QoS of VOIP Over Broadband Network

Mutasim Mohammed EL Khier

Assistant Professor, Department of Computer Science and Information, Majmaah University / College of Science in Al-Zulfi, Kingdom of SAUDI ARABIA

Corresponding Author: mm.elkir@mu.edu.sa

ABSTRACT

The paper discussed the implementation of the use of Voice over Internet Protocol services, which is resulting in the enormous growth of broadband network. The main objective of this paper is to evaluate the QoS of VOIP for different broad band networks. Wired, Wireless Local Area Network (WLAN), Worldwide Interoperability for Microwave Access (WiMAX) and Universal Mobile Telecommunication System (UMTS) networks were implemented in OPNET Modeler The quality is compared using different QoS parameters such as end-to-end delay, Mean Opion Score (MOS), throughput and jitter. The VoIP codes used in the measurements of QoS is G.729A with sampling rate of 8 kbps .The results analyzed and the performance evaluated will give network operators an opportunity to select the codec for better services of VoIP for customer satisfaction.

Keywords— QoS, UMTS, VoIP, WIMAX

I. INTRODUCTION

Ever since its advent VoIP has opened new doors for telephony bringing forward immense possibilities. The basic reason for the popularity of VoIP is the cost which is very low as compared to the conventional telephony services. The concept of transmission of voice over data stream makes it possible to have VoIP transmitted and received using anything that uses IP - laptops, PC's, WiFi enabled handsets etc..

Voice over Internet Protocol (VoIP) are likely to increase day by day, leading to rapid network improvements. There is a demand to reduce the differences between the qualities of voice and increasing the available bandwidth to provide the best VoIP service[1].VoIP has almost replaced the traditional Public Switched Telephone Network (PSTN) due to its cost effectiveness and the features being offered [2]. The wired Internet Protocol (IP) networks provide better VoIP services than wireless IP network since wireless networks have their own characteristics and impairments [3]. The unsolved problem caused by the wireless network in this area still needs to shed some light on the dedicated VoIP calls. In next generation networks wired and wireless systems have been combined in an innovative way under a single

framework [5]. The frequent handovers cause delays and packet loss in these network [6]. The VoIP call gets degraded and loses the packets more swiftly. An eternal solution is required for these heterogeneous systems for the VoIP communication.

The aim of this paper is to provide good quality of VoIP services in every network using G.729A code mentioned in table(1). The QoS of VoIP packets are evaluated in terms of end-to-end delay, MOS, throughput and jitter over Wired, Wireless, UMTS and WiMAX networks using the OPNET Modeler.


The Simulation Network

VoIP is a blend of hardware as well as software that uses the Internet as a transmission medium. IP networks are accomplished of processing all kinds of network traffic, which includes voice as well. The ability and quality of a VoIP communication for conversation is governed by various elements such as network settings, coding process, speech content, kind of error correction etc. In addition to voice calls the VoIP also offers services such as fax, send message service and voice-messaging applications. The process involved in transmitting these services over the packet switched network (Internet) are digitization and encoding of the analog voice signal followed by packetization, signaling and media channel set up. The analogous steps are employed excluding for decoding and digital-analog alteration to generate original signal at the receiving end.

II. VOIP AND CODECS

The demand for mobile and broadband services is rising dayby day. The VoIP users was increased in the with

the demand of reliable and good quality services. VoIP is an emerging technology for voice communication used these days.
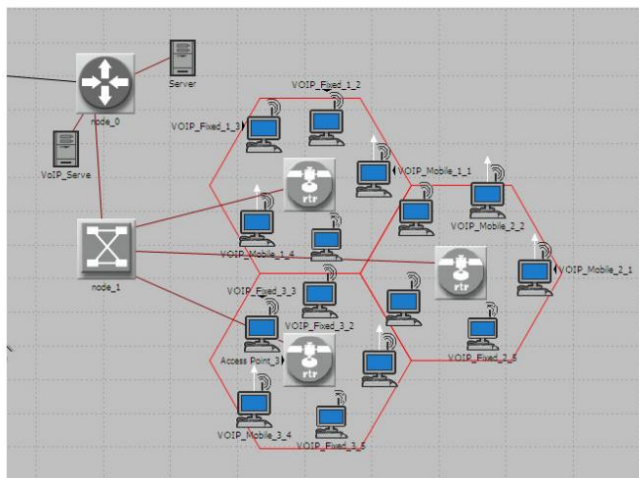
The Wireless Subnet

The services are not only being used for long distance calls but also for the short distant communications. The devices like IP phones and desktop systems provide some new features to the users. Keeping in mind the demand of the users, the operators are forced to improve the quality of communication.This can be achieved by increasing the bandwidth and making the IP backhaul that fulfills the demand of the users at lower cost providing better QoS.

III. VOIP CODECS

Codec is a coder/decoder which converts the audio signal to digitized version for transmission over the medium and then back into the original uncompressed version on the receiver side. This concept is the base of VoIP services. There are a number of codec used for VoIP communication each having its own bandwidth and characteristics. The codecs which are used in this paper is :G.729A using Conjugate structure Algebric Code Excited Linear Prediction (CS-ACELP) as illustrated in the TABLE 1.

TABLE 1
CHARACTERISTICS OF VoIP G.729Acoder

codec	Coding algorithm	Sampling rate
G.729A	CS-ACELP	8 kbps

IV. NETWORK MODELS

The tool used for simulations is OPNET Modeler as it provides the results very closer to the real time environment. The models were created by selecting the nodes and links from the object palette such that to reduce the impairments effect. Wired model designed, is a general IP network. Links in the wired design as shown in figure 1 consist of standard 100baseT lines from user to router and from router to internet cloud followed internet server is T1 line. WLAN design consists of user node and access point connected to the IP backhaul with a T1 line as shown in figure 2. UMTS model as in figure 3 comprises user equipments, node B and Radio Network Controller (RNC) which is connected to the packet switched network via Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN) which in turn is connected to the IP Network. Figure 4 represents the WiMAX model which is designed using the base station connected to the IP backhaul serving the VoIP users. A T1 line is used to simulate a perfect connection between router and server minimizing cable delay and allowing the difference caused by the codecs to be more noticeable. The attributes and parameter settings are made in the network models and various simulations are carried out for the codec. The reason for utilizing this modeling method is to allow performanceof the codec to be analyzed in an improved manner.
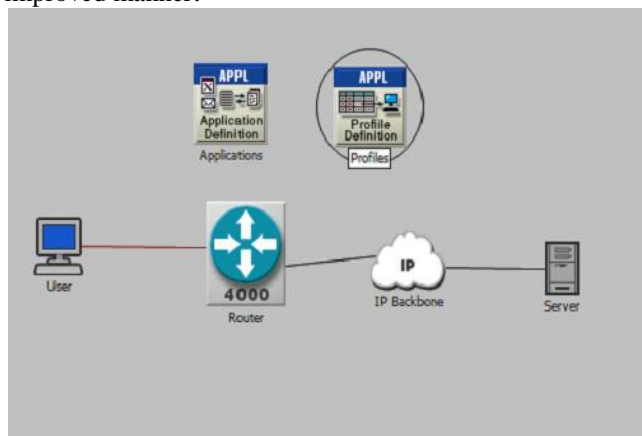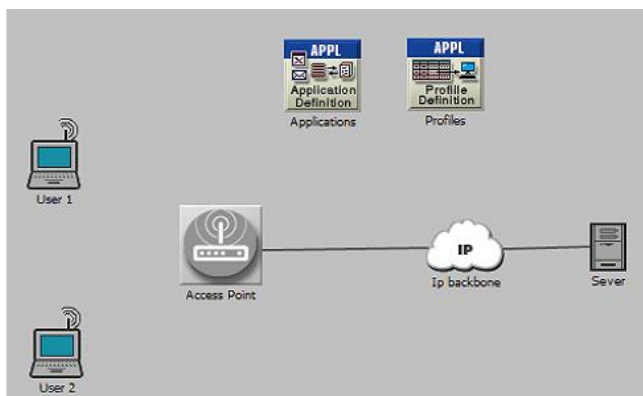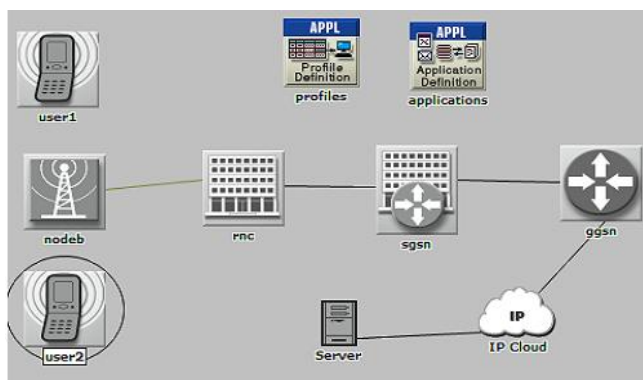

Figure 1: Wired Model

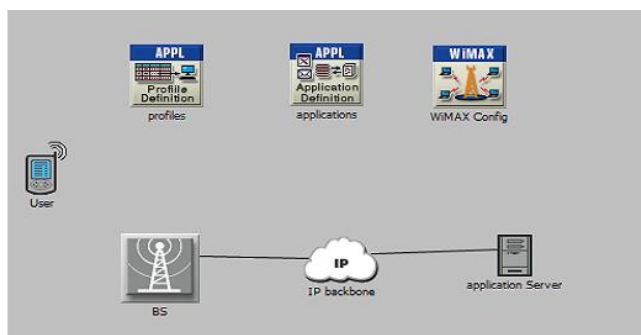Figure 2: WLAN Model

Figure 3: UMTS Model

Figure 4: WiMax Model

V.   RESULTS AND DISCUSSION

After execution of the simulation for G.729A codec in different networks. Theresults obtained are shown in figure5 to figure 9.

Figure 5 shows that the value of Mean Opinion Score is3.2 in wired, WLAN and WiMAX models. Comparatively, MOS value for UMTS model is 2.4 showing the bad quality of speech. WiMAX model along with the Wired model show the best quality of VoIP. Jitter and end to end delay in figures 6 and 7 show that WLAN and UMTS models undergo a delay in

packets and attain some jitter which in turn loses the packets. In WiMAX jitter and the amount of delay is very small hence providing the good quality of VoIP. Traffic sent and received is almost the same in wired and WiMAX models while there is some loss of packets in WLAN and UMTS networks as shown in figure 8 and 9. Jitter, delay and full reception of packets in WiMAX model represent that it gives best quality of voice while using G.729A.The QoS performance of WLAN and UMTS models is not effective as there is a delay and packets are lost.
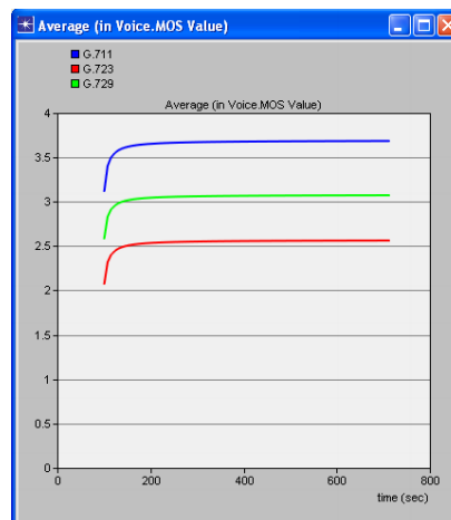

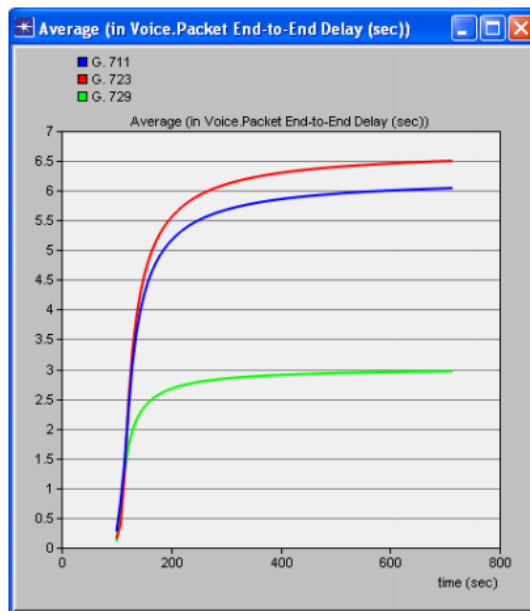Figure 5: Average voice MOS under various codecs

Figure 5: Average voice packet End to End delay (sec) under various audio codecs
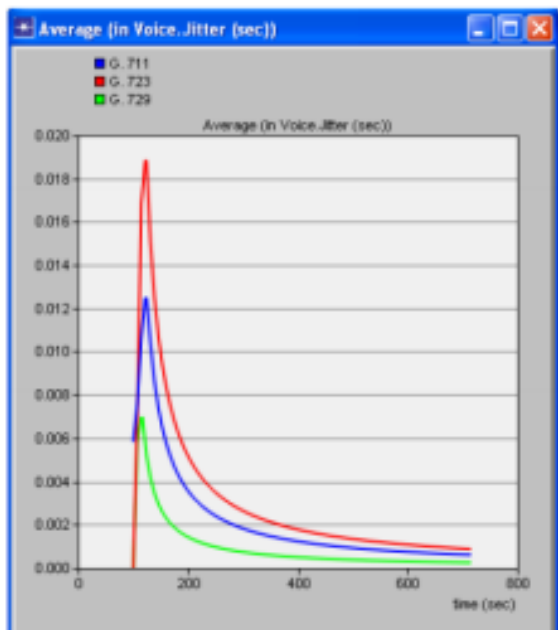
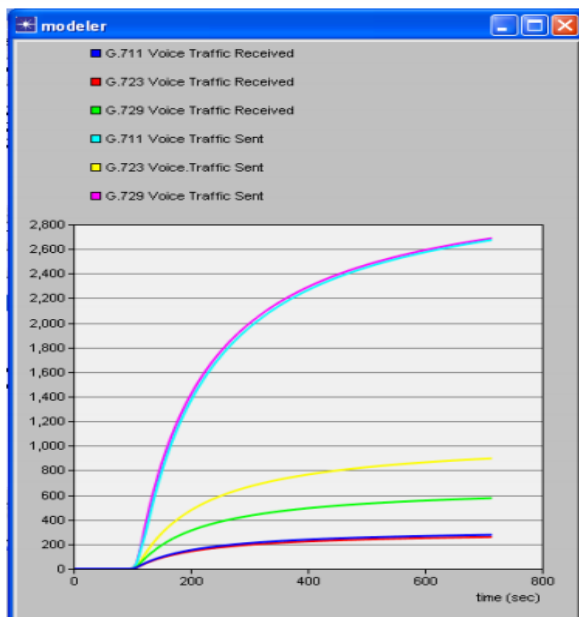Figure 6: Average Jitter under various audio codecs

Figure 7: Average voice traffic sent and received under various audio Codec

VI. CONCLUSION

The QoS Performance of VoIPcodecs using G.729A in different networks is analyzed using the OPNET Modeler. A variety of simulations are carried out to get the most effective and efficient results. On the basis ofresults obtained, it isconcluded that wired network QoS performs well compared with wireless broadband network for VoIP communications.

REFERENCES

[1] U. R. Alo & Nweke Henry. (2013). Investigating the performance of VOIP over WLAN in campus network. *Journal of Computer Engineering and Intelligent Systems, 4*(4), 47-58.

[2] S. Brak & et al,. (2013). Speech quality evaluation based CODEC for VOIP over 802.11P. *International Journal of Wireless & Mobile Networks, 5*(2), 59-69.

[3] Hussein, & et al,. (2013). The effects of different queuing algorithms within the router on QoS VoIP application using OPNET. International Journal of Computer Networks & Communications, 5(1), 117-124.

[4] Y. Jung & C. Manzano. (2014). Burst packet loss and enhanced packet loss-based quality model for mobile voice-over Internet protocol applications. *Journal of IET Communications, 8*(1), 41–49.