ENSC 427: Communication Networks
Spring 2014

THE PERFORMANCE ANALYSIS OF LTE NETWORK

http://www.sfu.ca/~jla235/427Project.html

Team 06
Guo'Enzo   301126666   shuohuag@sfu.ca
Lin'James   301126878   jla235@sfu.ca
Zhang' Yang   301128360   yza121@sfu.ca

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Acronyms

3GPP	**3rd Generation Partnership Project**
EPC	Evolved Packet Core
GSM	Global System for Mobile
HTTP	Hypertext Transfer Protocol
IP	Internet Protocol
ITU	International Telecommunication Union
LTE	Long Term Evolution
MOS	Mean opinion score
VoIP	Voice over Internet Protocol
QoS	Quality of Service
UMTS	Universal Mobile Telecommunication System

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Abstract

Long Term Evolution (LTE) is a standard for wireless data communications technology. LTE is the evolution of GSM/UTMS network technologies with increased capacity and speed using packet switched radio interface. It is widely used by many countries as the latest high speed mobile broadband technology; however, the main stream telephony services of the world, such as GSM, UMTS and CDMA2000, are circuit switched radio interface, so in order to adopt the LTE standard, the mobile voice services will shift from circuit switched to packet switched mode using the VoIP services.

This paper focuses on the quality of service (QoS) of Voice over IP (VoIP) and Web Browsing on the LTE network, which includes the analysis of End to End Delay, LTE delay, Mean Opinion Score (MOS) and Throughput. Moreover, this paper provides a detailed description of simulation models for network topology and elements using OPNET 16.0.

1. Introduction

Nowadays, people's life is closely related to mobile devices and the demand for high speed reliable mobile becomes increasingly strong. The 4th Generation (4G) – LTE is developed by 3rd Generation Partnership Project (3GPP) as the evolution of GSM/UTMS standards. Compare to 2nd Generation (2G) and 3rd Generation (3G) Networks; LTE has a significantly increased data rate: the designed maximum peak data rate for Downlink (DL) is 100 Mbps and peak data rate for Uplink (UL) is 50 Mbps. LTE also has the increased spectrum flexibility, the supported bandwidth is optimized from 1.4 MHz to 20 MHz. The core network of LTE is purely designed and optimized for packet switched radio interface, which means LTE network is not compatible with circuit switched network, such as GSM and UMTS.

Voice is the basic communication service which is well implemented in the circuit switched networks, by introduction of LTE network, voice service is shifted to packed switched mode from circuit switched mode and VoIP is applied to the network. VoIP convert the voice signal to digital packet and transfer the voice data via packed switched system. The biggest advantage of VoIP is that the cost of voice service is significantly decreased and this is a flexible voice solution for users; however, the QoS of VoIP depends on the network condition. If the network connection is slow, the voice quality will be poor and not behave in timely manner. Thus, we want to analysis the performance of VoIP on LTE network.

Moreover, the biggest advantage of LTE network is that LTE has the highest data rate. Mobile users can benefit from the high data rate and enjoy the mobile web browsing experience. We want to discover the performance of web browsing on LTE network.

In order to analyze the performance of VoIP and Web browsing on LTE network, we focus and analyze the following parameters:
- Jitter
- End-to-end delay
- LTE delay
- Throughput
- Mean Opinion Score (MOS)

With the simulation results, we will determine the performance of LTE network and discover the factors which can affect the performance.

2. Background

In this section, the background information of LTE network architecture will be demonstrated.

2.1 Overview of LTE Network Architecture

Figure 1 shows the architecture of LTE network. The LTE network architecture is called Evolved Packet System (EPS). EPS is an IP based architecture and provides IP users connectivity to packet data network. From the high level view, EPS contains 2 major parts: Evolved Universal Terrestrial Radio Access Network (E-UTRAN) and Evolved Packet Core (EPC).
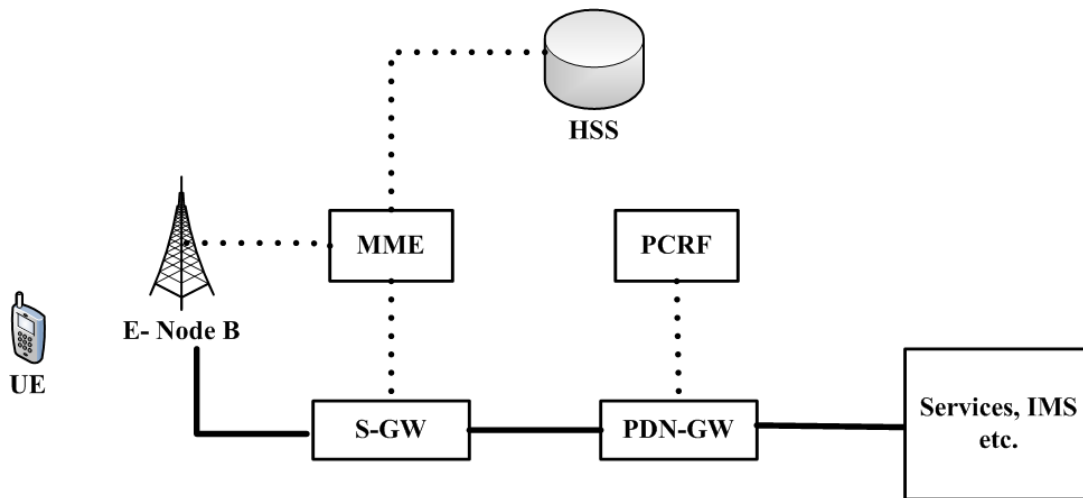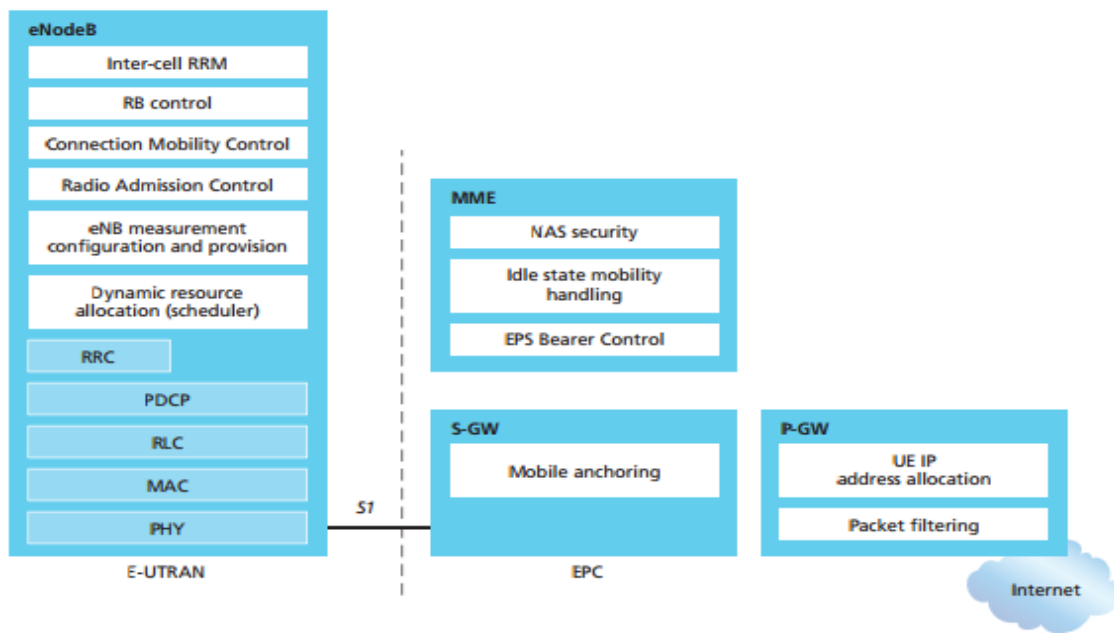

Figure 1 LTE Network Architecture

Figure 2 Functions of E-UTRAN and EPC

2.2 E-UTRAN

E-UTRAN only contains one node, the evolved NodeB (E-NodeB), which is connected to the user equipment (UE). E_UTRAN is connected to EPC via S1 layer. The E-NodeB is responsible in scheduling and optimization of user radio resource. Due to the self-control functionality, E-NodeB allows low response time.

2.3 EPC

EPC contains 5 logic nodes: Mobility Management Entity (MME), Home Subscriber Server (HSS), Proxy and Charging Rules Function (PCRF), Packet Data Network Gateway (PDN-GW) and Serving Gateway (S-GW). The detailed information of these nodes will be explained in the following sections.

2.3.1 MME

MME controls the plane signaling between user and EPC. The Non Access Stratum (NAS) protocol is running between UE and EPC.

2.3.2 HSS

HSS is a database containing all the related information of user and subscriber. It is responsible for providing support functions in mobility management, call setup, user authentication and access authorization.

2.3.3 PCRF

The main responsibility of the PCRF is performing the Policy and Charging Control (PCC) functions that are used to control the QoS configuration and tariff making of each individual user. PCRF also gives the specified tariff and QoS policies of each UE to the PDN-GW and the S-GW.

2.3.4 PDN-GW

The PDN-GW acts as an interface point of interconnect between EPC and external packet data networks. UE is allowed to have simultaneous multi-connectivity with different PDN-GW. The PDN-GW is responsible for establishing, maintaining and deleting the GPRS Tunneling Protocol (GTP) tunnels to S-GW. It is also responsible for performing IP address allocation, policy control, packet filtering for each user, charging, lawful Interception and packet screening.

2.3.5 S-GW

The SGW is responsible for routing and forwarding the data packets between users and E-NodeB. It produces a tunnel to transmit data traffic between the UE and external network. It is also responsible for acting as the mobility anchor for the user plane between inter- E-NodeB handover and as the anchor moves between LTE and other 3GPP technologies.

3. Main Parameters

In order to analyze the QoS of VoIP and web browsing on LTE network, we will focus on the follow parameters when analyzing the simulation results.

3.1 Jitter

In packet switched networks, packets are received in continuous streams and jitter is defined as the variation in time between each receiving. Jitter is the result of congesting in the IP network, which can occur at receiver side or carrier network. Jitter is a very important parameter in voice streaming because jitter can lead to poor voice quality and the level of jitter should be minimized. The level of jitter can be reduced by adding anti-jitter circuits, jitter buffers, dejitterizer, and filtering. According to International Telecommunication Union (ITU) standard, the average jitter should be less than 60ms and the ideal jitter should be less than 20ms.

3.2 End-to-End Delay

End-to-end delay is defined as the time for a packet to be transmitted from the source to the destination. The end-to-end delay is related to encoding/decoding delay, transmission delay, propagation delay, processing delay and queen delay. The end-to-end delay is an important parameter for real-time transmission because we want the voice stream is transmitted in the timely manner. According to ITU standard, the average end-to-end delay should be less than 150ms and the ideal end-to-end delay should be less than 50ms.

3.3 LTE Delay

LTE delay is the round-trip delay, which is the time from a packet is send to the acknowledgement is received. This delay is related to the distance between the user and base station, number of users and the applications (VoIP/web browsing).

3.4 Throughput

Throughput is the data rate that is successfully delivered over a channel. The unit for throughput is usually bits per second. By comparing the throughput of each scenario, it is easy to determine the QoS of each scenario.

3.5 MOS

MOS is a number value as an indication of the quality of received voice after codecs transmitted and compressed. It is s the arithmetic mean of all the individual scores, and can range from 1 (worst) to 5 (best). Furthermore, the score is affected by a number of factors such as jitter, end-to-end delay and packet loss. Table 1 shows the ITU standard for MOS.

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

Table 1 ITU Standard for MOS

4. OPNET Implementation

4.1 Overview of LTE Model in OPNET

The OPNET contains a large capacity models library that supports various protocols like TCP, UDP, SIP and it is capable of simulating applications like voice, FTP, web browsing etc. Moreover, the OPNET model has a hierarchical environment which is composted of the network model, node model and process model. All the three models need to be configured to perform the simulation. The LTE network model in OPNET is consisted of mobile nodes, an E-Node B and an EPC.

4.2 OPNET Simulation Topology

In this project, it contains two test cases for analyzing the performance of LTE network. First network, we simulated the Voice over IP (VoIP) in different distance and compared their result. Second network, we simulated the web browsing in different distance and number of IP user to analyze their result.

4.3 Voice over IP (VoIP) on LTE Configuration

In the VoIP configuration, we design two scenarios in different distance and we set eNodeB is equidistant from the two mobile users in both scenarios: one setting mobile users and eNodeB are 500 meters and one setting mobile users and eNodeB are 1000 meters.
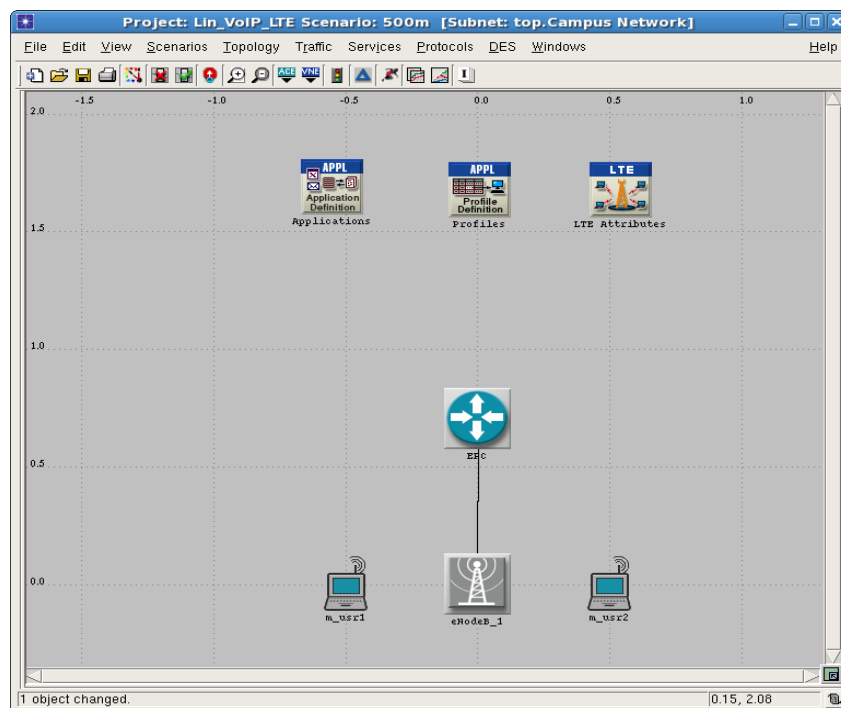

Figure 3 OPNET VoIP over LTE Design (500m)

Figure3 is the topology of our first scenario. We used the OPNET Application Definition attribute to set up the VoIP model and the VoIP Configuration parameter show in Figure 3

below. The VoIP application uses G.711 encoder scheme and Interactive Voice (6) as the type of service for creating the VoIP calls. After configuring the application, we was going to configure the Profile definition by using Profile Definition attribute. It set the start time of the simulation to 100 (off-set "60"+start time "40") seconds and the VoIP application is repeated continuously until the end of the simulation. It implied that VoIP calls will be established between source and destination starting at 100 seconds and the calls are added continuously until the end of simulation. Then, we chose 20MHz bandwidth in the e-NodeB. Next, we created second scenario by changing the distance between e-NodeB and mobile users.

Attribute	Value
Silence Length (seconds)	default
Talk Spurt Length (seconds)	default
Symbolic Destination Name	Voice Destination
Encoder Scheme	G.711
Voice Frames per Packet	1
Type of Service	Interactive Voice (6)
RSVP Parameters	None
Traffic Mix (%)	All Discrete
Signaling	None
Compression Delay (seconds)	0.02
Decompression Delay (seconds)	0.02
Conversation Environment	(...)

Figure 4 VoIP Configuration Parameters

4.4 Web Browsing on LTE Configuration

The following figure is one of the topologies we implemented for the web browsing. In this topology we are going to verify how the distance between e-NodeB and mobile station affect the performance of the web browsing. We set the voice service as the application. The figure below shows 3 different scenarios. Mobile users 1, 2, 3, 4 and e-NodeB are kept at the same position and the distance between mobile user 5 and e-NodeB is changed as 500m, 1km and 1.5km. We are going to analyze the effect of distance through the parameters: LTE delay and throughput.
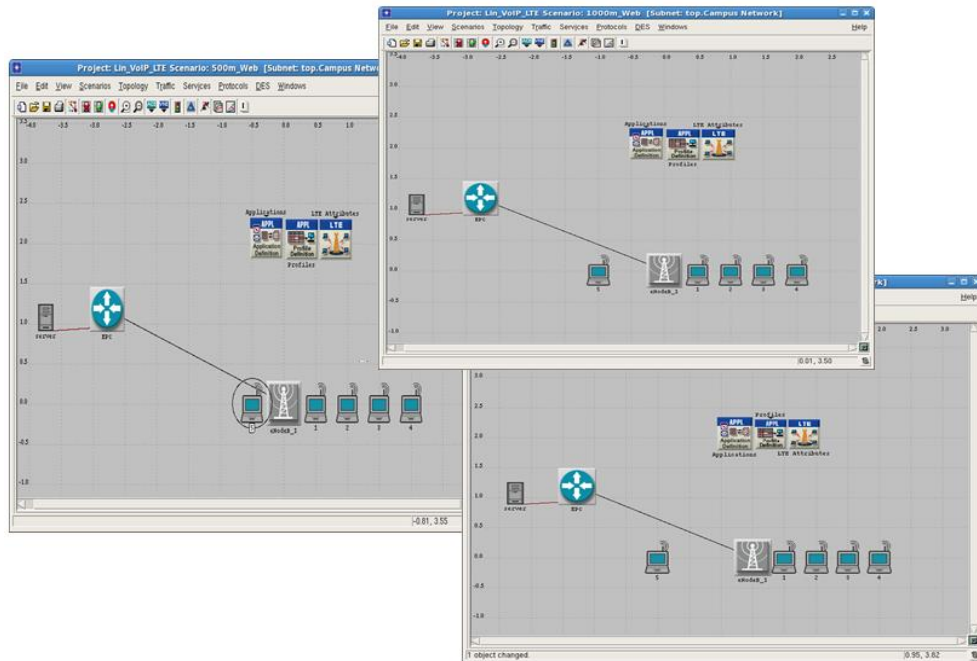
Figure 5 Scenarios for different distances

The following figure is the topologies we implemented to test the effect of number of mobile users in the same LTE network for the web browsing. We have the HTTP web browsing as the application and add the HTTP server to the LTE network. The topology contains shows 3 different scenarios. We kept 4 mobile users in the first scenario, 8 mobile users in the second scenario, and 16 mobile users in the third scenario. We are going to analyze the effect of number of mobile users through the parameters: LTE delay and throughput.
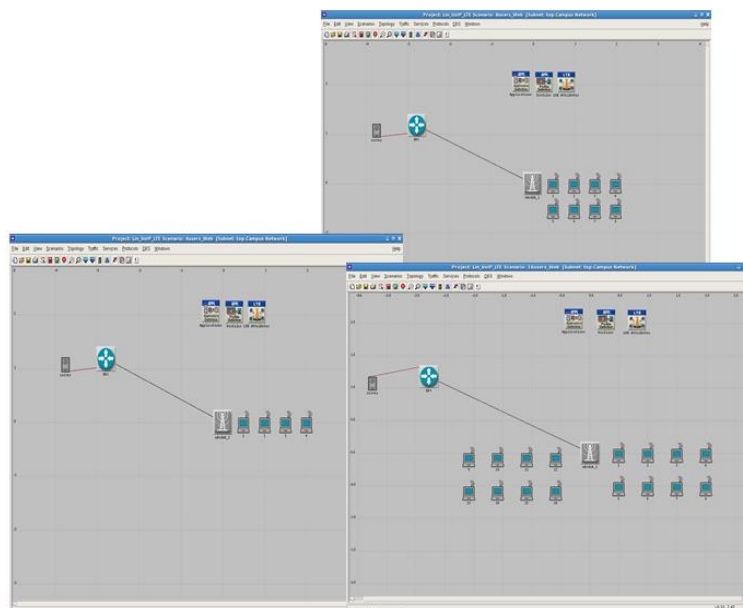
Figure 6 Scenarios for different number of users
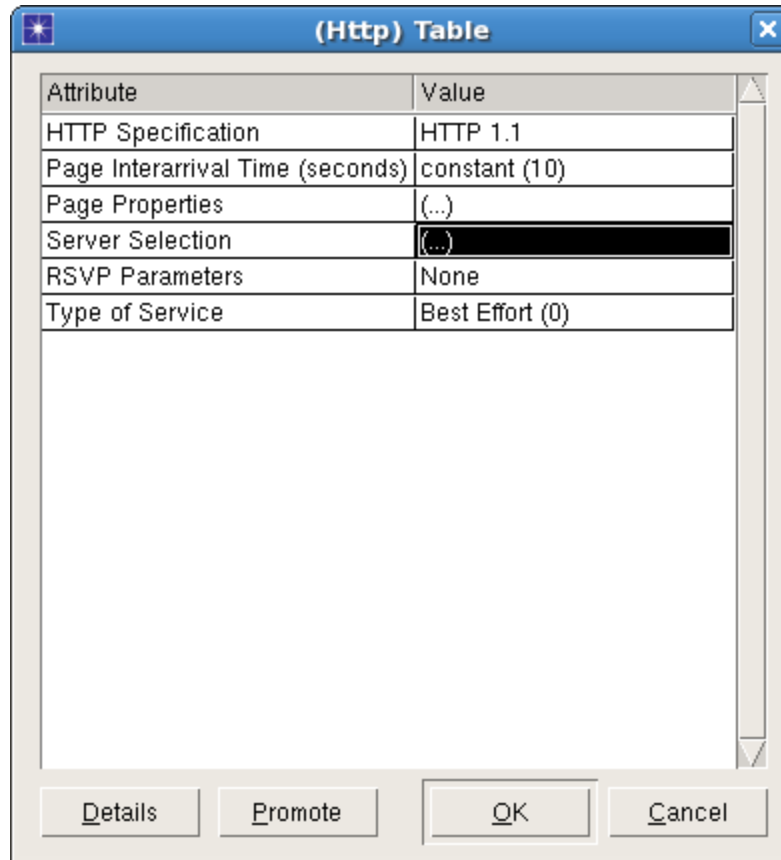
Figure 7 Web http configuration parameters

The above figure is the configuration parameter we set in the web http application. HTTP 1.1 is set as the HTTP Specification. Page Intertribal Time is set to be 10 seconds constantly. The page is set to be the combination of constant 1000 and medium image. As the figure 8 shows below:
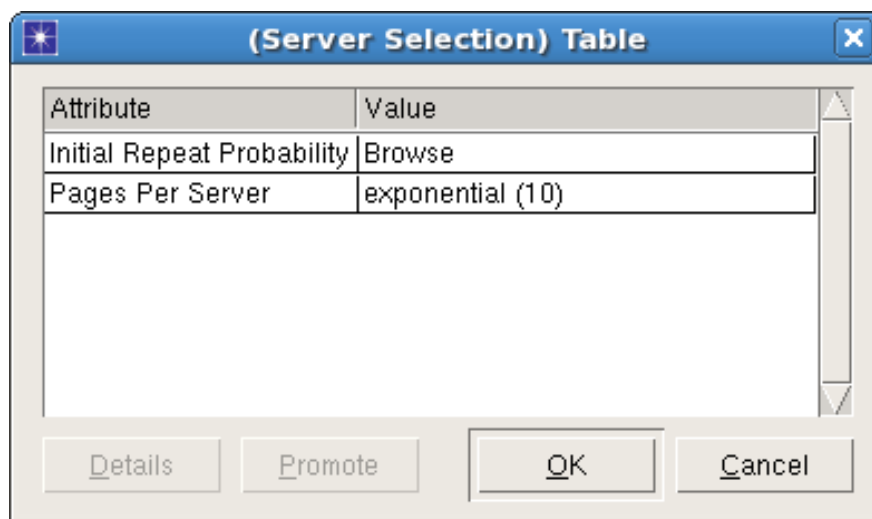
Figure 8 Web server page setup
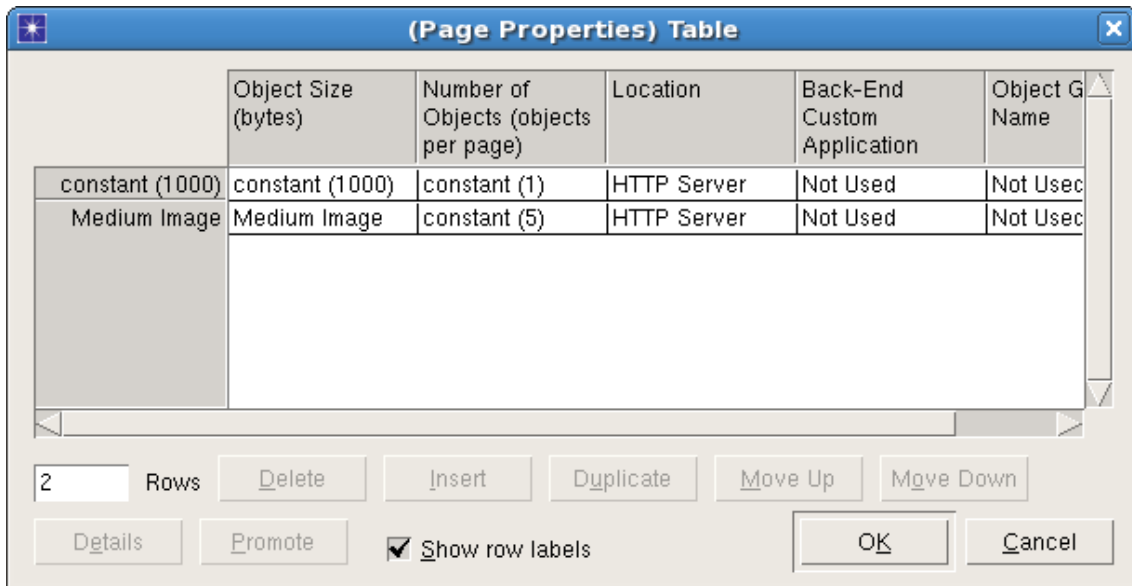
Figure 9 Page size setup

5. Simulation Results

5.1 VoIP Results

Jitter

The following figure shows the jitter result of VoIP. Due to the technical difficulty, we was not able to get the jitter result for 1000m scenario. At the beginning of simulation, the initial jitter was 33ms, which is much less than the ITU standard average jitter (60ms); after the voice call is stabilized, the jitter is less than 20ms, which is in the range of ITU ideal jitter range. Based on the 500m scenario result, we can conclude that the jitter performance of VoIP on LTE network is excellent.
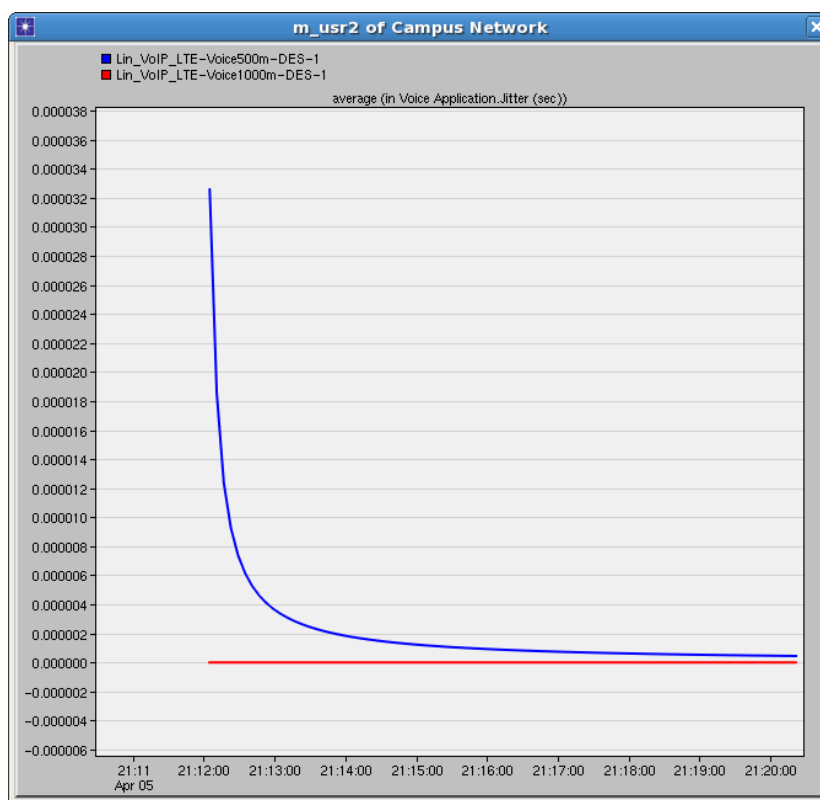
Figure 10 Jitter result for VoIP

End to End Delay

For the end-to-end delay, it is obvious that 500m scenario has better performance than the 1000m scenario. The average end-to-end delay for 500m scenario is about 81ms and the average end-to-end delay for 1000m scenario is around 97ms. Compare to the ITU standard, the end-to-end delays for both scenarios are below the average rate, which means the end-to-end delay performance of VoIP on LTE network meets the ITU requirement. In addition, the end-to-end delay is increased while the distance is increased.
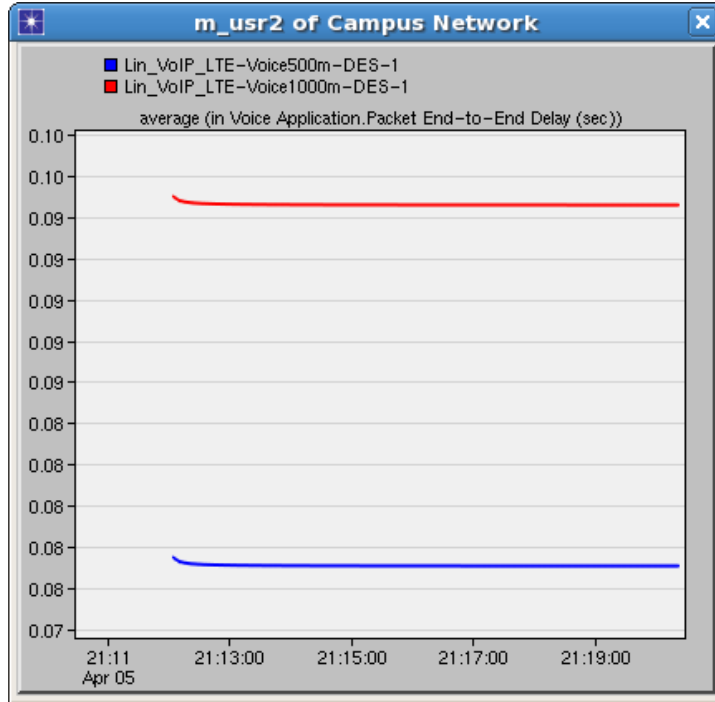
Figure 11 End-to-end delay result for VoIP

MOS

As figure 12 shows, the MOS for 500m scenario and 1000m scenario is 3.59 and 3.48 respectively. Based on the ITU standard, the voice quality is in the range fair to good. It is obvious that MOS is related to the distance of user, shorter distance can lead to better MOS.
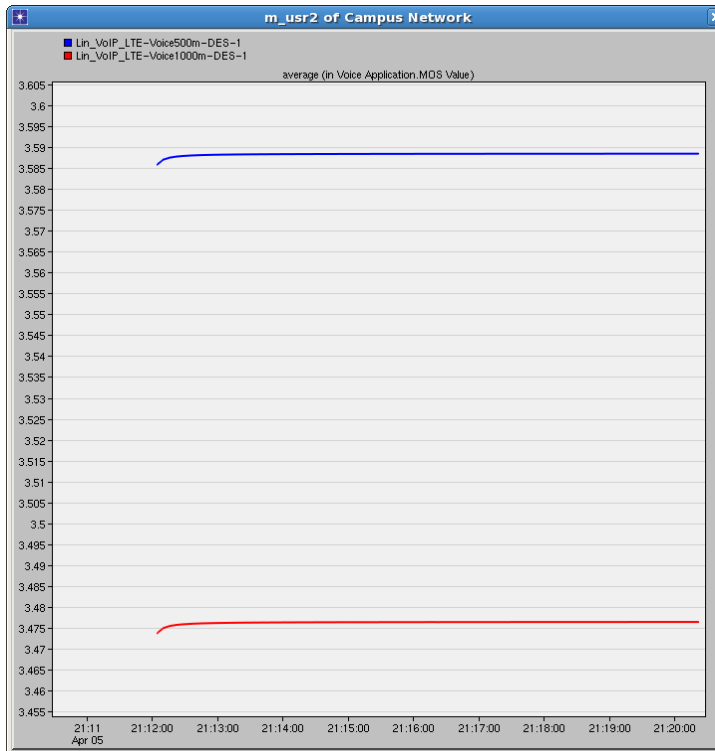

Figure 12 MOS result for VoIP

5.2 Web Browsing Result

LTE Delay of Various Distances

In the figure below, the blue line shows the LTE delay of the 500 meter scenario, the red line shows the result for 1000m scenario and green line shows result of 1500m scenario. It is obvious that the shorter distance between users and eNodeB can lead to the shorter LTE Delay. At the beginning of simulation, the 500m scenario has the best initial delay of the 3 scenarios. Then the LTE delay of these 3 scenarios starts to decrease. The final average LTE delay for 500m scenario is about 1.55ms and the final average LTE delay for 1500m scenario is around 1.57ms. When these 3 scenarios reach the stable, the delay of the 500m is still the lowest one.
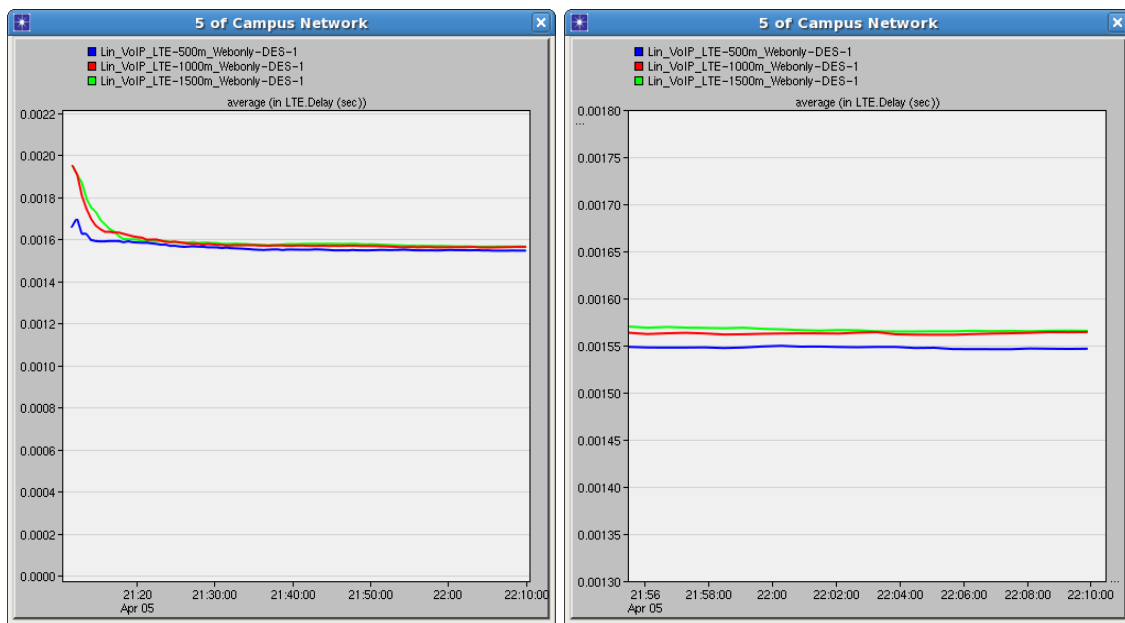
Figure 13 Delay of various distances

Throughput of various distances

Based on the maximum throughput value in figure 14, the maximum throughput for 500 meters, 1000 meters and 1500 meters are 3100 bits/sec, 3050 bits/sec and 3000 bits/sec, respectively. It is obvious that the shorter distance between users and eNodeB can lead to larger maximum throughput. This is what we expected.
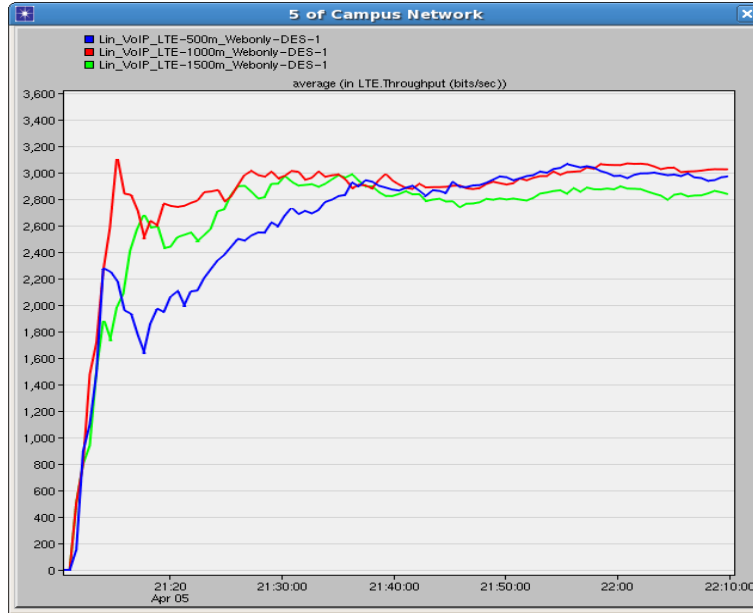
Figure 14 Throughput of various distances

LTE Delay of various number of IP user

In the figure below, the blue line shows the LTE delay of the scenario which contains 4 IP users while red and green lines show other two scenarios which contain 8 IP users and 16 IP users, respectively. It is obvious that increasing the number of IP user can increase the LTE Delay value. The average LTE delay for 4-user scenario, 8-user scenario and 16-user scenario are 1.75ms, 1.78ms and 1.79ms, respectively.
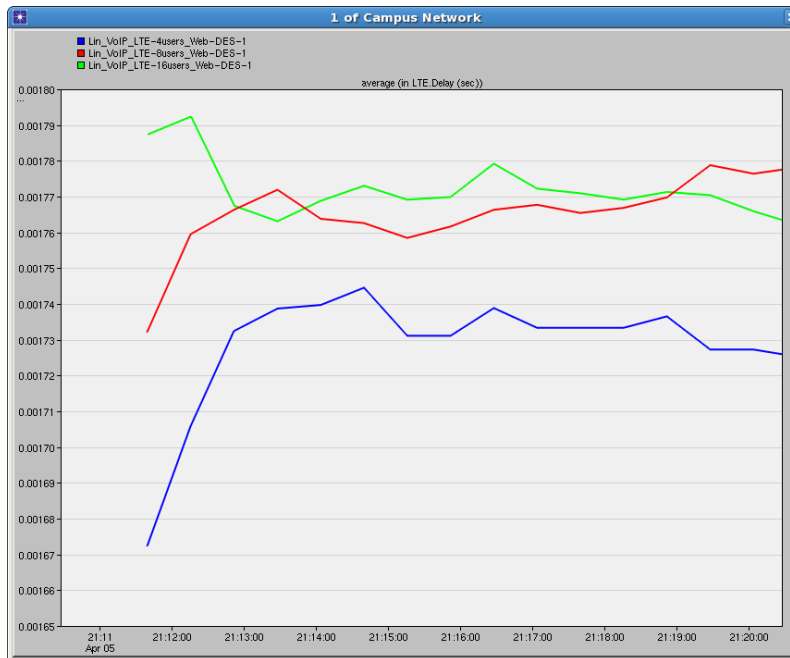
Figure 15 LTE Delay of Multi-user

Throughput of various number of IP user

The figure for multi-user throughput is shown below, it is obvious that the green line, which is the throughput of 16-user scenario, has the largest maximum throughput value and 4-user scenario has the smallest maximum throughput value. The maximum throughput for 4-user scenario, 8-user scenario and 16-user scenario are 2200 bits/sec, 2300 bits/sec and 2800 bits/sec, respectively.
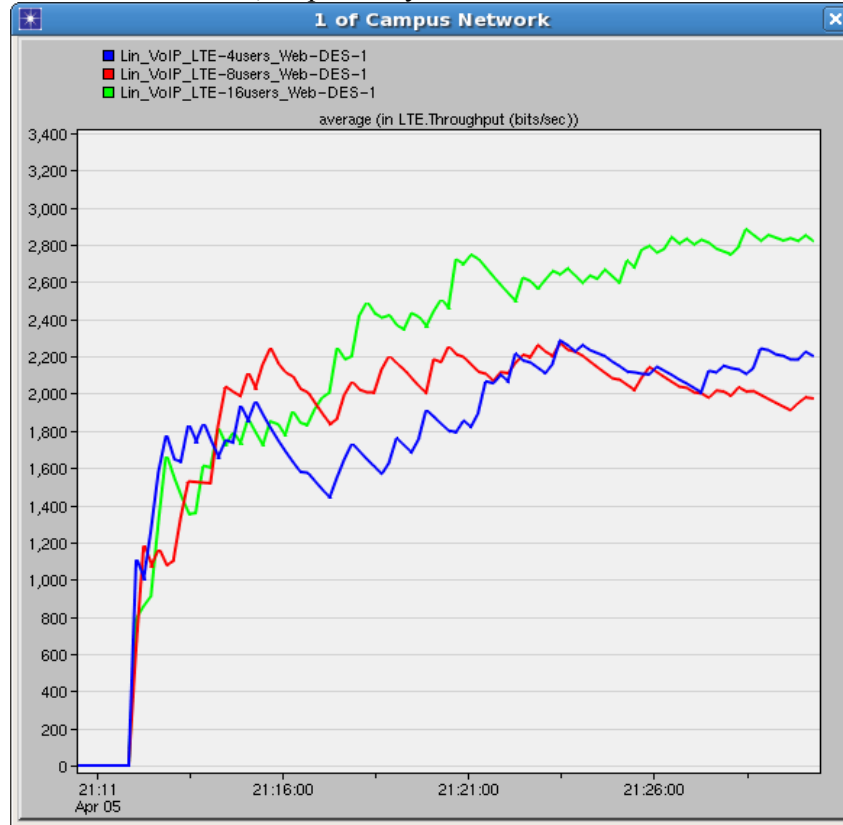
Figure 16 Throughput of Multi-user

6. Conclusion

For voice over IP on LTE, the results of OPNET simulation agree with theory. There is increment in End-to-End delay when the distance between users and eNodeB is increased. Moreover, the MOS value is decreased as we the increase the distance between users and eNodeB. In other word, when the distance between the users and eNodeB decreased, the quality of VoIP decreased.

For Web Browsing on LTE, it is obvious that the increment in the distance between users and eNodeB and the increment in number of IP users will increase the LTE Delay value and maximum throughput. Therefore, we can conclude that in the same network, the fewer users or the closer the user beside the eNodeB, the better internet browsing performance.

7. Future Work

For this project, we analyzed the performance of VoIP and Web Browsing on one single cell LTE network. In order to make the simulation more reliable, we can simulate the VoIP and Web Browsing data transmit on multi-cell LTE network in future. Furthermore, we can also simulate VoIP and Browsing work in different network, such as WIFI and WIMAX, and then compare the simulation results with what we did on LTE network. In addition, we only modulated VoIP or Web Browsing traffic model to analyze the performance of LTE network. We can combine various applications together on LTE network and analyze the effect to the performance.

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Appendix

LTE network parameters setup of VoIP

HTTP server parameters setup: