ISSN:2278 – 909X
International Journal of Advanced Research in Electronics and Communication Engineering (IJARECE)
Volume 7, Issue 4, April 2018

Simulation & Performance Analysisof LTE Network: Femtocell Perspective

Kausar Ali[1], Dr. R. P. Gupta[2], Dr. Y. C. Sharma[3]
[1]*PhD Research Scholar, ECE, Vivekananda Global University, Jaipur, India*
[2]*Ret. Scientist "G", CEERI Pilani, India*
[3]*Dean R & D, Vivekananda Global University, Jaipur, India*

***Abstract*— since the arrival of mobile phones the ever growing field is rapidly shadowing many older communication technologies. It also has led to the development of numerous latest technologies, LTE femtocell being one.The paper deals with the simulation and performance analysis of LTE femtocell network. The simulation is done in OPNET modeler. The evaluation of the network is done by comparing various parameters that decides the performance of the LTE femtocell network.**

***Index Terms*-LTE, OPNET, Jitter, LTE Delay, End-to-End Delay, MOS, Throughput**

I INTRODUCTION

These days, mobile devices have a great impact on people's life's and the demand for high speed reliable mobile connection has become progressively high. The Fourth Generation (4G) – LTE is developed by Third Generation Partnership Project (3GPP) as the development of GSM/UTMS standards. LTE has a significantly increased data rate when compared to Second Generation (2G) and Third Generation (3G) Networks. In LTE network the expected peak data rate for Uplink (UL) is 50 Mbps and peak data rate for Downlink (DL) is 100 Mbps. LTE also has other significant advantages such as the better spectrum flexibility, the supported bandwidth is improved 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 e.g. GSM and UMTS.

II ANALYSIS

The analysis work focuses on the Web Browsing and theQoS (quality of service) of VoIP (Voice over IP) on the LTE network. The work alsoconsiders the investigation of End to End Delay, LTE delay, Mean Opinion Score (MOS) and Throughput. Additionally, the workdelivers a thoroughdepiction of simulation models for network topology and elements using OPNET software.

Voice is the basic communication service which is well implemented in the circuitswitched networks, by introduction of LTE network, voice service is shifted to packedswitched mode from circuit switched mode and VoIP is applied to the network. VoIPconvert the voice signal to digital packet and transfer the voice data via packed switchedsystem. The principalbenefit of VoIP is that the cost of voice service is significantlydecreased and it is anadaptable voice preference for users.However, the network condition decides theQoS (quality of service) ofVoIP (Voice over Internet Protocol). Slow network connection leads to poor voice quality and the voiceservices will not work inappropriateway. Thus, performance analysisof VoIP on LTE network is required.

Additionally, the majorbenefit of LTE network is that it has the maximum data rate.Mobile consumers can get maximumvalue from the speedy data rate and willrelish the web browsingexperience. First of all the following parameters will be analyzed:

- Jitter
- End-to-end delay
- LTE delay
- Mean Opinion Score (MOS)
- Throughput

With the help of simulation results, the performance and the factors which can affect the performance of LTE network will be determined.

III MAIN PARAMETERS

By considering the following parameters while analysing the simulation results, the examination of QoS of VoIP and web browsing on the LTE network will be performed.

1. Jitter

While considering the packed switched networks, the packets are transmitted in continuous streams. The jitter may be defined as the deviation of time period between each transmitting bit. Congesting in the IP network results in Jitter and it happens at the receiver side. Due to jitter quality of voice can be poor and the intensity of jitter must be reduced, and hence it is asignificantconstraint in voice packet streaming. By adding anti-jitter circuits, jitter buffers, de-jitterizer, and filteringthe level of jitter can be decreased. According to ITU (InternationalTelecommunication Union) standard, the average value of jitter should be less than 60ms and the ideal value of jitter must be less than 20ms.

2. End-to-End Delay

The time taken by a packet to be transmitted from the transmitter to the receiver is defined as the End-to-end delay. The transmission delay, encoding&decoding delay, propagation delay and processing delay cumulatively forms the End-to-End Delay. It is a significant parameter for actual transmission because it is needed that the voice stream is transmitted in the well-timedway. According to ITU (International Telecommunication Union) standard, the value of average end-to-end delay must be less than 150ms and the value of ideal end-to-end delay must be less than 50ms.

3. LTE Delay

The total time taken from a packet being sent to the acknowledgement being received is called as the LTE delay. The distance between the consumer and base station, number of users and the applications (VoIP/web browsing) cumulatively decides the LTE delay.

4. Throughput

The rate of data successfully delivered to the receiver over a channel is known as throughput. The unit for throughput is measured as bits per second. By equating the throughput of different scenarios, it will beeasier to calculate and evaluate the quality of service(QoS) of each scenario.

5. Mean Opinion Score (MOS)

The quality of received voice after codecs transmitted and compressed is measured in a number value known as MOS. The MOS is the average of all the discrete scores, and it ranges from 1 (worst) to 5 (best). Moreover, the MOS score is influenced by severalfactors e.g. jitter, packet loss and end-to-end delay. Table 4.1 displays the ITU standards 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 Standards for MOS

IV OPNET IMPLEMENTATION

1. Overview of LTE Model in OPNET

The OPNET modeler contains a large library of models that supports several protocols such as TCP, UDP, SIP and it is well capable of simulating applications such as voice, web browsing, FTP etc. Additionally, the OPNET modeler works onhierarchicalsetting whichcomprises of the network model, node model and process model. All three models required to be configured to accomplish the simulation. The LTE network model in OPNET is comprised of mobile nodes, an E-Node B and an EPC.

2. Simulation Topology in OpnetModeler

For analyzing the performance of LTE network two test cases has been considered in the thesis work. In first network, simulation of the Voice over IP (VoIP) atvarious distances is being done andtheir simulationresults are compared. Insecond network, simulation of the web browsing at various distances and numerous IP consumers is being doneto analyze their simulation result.

3. Voice over IP (VoIP) in LTE Network

In the VoIP configuration, two scenarios in dissimilar distanceare being designed. Also,eNodeB is kept at halfway from the two mobile nodes in both scenarios.In one scenario mobile nodes and eNodeB are at 500 meters apart, and in the second scenario mobile nodes and eNodeB are 1000 meters apart.

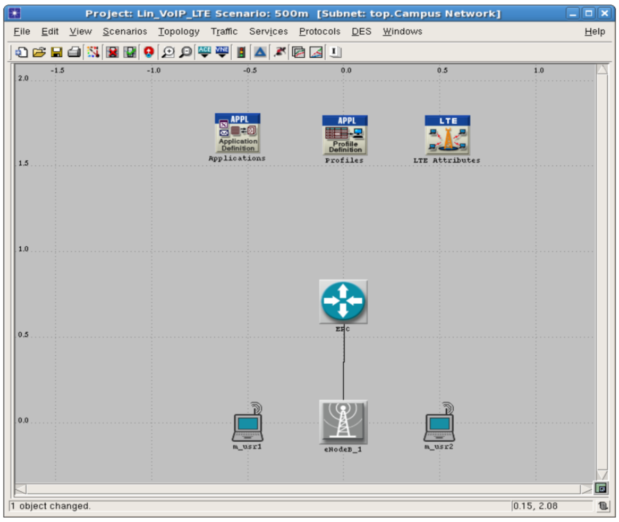

Figure 1: VoIP over LTE Design (500m) in OPNET Modeler

Figure 1 is the topology of the first scenario. The VoIP model and the VoIP Configuration parameter shown in Figure 2 below is setup by usingthe Application Definition attribute of the OPNET modeler. In the VoIP application, VoIP calls are generated by using the services of G.711 voice encoder and Interactive Voice. Once configuration of the application is complete, configuration of the Profile definition by using Profile Definition attribute is done. The configuration setsthestart time of the simulation to 100 seconds (off-set “60”+start time “40”) and till the end of simulation the VoIPapplication is reiteratedcontinuously. This simulation shows that VoIPcalls will be connected between transmitter and receiverbeginning at 100 seconds and thecalls are added constantlytill the end of simulation. Then, 20MHzbandwidth in the e-NodeB is chosen. Next, second scenario is created by altering thedistancebetween e-NodeB and mobile nodes.

(Voice) Table

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	(...)

Details Promote OK Cancel

Figure 2: Parameters of VoIP Configuration

4. Web Browsing on LTE Network

The following figure shows the topologies that have been implemented for the web browsing in LTE network. In this configuration the performance of the web browsing will be simulated and how the distance between e-NodeB and mobile nodes affects the performance of the network will be verified.The voice service will be the application. The figure below shows 3 different scenarios. Mobile nodes 1, 2, 3, 4 and eNodeB are kept atequidistance between mobile node 5 and eNodeB is changed as 500m, 1km and 1.5 km. The analysis will be for the effect of distance through the parameters: LTE delay and throughput.

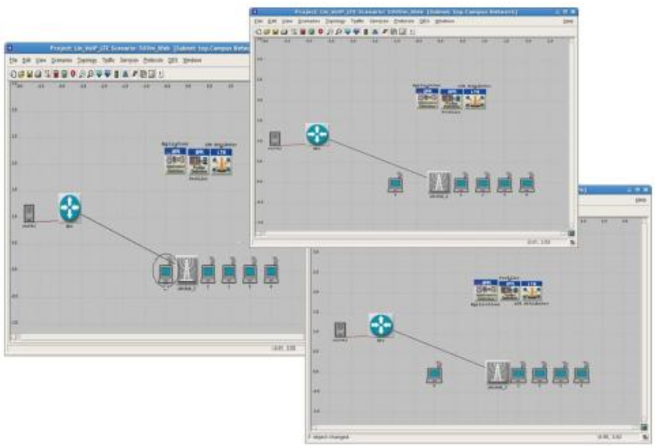

Figure 3: Scenarios for different distances

The following figure are the topologies that are being implemented to test the effect of number of mobileusers in the same LTE network for the web browsing. Those have the HTTP web browsing asthe application and add the HTTP server to the LTE network. The topology contains shows 3different scenarios. It has kept 4 mobile users in the first scenario, 8 mobile users in the secondscenario, and 16 mobile users in the third scenario. The analysis is to evaluate the effect ofnumber of mobile users through the parameters: LTE delay and throughput.

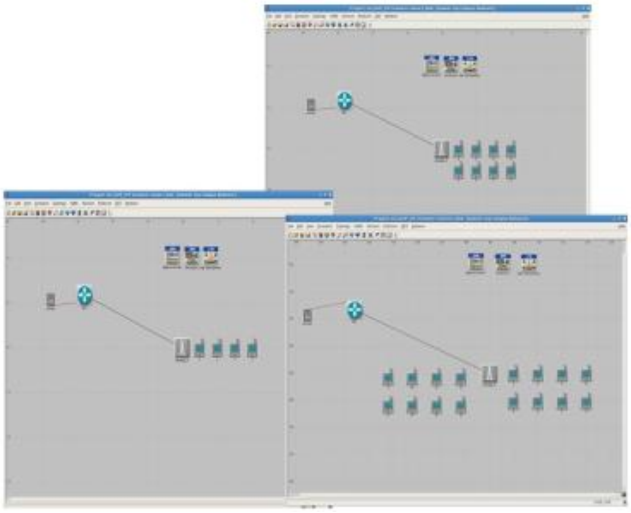

Figure 4: Scenarios for different number of users

(Http) Table

Attribute	Value
HTTP Specification	HTTP 1.1
Page Interarrival Time (seconds)	constant (10)
Page Properties	(...)
Server Selection	(...)
RSVP Parameters	None
Type of Service	Best Effort (0)

Details Promote OK Cancel

Figure 5: Web http configuration parameters

The above figure is the configuration parameter we set in the web http application. HTTP1.1 is set as the HTTP Specification. Page Intertribal Time is fixed 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:

(Server Selection) Table

Attribute	Value
Initial Repeat Probability	Browse
Pages Per Server	exponential (10)

Details Promote OK Cancel

Figure 6: Web server page setup

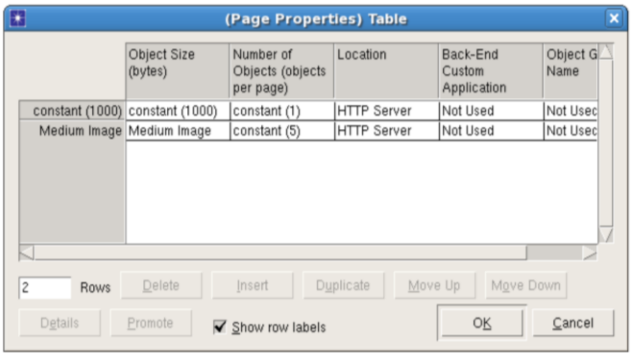

Figure 7: Page Size Setup

V SIMULATION RESULTS

1. VoIP Results

1.1 Jitter-The following figure shows the jitter result of VoIP. Considering the technical difficulties,the simulation 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 was stabilized, the jitter was less than 20ms, which is in the rangeof ITU ideal jitter range. Based on the 500m scenario result, it can beconcluded that the*jitter performance of VoIP on LTE network is excellent.*

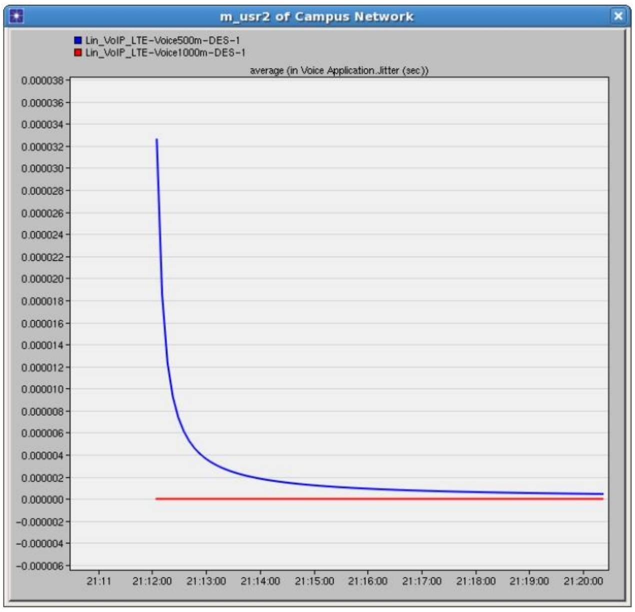

Figure 8: Jitter result for VoIP

1.2 End to End Delay-For the end-to-end delay, it isobvious that 500m scenario has better performance than the1000m scenario. The average end-to-end delay for500m scenario is about 81ms and theaverage end-to-end delay for 1000m scenario is around 97ms. Compare to theITUstandard, the end-to-end delays for both scenarios are below the average rate, whichmeans the end-to-end delay performance of VoIP on LTE network meets the ITUrequirement. In addition, the end-to-end delay is increasedwhile the distance is increased.

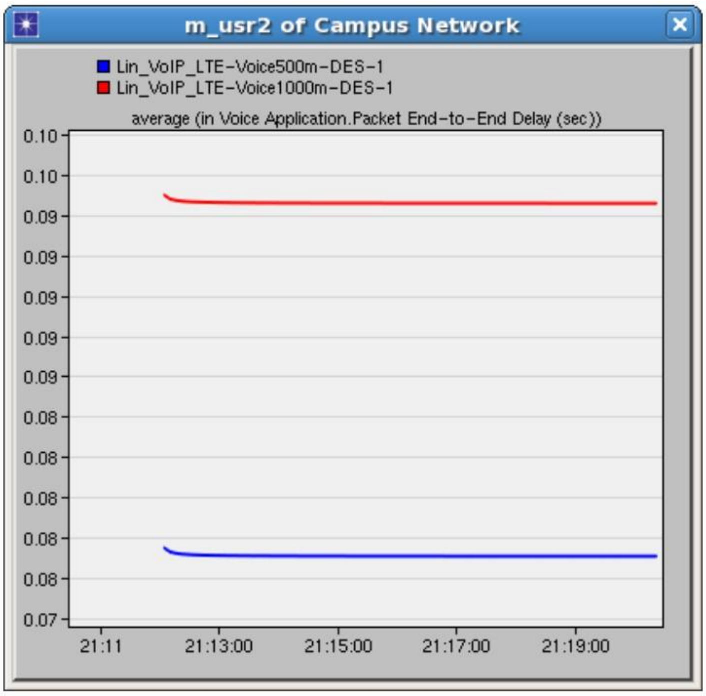

Figure 9: End-to-end delay result for VoIP

1.3 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.

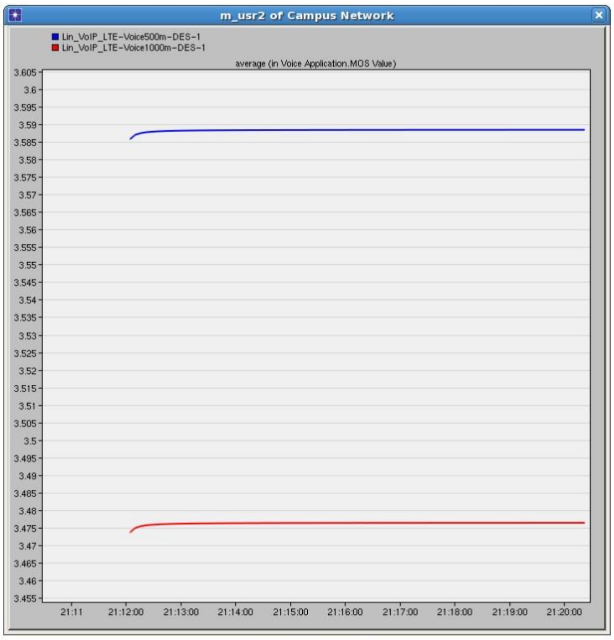

Figure 10: MOS result for VoIP

2. Web Browsing Result

2.1 LTE Delay of Various Distances

In the figure below, the blue line shows the LTE delay of the 500 meter scenario, the red lineshows the result for 1000m scenario and green line shows result of

1500mscenario. It isobvious that the shorter distance between users and eNodeB can lead to the shorter LTEDelay. At the beginning of simulation, the 500m scenario has thebest initial delay of the3 scenarios. Then the LTE delay ofthese 3 scenarios starts to decrease. The final averageLTE delay for 500m scenario is about 1.55ms and the finalaverage LTE delay for 1500mscenario is around 1.57ms.When these 3 scenarios reach the stable, the delay of the500m is still the lowest one.

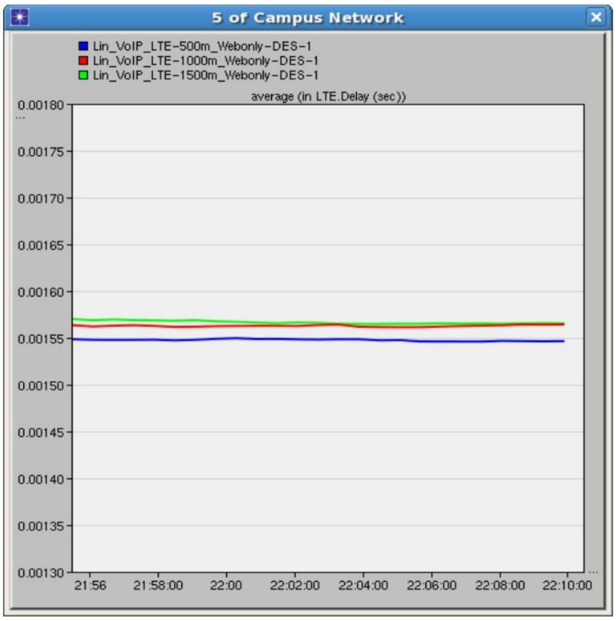

Figure 11: Delay of various distances

2.2 Throughput of various distances

Based on the maximum throughput value in figure 14, themaximum throughput for 500meters, 1000 meters and 1500 meters are 3100 bits/sec, 3050 bits/sec and 3000bits/sec,respectively. It is obvious that the shorter distancebetween users and eNodeB can lead tolarger maximumthroughput. This is what we expected.

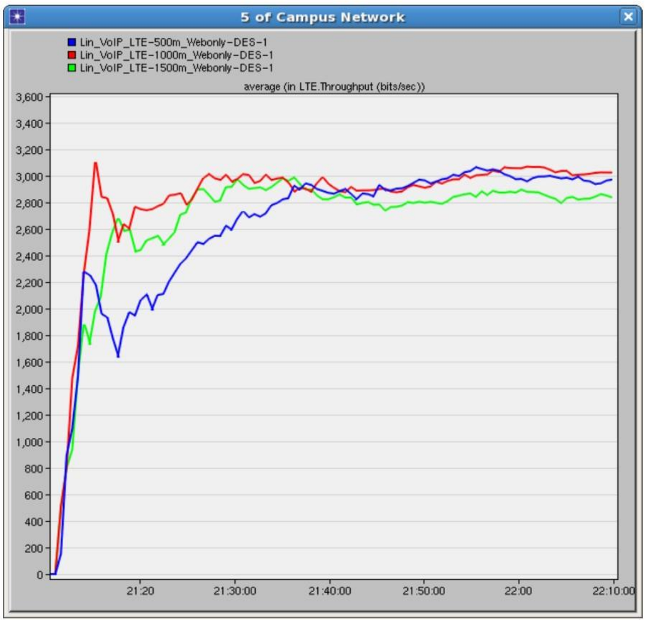

Figure 12: Throughput of various distances

2.3 LTE Delay of various number of IP user

In the figure below, the blue line shows the LTE delay ofthe scenario which contains 4 IPusers while red and green lines show other two scenarios which contain 8 IP usersand 16 IPusers, respectively. It is obvious that increasing the number of IP user can increase theLTE Delay value. The average LTE delay for 4-user scenario, 8-userscenario and 16-userscenario are 1.75ms, 1.78ms and1.79ms, respectively.

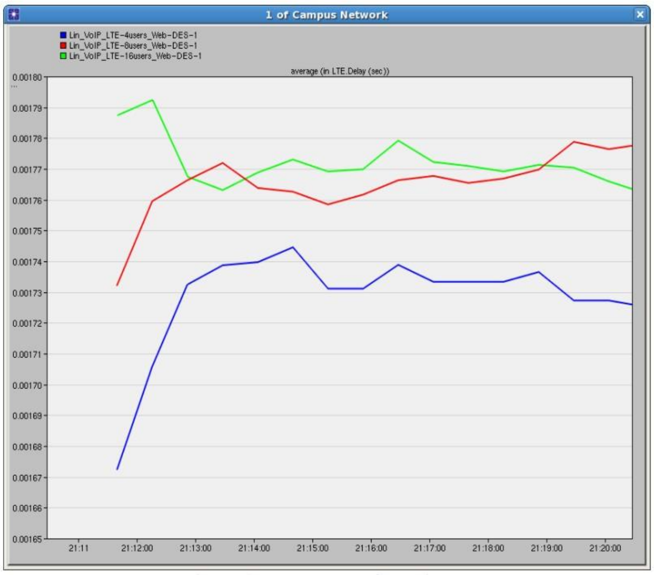

Figure 13: LTE Delay of Multi-user

2.4 Throughput of various number of IP user

The figure for multi-user throughput is shown below, it isobvious 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 maximumthroughput for 4-user scenario, 8-user scenario and 16-user scenario are 2200 bits/sec,2300 bits/sec and 2800 bits/sec, respectively.

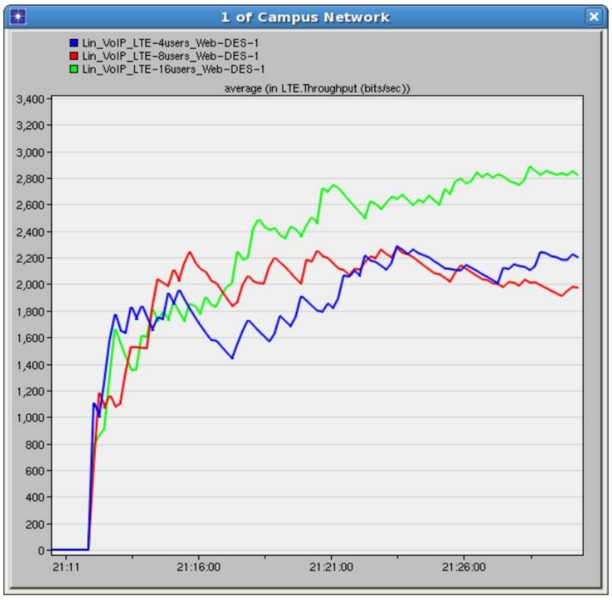

Figure 14: Throughput of Multi-user

VI CONCLUSION

For voice over IP on LTE, the results of OPNETsimulation agree with theory. There isincrement in End-to-End delay when the distance between users andeNodeB is increased.Moreover, the MOS value isdecreased as there is an increase in the distance betweenusers andeNodeB. In other word, when the distance between the users and eNodeB decreased, thequality of VoIP decreased.

For Web Browsing on LTE, it is obvious that theincrement in the distance between usersand eNodeB and the increment in number of IP users will increase the LTE Delay valueand maximum throughput.

Therefore, it can be conclude that in the same network, the fewer users or the closer the user beside the eNodeB, the better internet browsing performance.

Reference

[1]. Alcatel-Lucent Inc. "The LTE NetworkArchitecture".2009(online).Available:http://www.cse.unt.edu/~rdantu/ FALL_2013_WIRELESS_NETWORKS/LTE_Alcatel_White_Paper.pdf

[2]. B. Furht and Syed A. Ahson, "Long Term Evolution: 3GPP LTE Radio And CellularTechnology", Crc Press, 2009

[3]. Cisco systems Inc. "Understanding Codecs: Complexity, Hardware Support, MOS,and Negotiation". Feb 02, 2006 (online). Available:http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a00800b6710.shtml#mos

[4]. Cisco systems Inc. "Understanding Jitter in Packet Voice Networks (Cisco IOSPlatforms)". Feb 02, 2006 (online). Available:http://www.cisco.com/c/en/us/support/docs/voice/voice-quality/18902-jitter-packetvoice.html

[5]. D.Sahota. "LTE falls back on voice". Sep 10, 2013 (online).Available:http://www.telecoms.com/177492/lte-falls-back-on-voice/

[6]. G. Camarillo and M. Garcia-Martin. "The 3G IP MultimediaSubsystem(IMS):Merging the Internet and the Cellular Worlds", 2006.

[7]. Kurose, J. F. & Ross, K. W. (2010) Computer Networking: A Top-Down Approach.NewYork: Addison-Wesley. P 30

[8]. S. Sesia, I. Toufik, and M. Baker, "LTE – The UMTS Long Term Evolution – FromTheory to Practice". Second Edition including Release 10for LTE-Advanced, John Wiley& Sons, 2011

[9]. P.Guraauraj. "Masters Thesis: Voice overLTE". Jun 29, 2012 (online).Available: http://repository.tudelft.nl/assets/uuid:a1ee990a-31bc-4f8b-9e27-0b0617e5a08f/Msc_Thesis_Report_Prasanna_Gururaj.pdf

[10]. The Mobile Broadband Standard, 3GPP TS 22.173, IP Multimedia Core NetworkSubsystem (IMS) Multimedia Telephony Service and supplementary services; Stage 1

[11]. Torad, Mohammad, Dr. "Comparison between L TE and WiMAX Based on SystemLevel Simulation Using OPNET Modeler (release 16)." 28 Th NATIONAL RADIOSCIENCE CONFERENCE (2011).