
<td>UE_1_2_GBR2(Inter-actvMultimedia)</td>
<td>30000</td>
<td>30029</td>
<td>30167</td>
<td>63</td>
</tr>
<tr>
<td>Scenario3</td>
<td>UE_1_3_nonGBR1(BE)</td>
<td>29833</td>
<td>30027</td>
<td>30167</td>
<td>67</td>
</tr>
<tr>
<td>Scenario6</td>
<td>UE_1_1_GBR1(ExclIntEffort)</td>
<td>29833</td>
<td>30027</td>
<td>30167</td>
<td>66</td>
</tr>
<tr>
<td>Scenario6</td>
<td>UE_1_2_GBR2(Inter-actvMultimedia)</td>
<td>29833</td>
<td>30027</td>
<td>30167</td>
<td>67</td>
</tr>
<tr>
<td>Scenario6</td>
<td>UE_1_3_nonGBR1(BE)</td>
<td>29667</td>
<td>30027</td>
<td>30167</td>
<td>76</td>
</tr>
</tbody>
</table>
Chapter 6
Conclusion and Future Work

This thesis focuses and analyzes the QoS effects based on how the radio resources which are available should be allocated to various users for the particular principle in relation to the users precedence basis over LTE. In the context of LTE networks, schedulers use predefined algorithms based on origin to differentiate the flow of traffic. As our research objective for this study is concerned, we have investigated the performance of delivering video (video conferencing) traffic in the LTE network considering a comparative study between GBR (Guaranteed Bit Rate) and Non-GBR (non – Guaranteed Bit Rate) carriers with the priority for users. We have presented a simulation-based study of LTE performance with the focus on the scenarios of UL (uplink) and DL (downlink) for video conferencing with web traffic. LTE packet scheduler should intelligently allocate radio resources to mobile User Equipment’s (UEs) such that the LTE network adheres to its performance requirements. In our thesis work, we conduct a comprehensive performance evaluation of LTE scheduling algorithms for real-time application such as video conferencing traffic. The evaluation is carried out using the OPNET simulator. In order to analyze the performance LTE scheduling algorithm, our analysis involved with LTE Admission Control is twofold. First, 6 scenarios have been modeled in way that 3 of 6 scenarios deals with no LTE admission control techniques applied in the proposed network models while other 3 scenarios deals with LTE scheduling techniques applied. Secondly, video conferencing sessions are configured between two LTE cells with same number of UEs in which all UEs under each cell in the entire proposed network modeling scenarios. In order to make our evaluation more realistic we have applied various network loads so that we can observe how LTE scheduling techniques work its best in the case of highly loaded network.

According to the objectives of this thesis, three research questions set out in chapter 1 were aimed to understand and investigate the performance of video traffic considering three QoS metrics (i.e., Packet Delay Variation (PDV), Packet End-to-End Delay and Packet Sent and Received) when GBR and NGBR scheduling algorithms are applied in the network. Research Questions included in chapter 1 are as follows.
1) Question 1: What is the impact on the packet end-to-end delay for video conferencing when NGBR and GBR bearer are established with mobility in LTE network under varied network load?

2) Question 2: What is the impact on the packet loss performance for video conferencing when NGBR and GBR bearer are established with mobility in LTE network under varied network load?

3) Question 3: What is the impact on the packet delay variation performance for video conferencing when NGBR and GBR bearer are established with mobility in LTE network under varied network load?

Question 1 was about the performance of the packet delay variation for video conferencing when NGBR and GBR bearer are established with mobility in LTE network under varied network load. The obtained results involved in this question were discussed in chapter 5.

In the case of low load scenarios (1 and 4) and PDV, when scheduling applied, PDV for UE_1_1_GBR1 (ExclntEffort), UE_1_2_GBR2 (InteractvMultimedia), and UE_1_3_nonGBR1 (BE), being in the low load scenarios attains an average of 0.00014859 s (seconds), 0.00011867s and 0.00010708s, respectively which are less than that of when scheduling not applied scenario. For example, it is found that PDV of UEs without scheduling implementation in the low load scenarios gains about 0.00010053s, 0.00010045s and 0.00009421s, correspondingly.

Considering medium load network scenarios (2 and 5), we found that when scheduling is applied, PDV for UE_1_1_GBR1 (ExclntEffort), UE_1_2_GBR2 (InteractvMultimedia), and UE_1_3_nonGBR1 (BE) attains an average of 0.00027217s, 0.00014699s and 0.00014577s, respectively, which are less than that of when scheduling not applied case. For example, it is found that PDV of UEs without scheduling implementation in the low load scenarios gains about 0.00010053s, 0.00010045s and 0.00009421s, correspondingly.

High load network scenarios (3 and 6), we found that when scheduling applied, PDV for UE_1_1_GBR1 (ExclntEffort), UE_1_2_GBR2 (InteractvMultimedia), and UE_1_3_nonGBR1 (BE) attains an average of 0.02686s, 0.00100s and 0.00066s, respectively, which are less than that of when scheduling not applied case. For example, it is found that PDV of UEs without scheduling implementation in the low load scenarios gains about 0.00020903s, 0.00015403s and 0.00014592s, correspondingly.
Question 2 was – what is the impact on the packet end-to-end delay for video conferencing when NGBR and GBR bearer are established with mobility in LTE network under varied network load. In the case of low load scenarios and E2E, when scheduling applied, E2E for UE_1_1_GBR1 (ExclntEffort), UE_1_2_GBR2 (InteractvMultimedia), and UE_1_3_nonGBR1 (BE) in low load scenarios attains an average of 0.034518s, 0.031085s, and 0.023541s, respectively which are less than that of when scheduling not applied case. For example, it is found that E2E of UEs without scheduling implementation in the low load scenarios gains about 0.033863s, 0.032284s and 0.029076s, correspondingly. Considering medium load network scenarios (2 and 5), we found that when scheduling applied, E2E for UE_1_1_GBR1 (ExclntEffort), UE_1_2_GBR2 (InteractvMultimedia), and UE_1_3_nonGBR1 (BE) attains an average of 0.035199s, 0.031626s and 0.029187s, respectively, which are less than that of when scheduling not applied case. Scheduling not applied case, for example, it is found that E2E of UEs without scheduling implementation in the low load scenarios gains about 0.034425s, 0.033727s and 0.033079s, correspondingly. High load network scenarios (3 and 6), we found that when scheduling applied, E2E for UE_1_1_GBR1 (ExclntEffort), UE_1_2_GBR2 (InteractvMultimedia), and UE_1_3_nonGBR1 (BE) attains an average of 0.09216s, 0.05418s and 0.04828s, respectively, which are less than that of when scheduling not applied case. Scheduling not applied case, for example, E2E of UEs gains about 1.5879s, 0.5719s and 0.2064s, correspondingly.

Question 3 was – what is the impact on the packet loss performance for video conferencing when NGBR and GBR bearer are established with mobility in LTE network under varied network load? Detailed examination of this question and corresponding findings presented in chapter 5 reveals that low load scenarios(1 and 4) and packet sent/received (in bytes). When scheduling applied, packet loss for UE_1_1_GBR1 (ExclntEffort), UE_1_2_GBR2 (InteractvMultimedia), and UE_1_3_nonGBR1 (BE) in low load scenario 1, is found to be 0.33% 0.32% and 0.33%, respectively, which is more than that of scenario 4 ( scheduling applied scenario). Packet loss for UE_1_1_GBR1 (ExclntEffort), UE_1_2_GBR2 (InteractvMultimedia), and UE_1_3_nonGBR1 (BE) in medium load scenario 2( scheduling not applied) was appeared to be 0.51%, 0.60% and 0.51% which is more than that of Scenario 5 for corresponding UEs. Finally, packet loss in high load scenario 3 for UE_1_1_GBR1
(ExclntEffort), UE_1_2_GBR2 (InteractvMultimedia), and UE_1_3_nonGBR1 (BE) is found to be 3%, 4% and 8% more than that of scheduling applied scenario 6.

**Future work**

In our future work, we wish to emphasize on the performance of real-time applications considering large-scale-LTE networks along with all of scheduling algorithms. To that extent, we would also like to focus on some key issues related with LTE Quality of Service (QoS) considering real-time and non-real-time applications.
BIBLIOGRAPHY


[34] OPNET Technologies. OPNET Modeler Product Documentation Release 16.0. 2010


