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ENSC 833: NETWORK PROTOCOLS AND PERFORMANCE

Performance analysis of LTE for Voice and Video Content

Final Report

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ABSTRACT

The increasing use of wireless devices, such as smart phones, handheld devices, etc. and the emergence of numerous user applications supported by various mobile systems in the other hand, have urged the need for higher capacity and greater speed than the existing cellular communication networks. Long term evolution (LTE), referred to sometimes as the fourth generation technology (4G), has earned a rapid rise in popularity over the last few years as a leading technology considered for adoption to fulfill these needs in addition to its capability to provide high quality multimedia services.

The aim of this project is to investigate the potentials and limitations of LTE technology as a communication media for streaming audio and video content. We will analyze the quality of service (QoS) performance and its effects when prioritization and differentiation mix of traffic. We will study the effectiveness of the LTE standard to handle applications requirements in terms packet delay, delay jitter and throughput, with various design parameters, using Riverbed Modeler 18.5.

1. INTRODUCTION

The telecommunications industry is changing rapidly as the expectation and requirements for wireless communication systems continue to grow and services are drastically shifting from voice to data and from circuit-switched to packet-switched ones. In the recent years we have seen massive growth of data – from/to mobile users – that has driven standardization board towards the development of cellular network architecture.

The International Telecommunication Union (ITU-R) announced the fourth generation (4G) mobile communication requirements, named as the International Mobile Telecommunications-Advanced (IMT-Advanced), in 2008. The Third Generation Partnership Project (3GPP) towards IMT-Advanced was named as LTE and standardized in release 8 [1] as one of the milestones in the direction to offer richer user experience, higher data rates and more improved performances.

In today wireless networks, there is unprecedented emphasize on data and video traffic as emergent mobile devices, like iPhone, are more powerful, support a variety of applications such as VoIP, video conference, multi-player games, etc. and are capable to display high quality video contents. However, these new technological devices and applications enabled require a higher throughput, wider bandwidth and smaller delay in order to provide good end user quality of experience.

The aim of this research project is to analyze the performance of delivering video/audio streaming traffic over this access network, based on QoS-aware mechanism that LTE standard provides, in order to measure:

- End-to-end delay;
- Packet jitter;
- Throughput.

Since mobile users have a tendency, nowadays, to exploit several services at the same time, we will investigate the performance of LTE when delivering different services to the same mobile user in the same time and observe the effect of traffic prioritization and differentiation. We present an LTE performance simulation study with the focus on downlink (DL). Several-service scenarios with Voice over IP (VoIP), video and FTP service are represented. We use for this task the Riverbed modeler release 18.5.

The outline of this report is as follows. In Section 2, we provide description of LTE technology and describe the network design respectively. Simulation results are described in Section 3. We conclude and discuss about future work in Section 4.

2. MAIN SECTION

In this section, we present an overview of LTE technology and related works. Then we briefly describe the network configuration that is simulated in this project.

I. LTE Overview

LTE is specified by the 3GPP on the way towards fourth-generation mobile in release 8. The adaptation of LTE standard to the requirements of 4G mobile communications is performed later through the LTE advanced, in 3GPP release 9 and release 10 [3]. The main goal behind LTE is to design a network architecture that will be based on packet switching services unlike the earlier generation circuit switching networks. In release 8, the 3GPP defined a new physical layer based on orthogonal frequency division multiple access (OFDMA) for the downlink and single carrier frequency division multiple access (SC-FDMA) for uplink.

Among others, LTE design targets include [4], [5]:

- Mobile access (for moving vehicles) at speeds up to 350Km/h
- Throughput 100Mb/s in downlink increasing to a maximum of 326.4Mb/s using MIMO 4x4 within 20MHz bandwidth.
- Throughput 50Mb/s in uplink increasing to a maximum of 86.4Mb/s using MIMO 4x4 within 20MHz bandwidth.

A typical architecture of the LTE system is depicted in Figure 1 below.

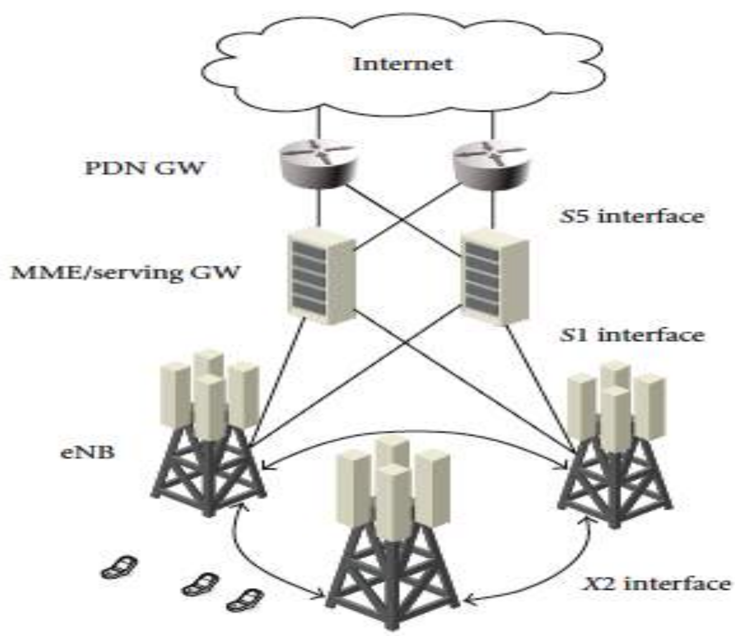

Figure 1 - LTE system architecture [5]

The network architecture called the Evolved Packet System (EPS) has a flat IP based architecture and is divided into the Evolved Universal Terrestrial Radio Access Network (E-UTRAN) and Evolved Packet Core (EPC). The overall architecture consists of the following elements:

- User Equipment (UE)
- Evolved-UTRAN (E-UTRAN)

- Evolved Packet Core (EPC)
- Mobility Management Entity (MME)
- Serving Gateway (S-GW)
- PDN Gateway (P-GW)
- Policy Control and charging Rules Function (PCRF)
- Home Subscriber Support (HSS)

LTE's packet switching offers a seamless Internet Protocol between User Equipment and P-GW without any disruption to users even during mobility. In radio access network (E-UTRAN) all services, including real-time, are supported over shared packet channels. E-UTRAN consists of one single entity known as eNodeB, which represents the base station that connects and controls the activities of the UE with EPC. ENodeB has all necessary functionalities for LTE radio access network including those related to radio resource management [5]. The relationship between the UE, EPC and E-UTRAN is depicted in Figure 2.

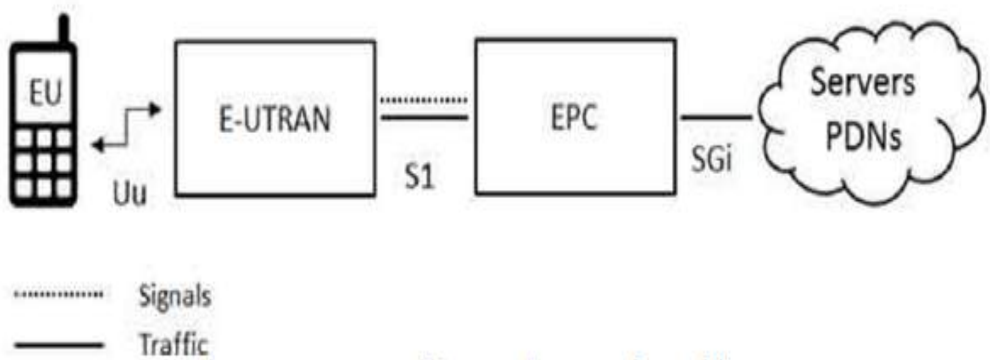

Figure 2 - LTE High Level Network Architecture [7]

The new core network (Evolved Packet Core) is an evolution of third generation systems node and it only covers the packet-switched domain. The EPC has three logical nodes: P-GW, S-GW and MME, besides some other supporting nodes: HSS, PCRF. P-GW allocates IP address to various UEs. S-GW acts as a mobility anchor for inter-working with other 3GPP GSM [4].

The Evolved Packet Systems (EPS) offers IP connectivity between UE and PDN for accessing the Internet and running various other services such as VoIP. The EPS architecture is comprised of the CN, E-UTRAN radio access network. CN provides access to external packet IP networks. CN also ensures security, privacy, QoS, and mobility. The LTE radio access is based on OFDM and provides highly flexible bandwidth from 1.4 MHz to 20 MHz. It also supports TDD (time division multiplexing) and FDD (frequency division multiplexing) [5].

II. Quality of service managements

Quality of Service (QoS) involves the data delivery between two end nodes with certain performances such as packet loss, jitter, latency, and bit rate. In LTE standard, an end-to-end class based QoS has been defined based on two main elements:

- Evolved Packet System (EPS) bearers.
- Quality of Service Class Identifiers (QCI).

EPS bearers define a connection-oriented virtual transmission channels carried out on a single Packet Data Network (PDN) connection. A default bearer, which provides a simple connectivity with no QoS policies, is created when user equipment is going to be attached to LTE networks. However, a dedicated bearer with QoS guaranteeing can be activated by LTE if a particular service needs QoS. Figure 3 shows the relationship between EPS bearer and PDN connection.

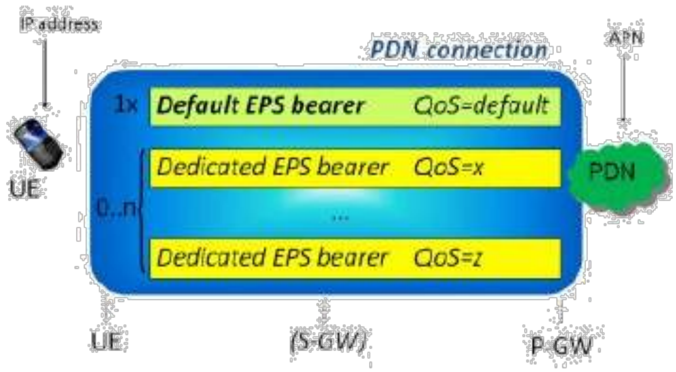

Figure 3 - Relationship between PDN connection and EPS bearer [10]

Network services are usually classified into real-time and non real-time services. Hence the bearers are classified into two categories (based on the offered QoS): Guaranteed Bit Rate (GBR) and non-GBR bearers.

GBR bearers have minimum rate guarantees and are required to go through admission control when their radio bearers are created. Dedicated transmission resources are permanently allocated during the bearer establishment thus, GBR bearer is suitable to provide services like voice and video telephony. However, non-GBR bearers are best effort bearers with no resource guarantees at all. Therefore are suitable for applications such as FTP file transfer, web browsing, email etc.

Each bearer is assigned a quality of service class identifier (QCI) and an Allocation and Retention Priority (ARP) in order to meet intended QoS requirements at the radio interface. Where, each QCI is characterized by priority, packet delay budget, and acceptable packet loss rate. Bearers with higher priority value will be destroyed first in case of congestion. Packet error loss rate and packet delay budget represent the maximum acceptable rate of IP packets that are not successfully received by the PDCP layer and the maximum acceptable end-to-end delay between the UE and the PDN-GWT respectively. The last parameter is applicable only for GBR bearers. The set of standardized QCIs and their QoS requirements are provided in Table 1.

The allocation and retention priority of the bearer is used during call admission control and new bearer establishment requests for GBR bearers only. However, packet forwarding treatment (e.g., scheduling policy, queue management policy, rate control policy etc.) are instead, determined by QoS parameters such as QCI, GBR etc.

Table 1 EPS Bearer Definitions [6]

QCI	Resource Type	Priority	Packet Delay Budget	Packet Error Loss Rate	Example Services
1	GBR	2	100 ms	10^{-2}	Conversational Voice
2		4	150 ms	10^{-3}	Conversational Video (live streaming)
3		3	50 ms	10^{-3}	Real-Time Gaming
4		5	300 ms	10^{-6}	Non-Conversational Video (buffered streaming)
5	Non-GBR	1	100 ms	10^{-6}	IMS Signaling
6		6	300 ms	10^{-6}	• Video (Buffered Streaming) • TCP-based (e.g., web, e-mail, chat, FTP, point-to-point file sharing, progressive video, etc.)
7		7	100 ms	10^{-3}	• Voice • Video (Live Streaming) • Interactive Gaming
8		8	300 ms	10^{-6}	• Video (Buffered Streaming) • TCP-based (e.g., web, e-mail, chat, FTP, point-to-point file sharing, progressive video, etc.)
9		9			

III. Related work

Actually, several papers deal with the performance evaluation of LTE. In [5] the main technical features of LTE and LTE advanced standard are described. The capacity of LTE systems is analyzed in terms of maximum achievable throughput and cell capacity distribution.

In [9], the authors studied user-perceived quality of service (QoS), for a mobile VoIP application in LTE, using various system bandwidths and codec types. Simulations were achieved using the OPNET environment. Results, presented in terms of Mean Opinion Score (MOS) values, showed that a variety of codec like G.711 and GSM may be used for delivering voice services with good enough quality of experience (QoE). In [10], the authors analyzed the influence of voice codec on end-to-end voice over LTE performance using several voice codec (AMR 12.2k, IS 641, etc.). In this study type of service (ToS) offered by LTE network through quality of service class identifiers (QCI) and EPS bearers are considered to be best effort only. Here again simulation was conducted under OPNET modeler environment. The results showed end-to-delay, voice traffic sent and received, voice packet delay variation as well as LTE downlink /uplink delay and underlines the better performances reached by G.711 and GSM EFR codecs.

The impact of differentiation and prioritization of delay-critical traffic like VoIP over other delay-insensitive intensive traffic like web surfing or download has been analysed in [11]. The performance of prioritized VoIP is compared with Best Effort (BE) VoIP. Theoretical and empirical study using Matlab tool are conducted. Simulation results showed the marginal degradation of delay-insensitive services while prioritizing VoIP service due to small VoIP packet sizes at the price of more efficient radio resource utilization.

In [12], the possibility of developing a multi-standard system has been investigated, using OPNET Modeler, where performance of multiple radio technologies cooperating was evaluated. Authors model investigate the packet scheduling algorithms based on channel state feedback. Network fairness and maximum throughput difference are analysed compared to a wireless homogeneous network.

In [13], authors have investigated video streaming over LTE and studied the impact of 3D video formats and their average encoding on the quality of service experienced by the end users. It should be noted that the most important aspect of characterizing 3D streams is the depth of perception. This later requires the transmission of additional information as well as more bandwidth and a lower loss probability. In this study a test-bed was used to calculate the ratio of packets lost, peak signal to noise ratio, delay and goodput were adopted for measuring QoS and QoE. The results underlined the effect of higher network load on quality of multimedia services offered to mobile users.

In [14], authors investigate video quality impact on the air-interface video capacity. Video capacity for LTE in the context of real-time video on the downlink was evaluated using C/C++ built simulator. The video capacities have been estimated for different system deployment scenarios and video data rates. In [15], authors analyzed the network traffic behavior of playing videos on mobile devices using a Test bed.

In this project we will study the impact of transmitting various services in the same time from the same user equipment. QoS traffic differentiation will be considered while observing the effect of prioritizing VoIP traffic over other services. For this purpose we will use the G.711 and GSM EFR codecs highlighted in the previous literature reviews for our VoIP study.

IV. Evaluation Platform

Riverbed modeler was chosen to design network model and assess the performance in real scenarios. It is proprietary simulation software based on object oriented and Discrete Event System (DES). Riverbed modeler offers an intuitive graphical interface which allows users to work and view the results easily. Moreover, it offers the possibility to display and plot different kind of graphs using analyze time series, probability, histograms, etc. For these reasons we chose modeler among different simulators and in this project, we used Riverbed modeler 18.5 for designing and developing simulation model.

V. LTE Simulation Model

In this section we will describes the network model implemented in Riverbed Modeler, as well as its parameters.

Figure 4 bellow shows topology of simulated LTE network in a given area. Our model is composed of two subnets: an LTE client subnet composed of four cells and a server subnet that provisions on demand video and ftp services to the mobile users.

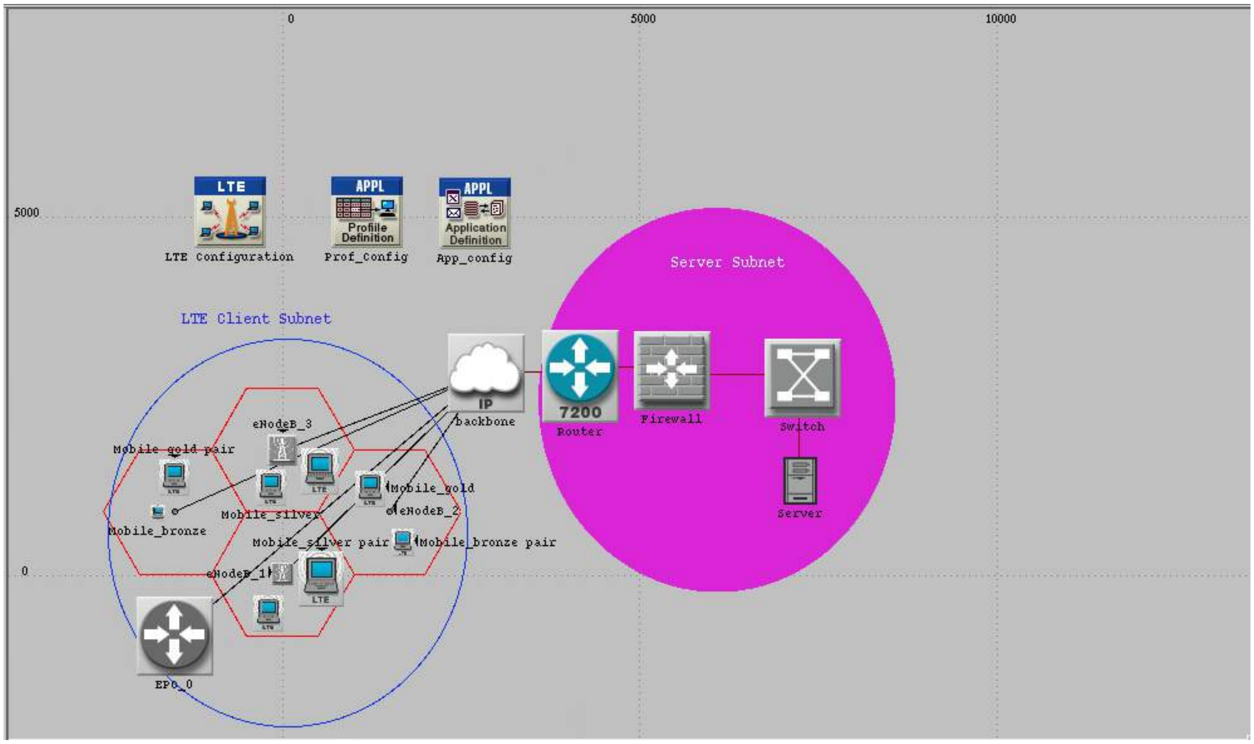

Figure 4 - Simulation model network topology

LTE client subnet includes two LTE mobile users and one eNodeB on each cell. It includes also an Evolved Packet Core (EPC) node connected to the backbone internet via a PPP_DS3 link which represents the Ethernet connection operating 44.736 Mbps.

LTE mobile users access the same on demand services from server and communicate with each other for VoIP. Since our goal is to observe the performance of LTE network while prioritizing and differentiating traffic we defined different class of service for each mobile user. Where, each service class defines an EPS bearer flow that ranges from dedicated bearer with QoS guaranteeing to best effort service flows. Mobile users with the same class definition communicate together for the VoIP service.

The server subnet provisions a server capable of streaming stored audio, video and FTP, contents to clients on request. The video server resides on a 100Mbps IP network behind a firewall which is connected to an access router. Router is connected to the Internet via a 1 GB Ethernet link.

Our model includes also three management nodes:

- App_config: application configuration node consists of name and description table for different parameters for different application such as voice and FTP.
- Profile_config: profile configuration node is used for creating a user profile which can produce application layer traffic.
- LTE_configuration: LTE attribute definer node is used to keep PHY pattern and EPS bearer explanation in the network.

Different Scenarios have been considered where each one includes a specific definition of applications and service classes to captures the QoS requirements.

VI. LTE Configuration

The LTE specific configuration involved different areas. We will present the definition of each of them in this section.

1 General Configuration Parameters

The general parameters used in the process of all simulation scenarios are detailed bellow. LTE Mobile users' configuration is modeled following parameters listed in Table 2.

Table 2 User Equipment Configuration

Parameter Description	Parameter Values
Antenna Gain	-1 dBi
Modulation and coding scheme	9
Multipath Channel mode (Downlink)	LTE OFDMA ITU Pedestrian B
Multipath Channel mode (Uplink)	LTE SC-OFDM ITU Pedestrian B
Pathloss	Free space

One of the important entities is the mobility configuration, which is modeled as pedestrian user with limited mobility and speed. Figure 5 shows the mobile user configuration's attributes.

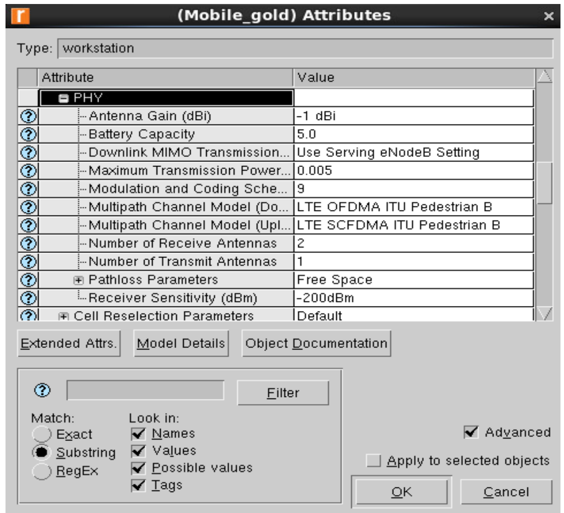

Figure 5 - LTE mobile user configuration

eNodeB and EPC network are modeled following parameters listed in Table3.

Table 3 LTE Network Settings

	Parameter Description	Parameter Values
eNodeB	LTE bandwidths	20 MHz
	Duplex mode	FDD
	eNodeB antenna gain	15 dBi
	Receive / Transmit antennas	2/2
	Receiver sensitivity	-200 dBm
EPC	N3 buffer size	8192 bytes

Figure 6 shows the eNodeB configuration's attributes.

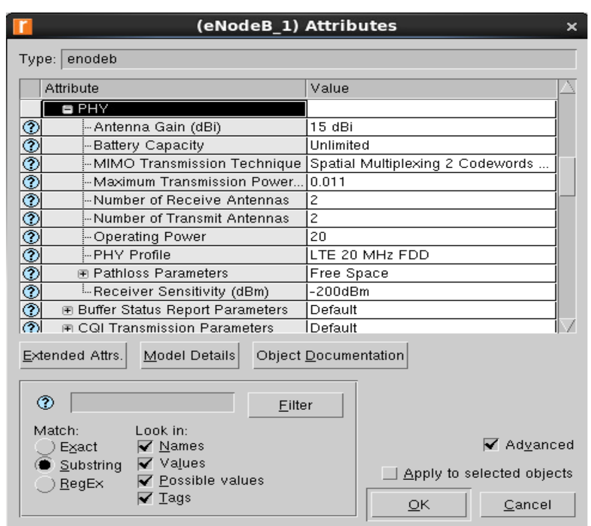

Figure 6 - eNodeB Configuration's Attributes

2 Class of service

As mentioned in section II, a service class captures the QoS requirements of service flows thus the traffic flows between the base station enodeB and the subscriber. For a given service class, the key parameters are class identifier, retention priority, min sustainable data rate for GBR- which is minimum guaranteed rate over the air- maximum sustainable data rate for NGBR, as well as the type of service.

The available bandwidth resources are allocated to GBR first, then to NGBR flows according to QCI level. Lastly, any remaining resources are then assigned to Best effort (BE) flows.

For this project, we have defined different class of service for mobile users according to the 3GPP Policy and charging control architecture [6]. Class of service definition is resumed in Table4.

Table 4 Class of service definition

EPS bearer Definition	QCI	Retention Priority	UL/DL Max bit rate voice	UL/DL bit Max rate video	Type of service
Platinum	1 (GBR)	2	1 Mbps	1.5/10 Mbps	Multimedia / Interactive voice
Gold	2 (GBR)	4		1.5/6 Mbps	Excellent effort
Silver	6 (NGBR)	6		1.5/5 Mbps	Background
Bronze	7(NGBR)	7		1.5/5 Mbps	Best effort

Subsequently, the LTE attribute definer was configured to map EPS bearer definition to the classes of service definition in Table 2.2. Moreover, each uplink service flow was configured with specific burst profile. Figure 7 shows LTE attribute definer configuration attributes.

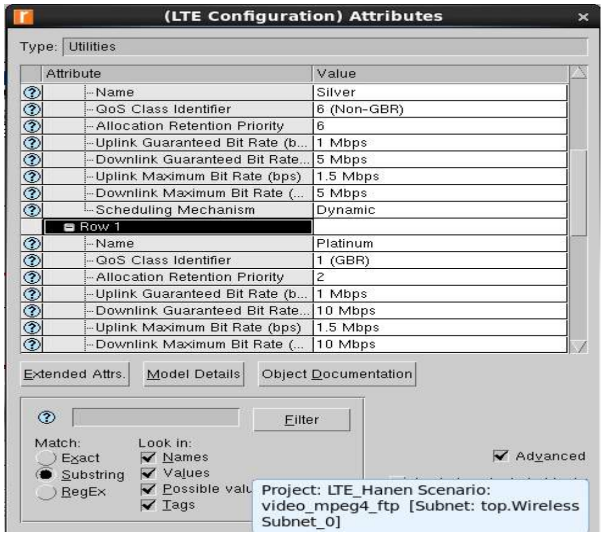

Figure 7 - LTE Service Class Configuration

LTE mobile users were configured to map the uplink and down link service flows to a specific type of service (ToS). Figure 8 shows one of the LTE mobile user (platinum class) configuration's attributes.

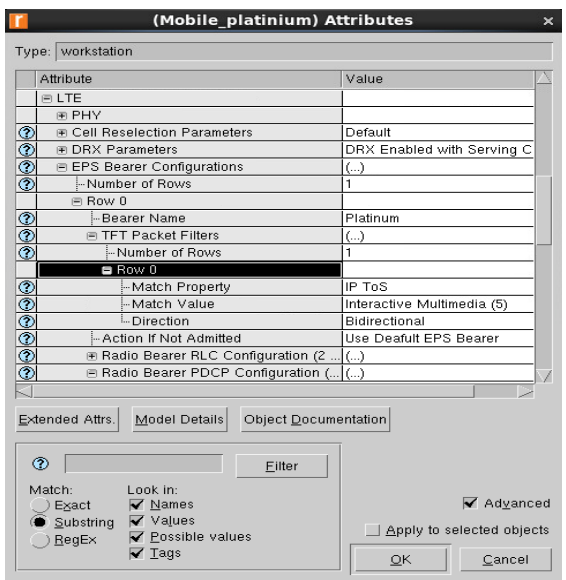

Figure 8 - LTE mobile users EPS bearer's configuration

VII. Traffic Configuration

Traffic is a key aspect of this project; we investigate the performance of LTE when delivering different services to the same mobile user in the same time. This will help us to observe the performance metrics from more realistic prospective comparing the real world since mobile users have a tendency, now a day, to exploit several services at the same time.

1 General Configuration

In this project, a mix of traffic has been considered while prioritizing one among the others in each scenario. Scenarios include the following traffic mix:

- VoIP and FTP: where different class of service are considered for voice while assigning best effort only to FTP service.
- Video and FTP: where different class of service are considered for video while assigning best effort only to FTP service.
- VoIP and video: we considered the best (platinum) and the worst (best effort) class of service for the voice and video.

According to [9] and [10] studies, G.711 and GSM EFR codecs guarantee the best performance in terms of MOS therefore we considered the G.711 in this project. Voice characteristic are listed in Table 5.

Table 5 Voice Characteristics

Parameters	Values
Voice codec	G.711
Transmission rate [kbit/s]	64
Sampling frequency [KHz]	8
Voice Service	PCM Quality Speech
Type of Service	Interactive voice

The video traffic source is a 2-hour MPEG-4 Matrix III movie trace [16] which characteristic are listed in Table 6.

Table 6 Video Characteristics

Parameters	Matrix III
Resolution	352x288
Codec	MPEG-4
Frame Rate (frames/sec)	25
Mean Rate (Mbps)	0.637
Frame Compression Ratio	47.682

2 Application Definition Attributes

Application definition attribute consists of several predefined applications that were customized as per user demands. In this project we used among those predefined applications the video, voice, and the FTP application. Figure 9 and 10 show the application definition attribute used in the simulation model for the video and FTP application.

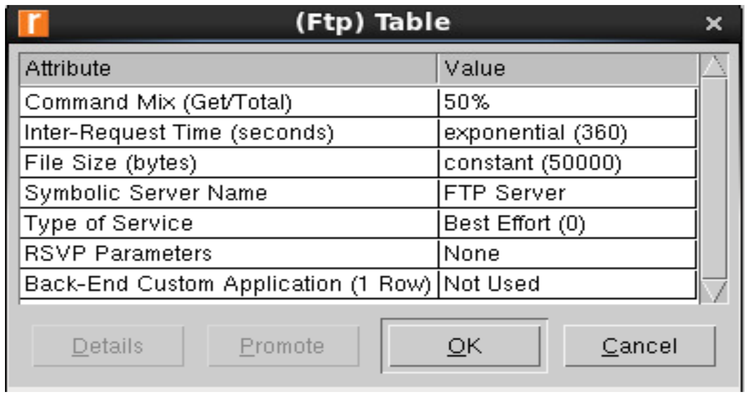

Figure 9 - FTP attribute configuration table

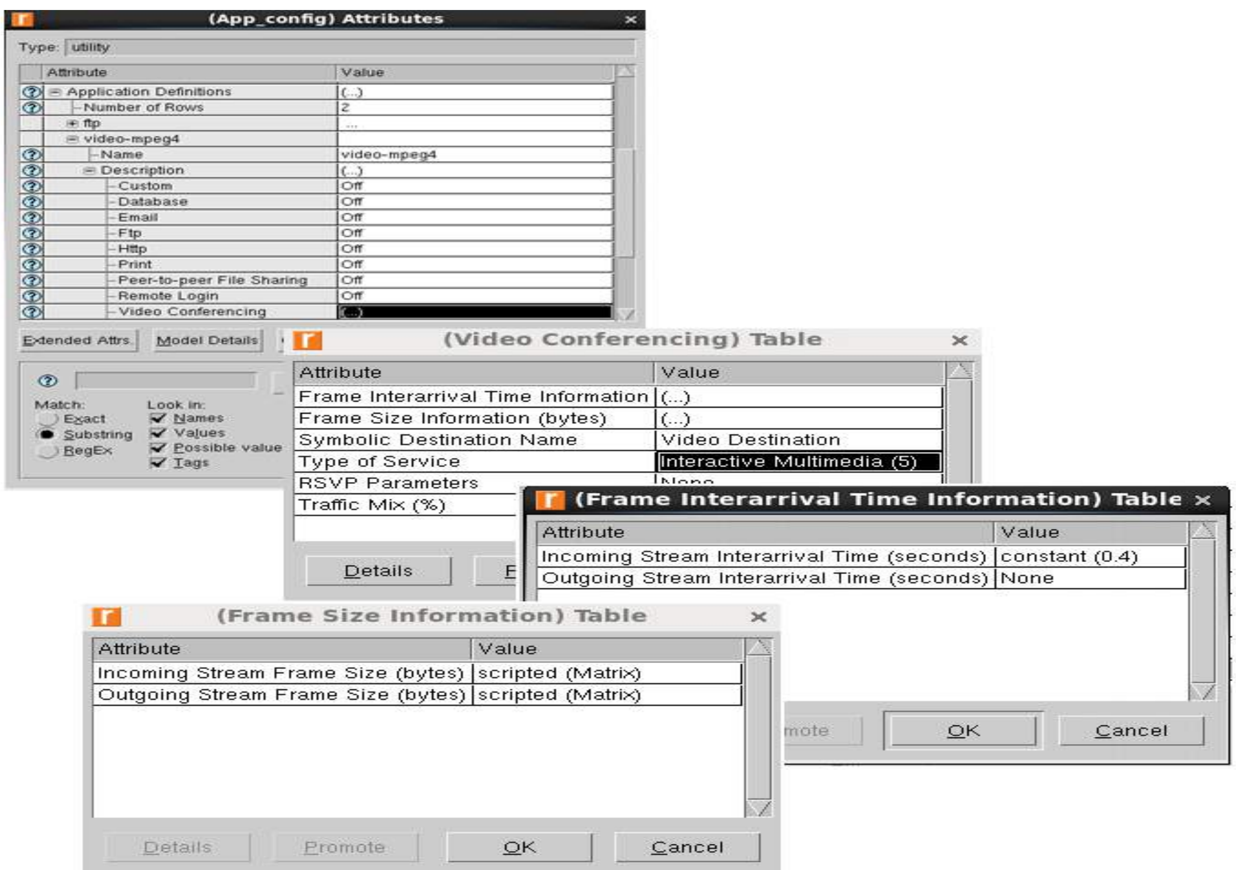

Figure 10 - Video conferencing attribute configuration tables

Video application was configured to map the video characteristics as defined in Table 6 in term of frame inter-arrival time and frame size with reference to the video file (Matrix).

FTP, application attribute is configured for heavy load traffic and is modeled for setup background traffic in the simulation.

3 Profile Definition Attributes

It is necessary to configure the profile definition. Figure 11 illustrates the profile definition attribute that is used in simulation.

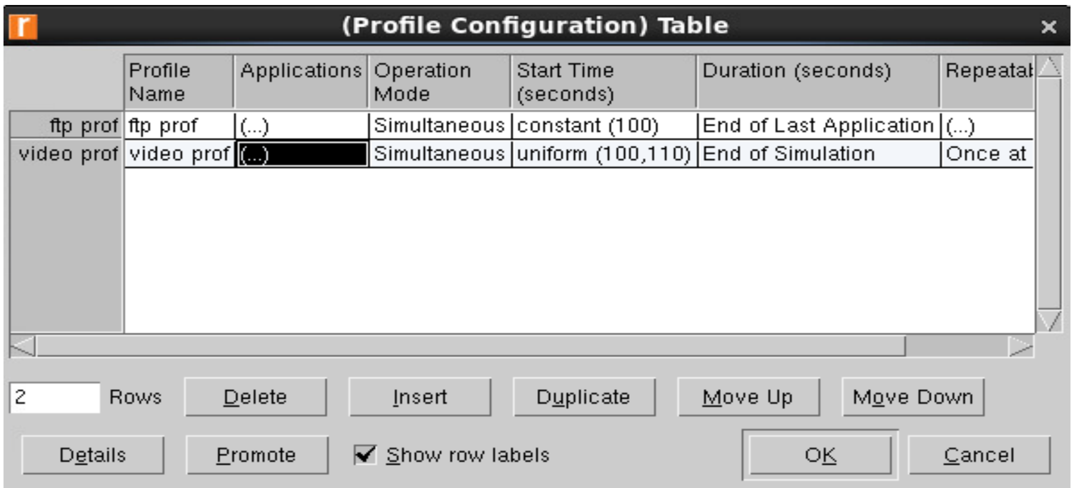

Figure 11 - Profile Definition Attributes

4 Configuration of video server in server subnet

An on demand server is used to configure both FTP and video. Figure 12 illustrates the profile definition attribute that is used in simulation.

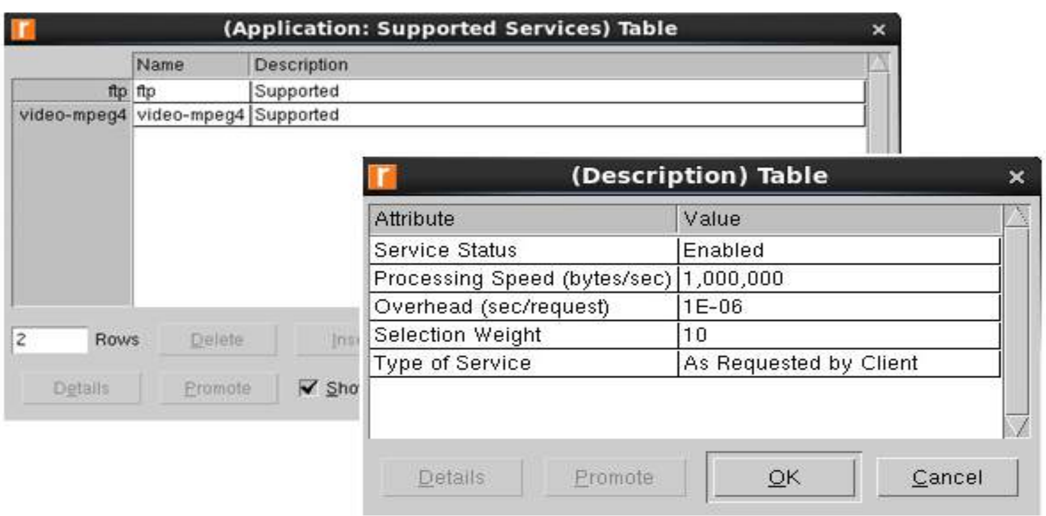

Figure 12 - Server Profile Definition Attributes

3. SIMULATION RESULTS

The aim of this project is to evaluate the performance of QoS in LTE network. Three scenarios have been considered where each on them defines a traffic mix and a prioritisation scheme. We will discuss in this section about End to End delay, packet delay variation and throughput performance metrics. All cells are working as a source and destination as well for voice. All mobile users stream video and FTP from the same on demand server.

1 Scenario 1- VoIP and FTP traffic mix

In this scenario, mobile users mimic a file download during a voice call of a real LTE user's behavior. Voice traffic is loss and delay sensitive conversely FTP traffic delay-insensitive intensive traffic. Different class of service are configured for voice service. A pair of mobile users, each in a different cell, with the same class of service is configured to communicate together. FTP traffic is configured for heavy load traffic and is assigned Best effort class of service. FTP traffic start around 3 min of simulation. Simulation run time is set to 15 min.

a. End-to-end delay

End-to-end delay is the time taken for a packet to be transmitted across a network from source to destination. Figure 13 displays the end to end delay for VoIP service for all defined EPS bearers and Figure 14 displays the end to end delay for 3 EPS bearers namely Gold, Platinum and Silver. According to ITU Telecommunication Standardization Sector (ITU-T) [17] values, if the end to end delay is less than 150 ms then it is good, it is acceptable even if it's less than 300 ms but it is considered poor if it is more than 300 ms.

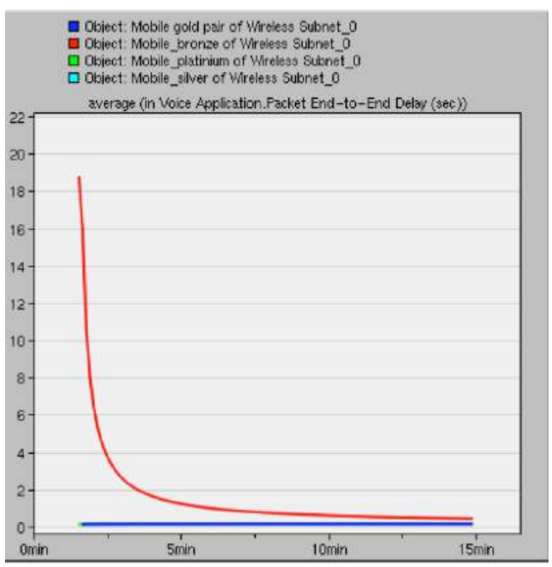

Figure 13 - End to End delay for all 4 EPS bearers

The mobile user with bronze class of service, i.e. best effort EPS bearer, presents quite higher end to end delay but it still in the acceptable range. All other defined EPS bearers present a good end to end delay as illustrates Figure 14.

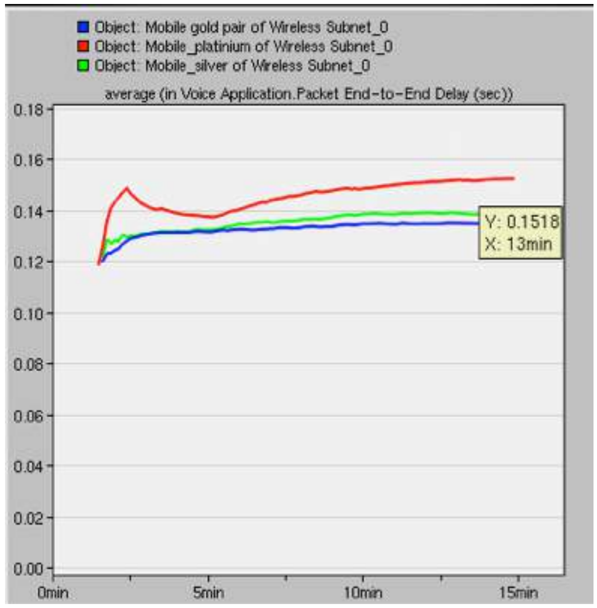

Figure 14 - End to End delay for 3 EPS bearers

b. Packet Delay Variation

A network with constant latency has no variation (or jitter).Packet delay Variation (PDV) is an important QoS factor in assessment of network performance. Jitter is basically the difference between end-to-end delay in selected packets in a flow and any lost packets being ignored. Figure 15 indicates the jitter for VoIP for all 4 EPS bearers and Figure 16 indicates jitter for 3 EPS bearers namely Gold, Platinum and Silver. According to [17] values, if jitter is under 20 ms then it's considered as good, but it is considered poor if it's more than 50 ms.

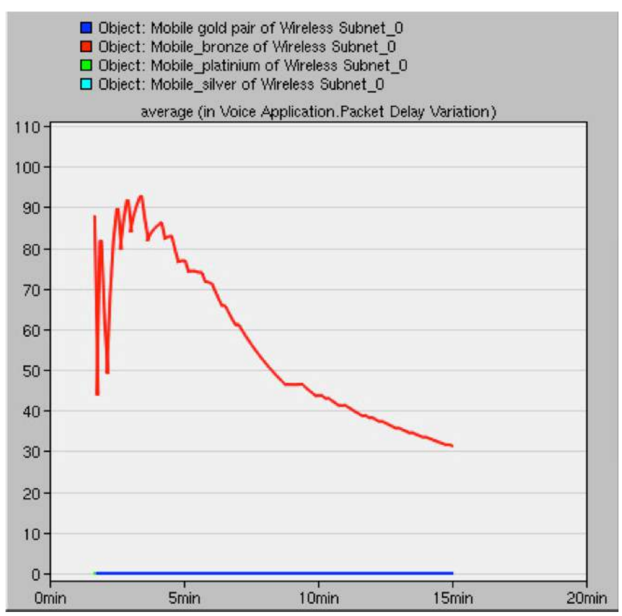

Figure 15 - Jitter for all 4 EPS bearers

The mobile user with bronze class of service, i.e. best effort EPS bearer, presents quite higher jitter, especially when FTP file download start, which is considered poor for VOIP service. All other defined EPS bearers present a jitter in the good range as illustrates Figure 16.

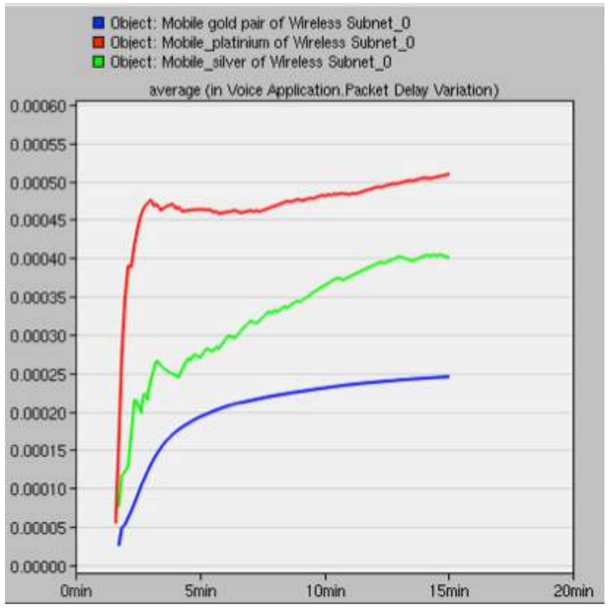

Figure 16 - Jitter for 3 EPS bearers

c. Throughput

Throughput is the average rate of successful message delivery over a communication channel. The throughput is usually measured in bits per second (bit/s or bps) and sometimes it is measured in data packets per second or data packets per time slot. Where, the minimum transmission rate is defined between 10 Kbps and 5Mbps.

Figure 17 describes throughput received by each EPS bearers in LTE as well as the throughput received for the FTP application. Here we can remark that even if higher priority is given to voice packets, FTP packets can still receive a part non negligible of the total throughput.

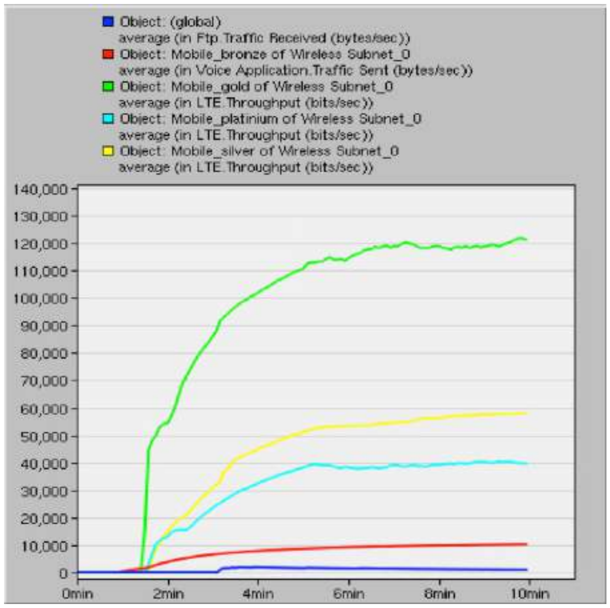

Figure 17 - Throughput in LTE

Figure 18 describes the throughput received by FTP traffic by mobile user in each class of service. We can remark that FTP service was able to receive a higher throughput in case of mobile user (Bronze) configured to best effort service for voice as well as FTP thing that is evident.

Consequently, prioritization of VolP over FTP typically does not cause large quality degradation of FTP services due to small VoIP packet sizes.

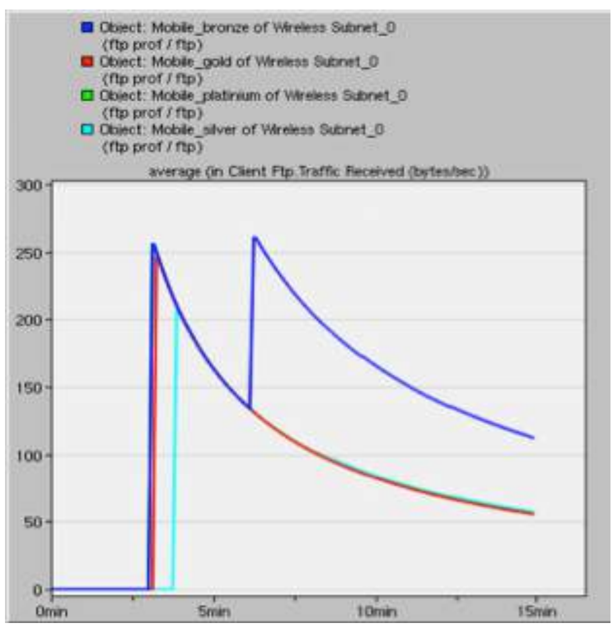

Figure 18 - Throughput for FTP

2 Scenario 2- Video and FTP traffic mix

In this scenario, mobile users mimic a file download during a movie video streaming of an LTE user's behavior. Video traffic is loss tolerant but delay sensitive conversely FTP traffic delay-insensitive intensive traffic. Different class of service are configured for video service. Mobile users, in a different cell stream the same video file from the on demand server. FTP traffic is configured for heavy load traffic and is assigned best effort class of service. FTP traffic start around 10 min of simulation. Simulation run time is set to 2 hours.

a. End-to-end delay

The values are averaged over the two hours movie duration. Figure 19 displays the end to end delay for Video for the different EPS bearers namely Gold, Platinum and Silver and Bronze. According to ITU Telecommunication Standardization Sector (ITU-T) [17] values, ideally end to end delay value should be below 10 ms and under 300 ms in average. In this scenario, end to end delay values for different EPS bearers are close to the ideal value even that mobile users are downloading two different files in the same time. The best values are recorded for the GBR bearer.

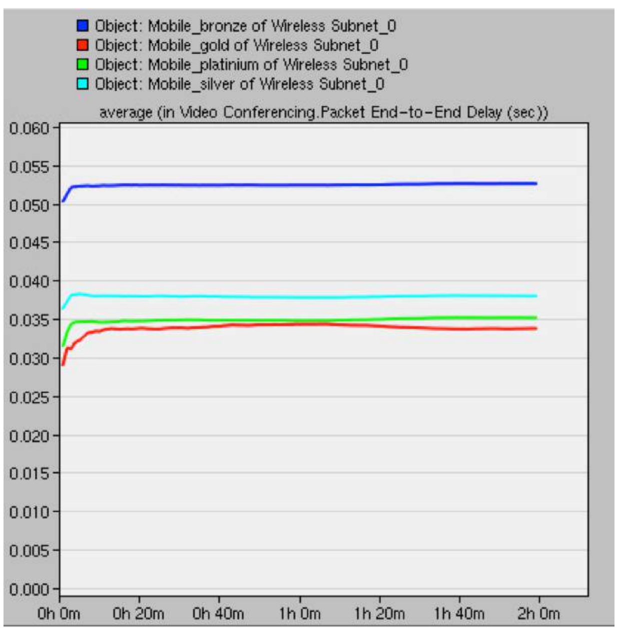

Figure 19 - End to End delay for Video

b. Packet Delay Variation

The values are averaged over the two hours movie duration. Figure 20 indicates the jitter for Video for the different EPS bearers. Ideally, jitter value should be below 20 ms and under 60 ms in average. In this scenario, al EPS bearers jitter values are in the ideal range. The best values are recorded for the GBR bearer.

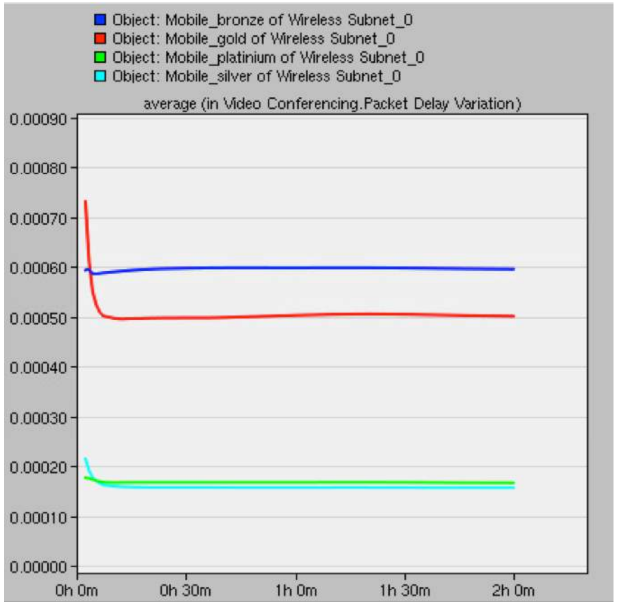

Figure 20 - Jitter for Video

c. Throughput

Figure 21 describes throughput received by each EPS bearers in LTE as well as the total throughput received for the FTP application. The values are averaged on two hours movie duration. The minimum transmission rate is acceptable for video if it range from 10 Kbps to 5Mbps and some Kbps for FTP. Here we can remark that even if higher priority is given to video packets, FTP packets can still receive a part of the total throughput.

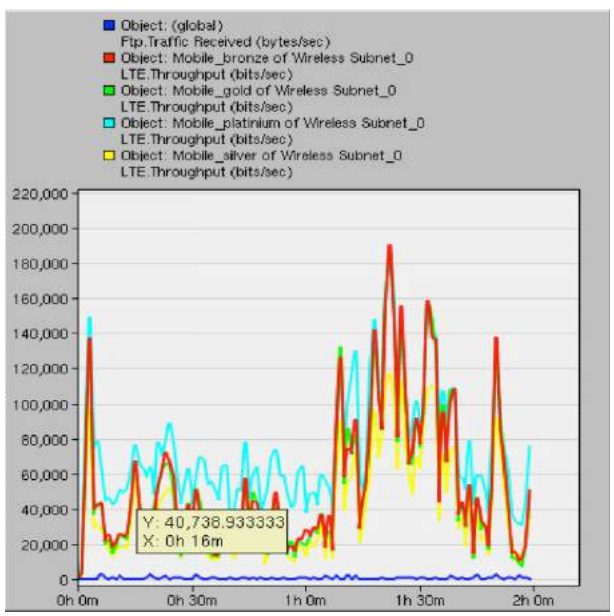

Figure 21 - Throughput in LTE

Figure 22 describes the throughput received by FTP traffic by mobile user in each class of service. We can remark that FTP service was able to receive a higher throughput in case of mobile user (Bronze) configured with best effort service for video as well as FTP thing that is evident and smaller one with the GBR class (platinum).

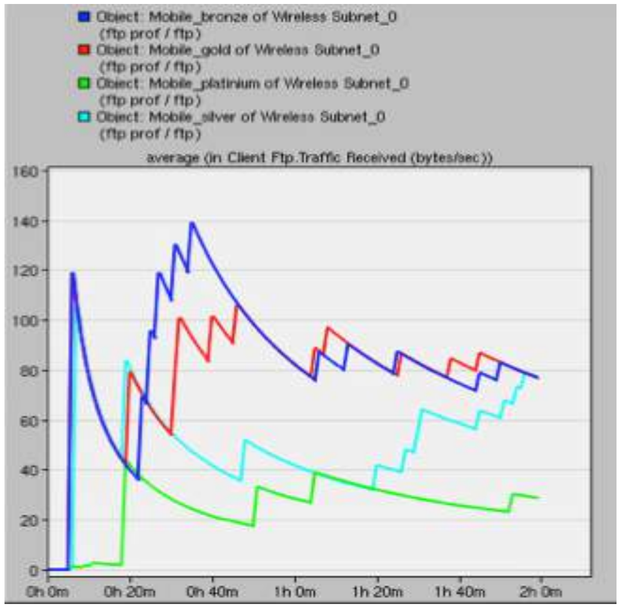

Figure 22 - Throughput in FTP

3 Scenario 3- VoIP and Video traffic mix

In this scenario, mobile users mimic a Skype call real LTE user's behavior where voice and video are streamed in the same time. Two class of service are configured for voice and video services: the GBR platinum class and the best effort class Bronze. A pair of mobile users, each in a different cell, with the same class of service is configured to communicate together. Video traffic starts around 10 min of simulation. We used the same video file simulated previously. Simulation run time is set to 30 min.

a. End-to-end delay

Figure 23 displays the end to end delay for VoIP and Video for Platinum and Bronze EPS bearers. Ideally end to end delay should be below 10 ms and the average should be less than 300 ms. In this scenario, end to end delay values are in acceptable range for voice and in ideal value for video.

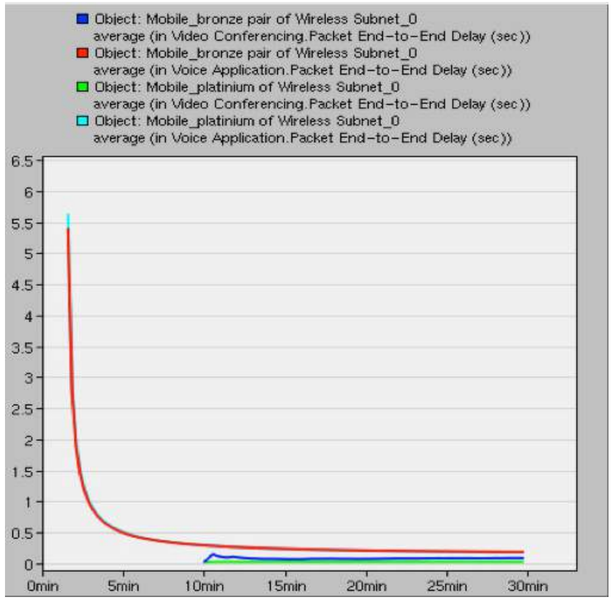

Figure 23 - End to End delay for Voice and Video

b. Packet Delay Variation

Figure 24 illustrates the jitter for Voice and Video for Platinum and Bronze EPS bearers. Ideally end to end delay should be below 20 ms and the average should be less than 60 ms. In this scenario, jitter values are in acceptable range for video for bronze bearer -best effort- and in ideal value for voice and video with GBR bearer. Consequently, prioritization of VolP over video typically does not cause large quality degradation of video services due to small VoIP packet sizes.

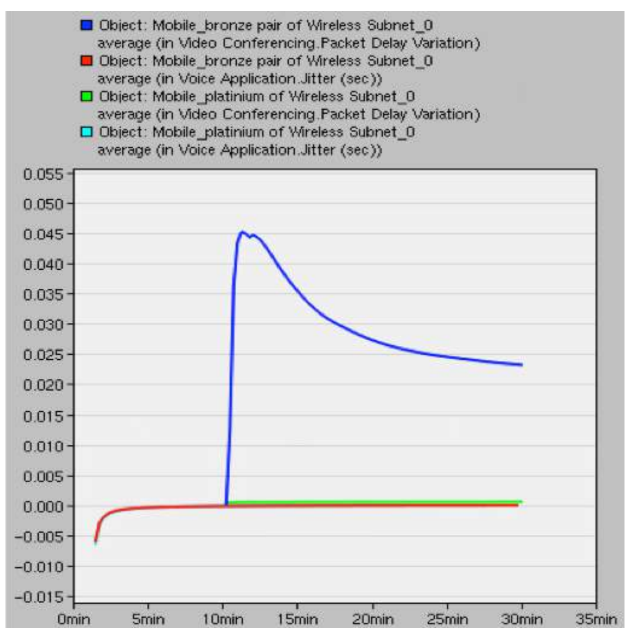

Figure 24 - Jitter for Voice and Video

4. CONCLUSION

In this research we have investigated the effect of QoS performance for Voice over IP and video conferencing in the LTE network with End to end delay, packet delay variation and throughput metrics. Specifically we investigate the impact of differentiation and prioritization of delay-critical traffic like VoIP over other delay-insensitive intensive traffic like FTP file download in purpose to study the impact of transmitting various services in the same time from the same user equipment. Three networks scenarios have been created with three kinds of traffic mixes.

Riverbed Modeler 18.5 has been used to simulate the network scenarios. We used the values averaged on two hours movie duration. The simulation result shows that GBR and Non GBR bearers have impact on voice under congested network. In all cases we can conclude, that VoIP and video applications perform well in most case while achieving different levels of performance by employing differentiation. Highest priority GBR bearers are getting more opportunity to use available resources compared to NGBR.

The prioritization of VoIP over FTP and video does not cause large quality degradation of other services due to small VoIP packet sizes. We noticed that video responds better than VoIP application in a LTE network.

In future, there are scopes for studies focusing on other QoS metrics for GBR and NGBR bearers' behavior with mobility.

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