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QoS Voice and Video over IP at LTE Network

Aasim Ahmed*[1], *Khalid Hamid*[2]

[1]*Faculty of Engineering, Alneelain University, Khartoum. Sudan*

[2]*University of Science and Technology, Khartoum. Sudan*

Corresponding author E-mail: aasim124@live.com

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ABSTRACT

This paper identifies the parameters that affect the Quality of Service (QoS) of Voice and video-over-Internet Protocol (VoIP) through heterogeneous networks such as LTE and between them. They are carried out by using the OPNET simulation tool. Optimization of the network for both intra- and inter-system traffic to mitigate the deterioration of the QoS are discussed. Also, this paper focuses on some parameters of QoS: jitter, delay, packet loss, throughput and MOS.

Keywords— Voice over Internet Protocol; Quality of Service; OFDMA; SC-FDMA;

I. INTRODUCTION

Long-term evolution (LTE) is a new technology that is an extension of the previous generations of mobile phones, which started in the first generation or what was known here in the region by GSM and the second generation, which is most prevalent globally, which began to spread in the early nineties, and this generation was marked when it first generation as was a technology Digital quality, greater capacity and security than the first generation. After a period of success, the second generation did not meet some of the requirements imposed by the rapid development in the world of telecommunications, especially information technology, so the third generation was developed. The first improvement of the 4G network is of course speed, with the new technology providing faster data transfer speeds of 4 to 10 times then 3G. It is difficult to give figures here because the maximum 3G speed is not fixed and varies from country to country, but in general, the 4G will be several times faster by any means[1]. Other Increase productivity and transfer data are more quickly, Low latency, Plug and play, improve users' uses, Lower operating costs and Seamless. LTE mobile technology provides ever new and better ways of providing higher level of the graphic user interface and it provides high level of online gaming, multimedia and better video quality. Speed is the most important to you as a regular user, but the new standards will offer a number of other improvements such as security and protection, better switch between towers and networks, better international roaming, higher data rate, and other improvements. LTE can provide us download data rates up to 150Mbps for the multi antenna (2x2) multiple input-multiple outputs (MIMO) which are for the highest terminal category and for upload of data up to 50 Mbps. LTE makes efficient use of the spectrum with the available bandwidth of 1.25 MHz to 200 MHz LTE uses Orthogonal Frequency Division Multiple Access (OFDMA) IEEE 802.16.[2-5] table 1 show LTE characteristics.

Table 1: Characteristics 4G Network Technologies:

Key features	LTE Network
Data rate	20 – 100 Mbps
Frequency band	2 – 8Ghz
Bandwidth	About 100MHz
Switching technique	Fully digital with packet voice
Radio access technology	OFDMA_ IEEE 802.16.
IP	IPv6.0
QoS and security	Supported
Multi-antenna techniques	Supported
Multicast/broadcast services	Supported

II. QUALITY OF SERVICE (QoS)

This paper is interested in the quality of service that LTE can add to users and operators. Quality of Service enables operators to optimize the use of network resources. QoS is effective to improve the overall performance of the network. The term communication arises when the data flow between sender and receiver while there is reliable magnitude of packet loss, delay jitter, delay and data rate for proper guarantees of QoS. For the real-time applications such as voice, and video conference QoS is essential. In heterogeneous network resource reservation and traffic congestion are important issues in terms of the performance of overall network[6].

III. VIDEO AND VOICE OVER IP

Voice and video over IP is a digital movement of audio and video using data networks to hold calls, meetings and conferences. The packet switching is used here. The data is divided into packets. These packets take anyway to reach the other part. The circuit does not need to reserve the length of the call or conference period between the sender and the receiver. Therefore, all possible channels are used. This method is better than the old method[3]. It is called circuit switching. To make calls, meetings, and conferences via internet protocols, control information and signals to be exchanged between network entities[7].

These are complex facts because the Internet is shared across different types of devices and different types of networks. Hence, the need for a Session Initiation Protocol. SIP works alongside and in complement with the existing real-time protocols[8]. The source and destination endpoints are known as the user agents discover each other and then negotiate the parameters for the efficient exchange of information by the use of SIP. The necessary user agents and intermediary nodes are handled by SIP by the creation of proxy servers. These proxy servers can then request and respond to invitation registration and other such SIP requests [6]. SIP is a transport protocol independent of the type of session being established. SIP is designed to be agile flexible and to handle various types of multimedia data exchange [3, 9-12].

QoS parameters of voice and video the QoS parameter differs in a video, voice traffic, and it can be quantified by a range of different metrics, such as jitter, delay, throughput, packet loss and MOS.

Delay: the amount of time that takes a packet to transverse the network; jitter, the variation in delay from packet to packet; bandwidth, the data rate that can be supported on the network; and packet loss, the percent of packets that do not make it to their destination for various reasons.

Jitter: This refers to the variability of latencies for packets within a given data stream and should not exceed 20 - 50 milliseconds. an under-run condition may occur at the receiving endpoint, potentially causing either blocky, jerky video or undesirable audio. Too much jitter can cause inter-stream latencies.

Packet loss: This term refers to the loss or dose quenching of data packets in a real-time audio/video data stream.

MOS: the main opinion score recommended by ITU-T in 1996 is the most widely used subjective measure of voice quality. MOS value is normally obtained as an average opinion of quality based on asking people to grade the quality of speech signal on the five points scales (excellent =5, good=4, fair=3, poor=2, bad=1) under controlled conditions as set out in the ITU-T standard p.800.

II. SIMULATIONS

Simulation activities of video and voice services are performed using OPNET Modeler 17.5 PL6 software tool, based on Discrete Event Simulation (DES) approach. Since the purpose of this work is to analyze only influence of different voice coded on end-to-end performance of video and voice, Type of Service (ToS) considered is only Best Effort (BE) [13]. OPNET SETTING: Figure1 Show the topology of simulated LTE network in a typical campus area 10 x 10 Km.

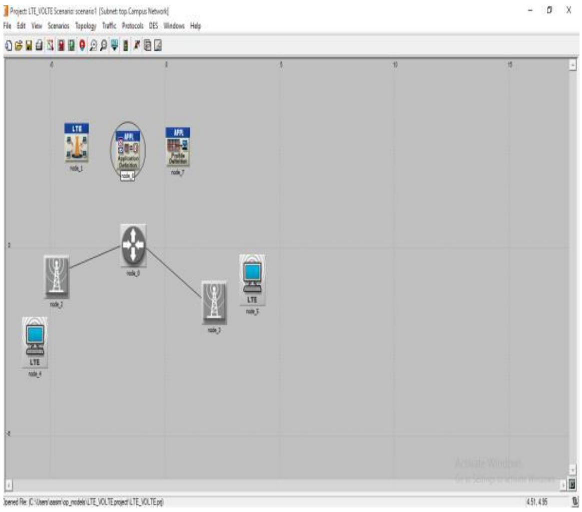

Figure 1: The topology of simulated LTE

Models of LTE network nodes are: LTE wkstn: LTE workstation or UE Entire network topology is composed two UEs: - Voice and video source, Voice and video destination

LTE_enodeB_3sector_slip4_adv_1_upgvrade: LTE_enodeB with 3 sectors two different enodeBs are considered, one for each UE.

LTE_epc_atm8_ethernet8_slip8_adv: LTE EPC node. ppp_DS3: link mode.

OPNET modeler management nodes are: app_config: application configuration node. profile_config: profile configuration node. In table 2 haw to User Equipment Setting in OPNET and Figure 2 show ENodeB Setting in OPNET.

Table 2: User Equipment setting

Parameter Description	Parameter Value(s)
Antenna gain	- 1 dBi
Multipath Channel mode (Downlink)	LTE OFDMA ITU Pedestrian B
Multipath Channel mode (Downlink)	LTE SC-OFDM ITU Pedestrian B
Path loss	Free space
Receiver sensitivity	-200 dBm

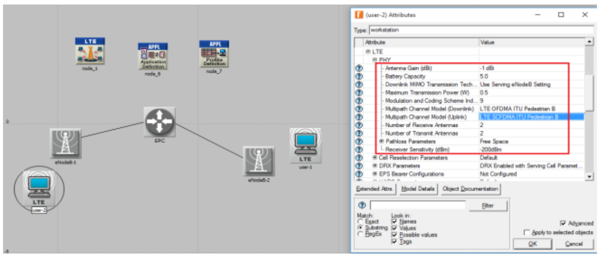

Figure 2: ENodeB setting

Application configuration in OPNET modeller several applications are predefined and suitable. In this paper voice and video, the application is selected. A new two applications are created (voice, video). The applications are launched with a start offset of 40 seconds till the end of the simulation period. Profile configuration unique two profiles are created. One named Voice Profile and anther named video profile. Figure 3 show voice and video setting in OPNET.

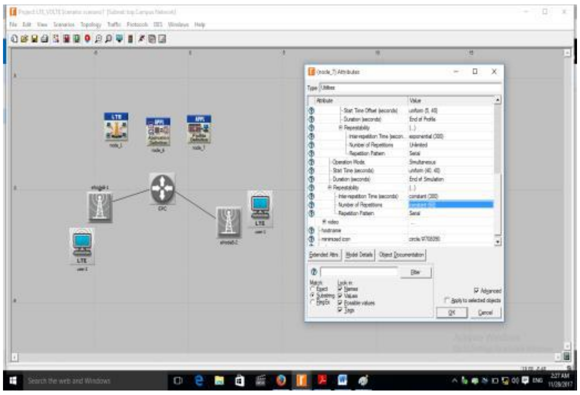

Figure 3: voice and video setting

DES settings: For all scenarios, simulation period is 3 minutes. Main DES settings are listed in Table 3:

Table 3: DES Setting

Attribute	Value
Duration	120 second
Seed	128
Value per statistic	100
Update interval	5000.000 events
Number of runs	1

Statistics: DES statistics selected for Voice and video simulation. Global statistics: (simulation results are provided at the entire network level): IP, LTE, Voice, and video conference. Node statistics: (simulation results are provided at network single node): IP, LTE, LTE PHY, UDP, Voice, and video conference.

III. SIMULATION RESULTS

In this paper simulation results based on KPIs discussed. Figure 4 shows for each scenario their graphic representation in terms of voice jitter Figure 4, voice packet delay variation Figure 5, voice end-to-end packet delay Figure 6, MOS Figure 7, Video packet Delay variation Figure 8, video end-to-end packet delay Figure 9.

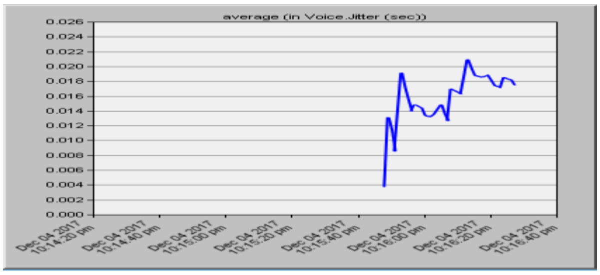

Figure 4: voice jitter

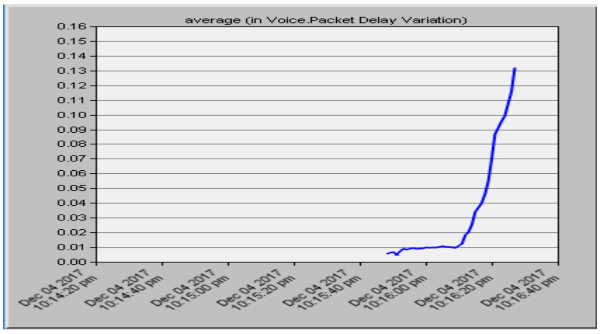

Figure 5: voice delay

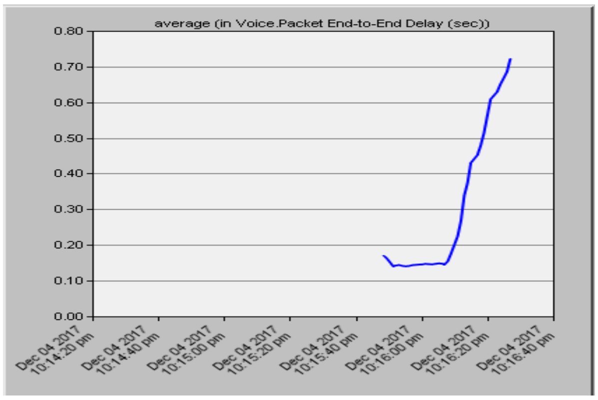

Figure 6: voice PacketD

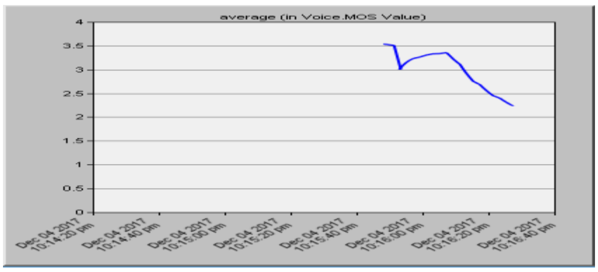

Figure 7: MOS

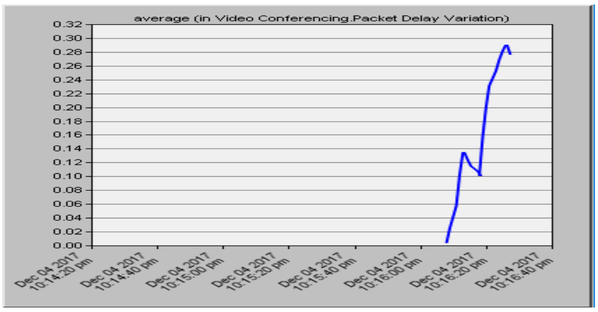

Figure 8: video Delay

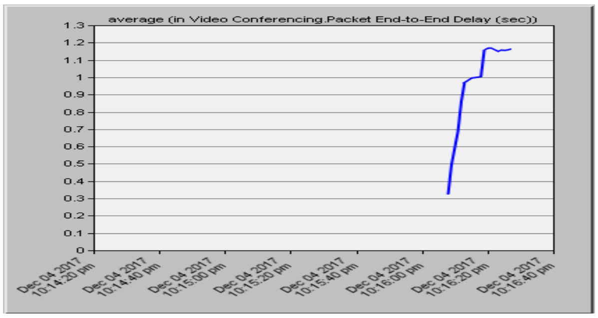

Figure 9: video Packet Delay

IV. CONCLUSIONS

Purpose of this work is to evaluate end-to-end QoS of voice and video. Network transmission factor is considered. An efficient analysis of end-to-end QoS KPIs is presented. It based on MOS, end-to-delay, voice traffic sent and received, jitter, voice and video packet delay variation, voice and video downlink and uplink delay.

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