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Evaluation of G.711, 723.1 and 729A VIOP Codec Using Guided and Unguided Communication Network

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Abstract: ***Voice over Internet Protocol is a certain technology that lets telephone calls to be easily made over a wide number of computer networks like the Internet. There are different implementations of VoIP on the internet at the present time. Numerous of leading telephone companies like AT&T have totally transferred to VoIP. It is still not obvious how the performance differs with several network conditions, considering that the Internet does not provide QoS warranties. The main goal of this paper is to assess the performance of G.711 VIOP coder for various 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, MOS, throughput, and jitter. The VoIP co codecs applied in the measurements of QoS are G.711, 723.1 AND 729A. Simulations appeared that G.711 is the best project that gives high quality of voice in Wireless Local Area Network (WLAN) communications. The results analyzed and the implementation evaluated will award network operators a suitable chance to select the codec for better services of VoIP for well customer satisfaction.***

Keywords: VoIP; WLAN; WIMAX; UMTS; Codec; QoS

1. Introduction

Voice over Internet Protocol (VoIP) practices is potentially growing day by day resulting in the request for of quick improvements in the networks. There is a high desire to decrease the difference between the qualities of voice and increasing the available bandwidth to give the best VoIP services comparative to the classic circuit switched telephony [1]. VoIP has almost substituted the classic Public Switched Telephone Network (PSTN) because of its effectiveness of cost and provided features [2]. The wired Internet Protocol (IP) networks supply better VoIP services comparison to the wireless IP network as wireless networks have their own advantages and weakness [3]. The unsolved cases caused by the wireless network in this zone still need some true work for spotlighting VoIP calls. In the following generation networks wired and wireless systems have been combined in a creative way under a single framework [5]. The chronic hangovers cause lateness and packet loss in this network [6].The VoIP call gets degraded and loses the packets more swiftly. An eternal solution is wanted for these varied systems for the VoIP communication. This paper is to provide perfect quality of VoIP services in every network and analysis is done using different codecs mentioned in table1.VoIP packets are tested focusing all the major parameters such as end-to-end delay, MOS, throughput and jitter over Wired, Wireless, UMTS and WiMAX networks using the OPNET Modeler.

The investigators have been able to achieve some perfect results but as most of the IP networks and their underlying protocols [4] in use at present were performed keeping in mind the data services, not the real-time and delay-sensitive voice services [18], there is yet a real need of providing better QoS as per the orders of the users. Each set of CODECs is being used nowadays having its own characteristics [19]. Mostly used the codec for VoIP is G.711 which provide perfect results for wired network depending on the environment and conditions [20] but when it comes to wireless networks the quality is degraded. The internet services are becoming so complex that VoIP performance parameters need some actual measurements, unlike the classical telephone networks which were dependent on mathematical designing. This research is prepared to provide a good type of VoIP services in every network and analysis is done using various codecs shown in table 1.

2. VoIP and Codecs

The demand for mobile and broadband services is rapidly growing day by day. The latest decade has seen the ever-increasing VoIP users with the request of good reliable and high-quality services. VoIP is an emerging technology for voice communication used these days. Nowadays VoIP is an emerging technology for voice communication used. The services are used for both long distance calls and the short distance communications. The devices like IP phones and the VoIP enabled desktop systems to perform cost effective and also provide some new features to the users. Consideration in mind that the requests of the users have forced the operators to improve the communication quality.

This can be accomplished by increasing the bandwidth and making the IP backhaul that attains the user's demand at lower cost providing better QoS.

2.1 VoIP Codecs

Codec is a coder/decoder which transforms the audio signal into the digitized version for transit over the medium and then back into the original uncompressed version on the recipient side. This concept is the principle of VoIP services. There are several numbers of codec used for VoIP communication each of them having its own bandwidth and characteristics. The codecs which are used in this research paper are listed in table I below.

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Table 1: Characteristics of VoIP Codecs

CODEC	Coding Algo	Sampling rate
GSM- FR	PRE-LTP	13 kbps
G.711	PCM	64 kbps
G.723.1	ACELP	5.3 kbps
G.729A	CS-ACELP	8 kbps

3. Network Models

The tool used for simulations in OPNET Modeler as it provides the results very nearer to the real time environment. The models were formed by selecting the nodes and links from the object palette so as to reduce the losses/impairments affect. Wired model designed, is a general IP network. Links in the wired designing as shown in figure 1 consist of standard 100base T lines from user to router and from router to internet cloud followed internet server is T1 line. WLAN designing consists of user node and the access point connected to the IP backhaul with a T1 line as appeared at figure 2. UMTS model as in figure 3 includes user equipment, node B and Radio Network Controller (RNC) which are joined to the packet switched network via Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN) which in turn is linked to the IP Network. Figure 4 represents the WiMAX model which is designed using the base station linked to the IP backhaul serving the users of VoIP. A T1 line is used to simulate an ideal connection between router and server minimizing cable lateness and let the difference caused by the codecs to be more remarkable. The attributes and parameter settings are made in the network models and different simulations are holding out for the codecs. The reason for using this modeling method is to allow performance of the codecs to be analyzed in an improved way.

3.1 Wired

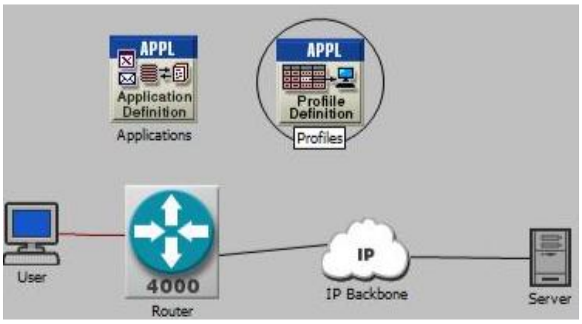

Figure 1: Wired Model

3.2 WLAN (Wireless)

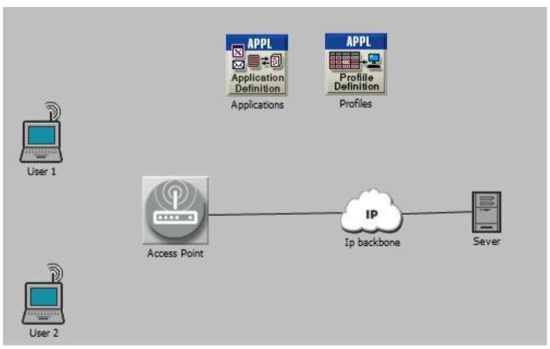

Figure 2: WLAN Model

3.3 UMTS

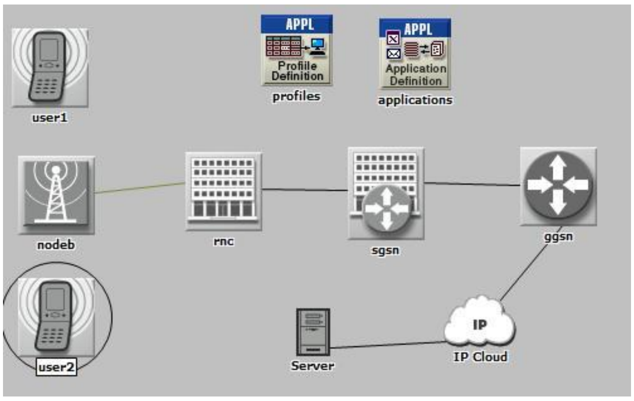

Figure 3: UMTS Model

3.4 WiMAX

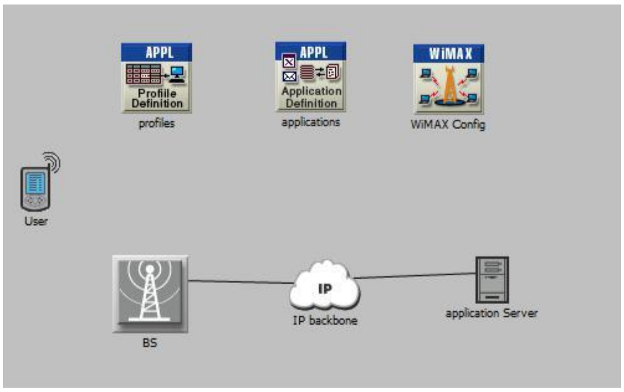

Figure 4: Wimax Model

4. Results and Analysis

The comparative analysis of UMTS, WiMAX, Wired and WLAN a network using each code while keeping the simulation environment and attributes same, is discussed in this part. The performance of each codec is estimated, in the network models depending on the QoS.

4.1 Codec G.711

This simulation is completed for the G.711 codec in various networks. The results appear below are used to assess the performance of G.711. It is analyzed from figure 5 that the value of Mean Opinion Score (MOS) is 3.7 in wired, wireless and WiMAX models, showing the good quality of speech. Comparatively, MOS value for UMTS model is 2.5 showing the worst quality of speech amongst all the models. Jitter and end to end delay in figures 6 and 7 shows that UMTS and WiMAX models undergo a delay in packets and attain some jitter. In UMTS a jitter and significant amount of delay are attained degrading the quality while in WiMAX, delay, and packet loss effect the communication. Traffic sent is almost same in all the models as shown in figure 8 while the traffic received in figure 9 shows that in WiMAX and UMTS there is the loss of the packets as compared to wired and WLAN networks. Jitter, delay and less reception of packets in UMTS model represent that it gives the worst quality of voice while using G.711. The performance of WiMAX models is also not effective as there is a delay and it loses packets. Wired and WLAN models give the best performance while using G.711.

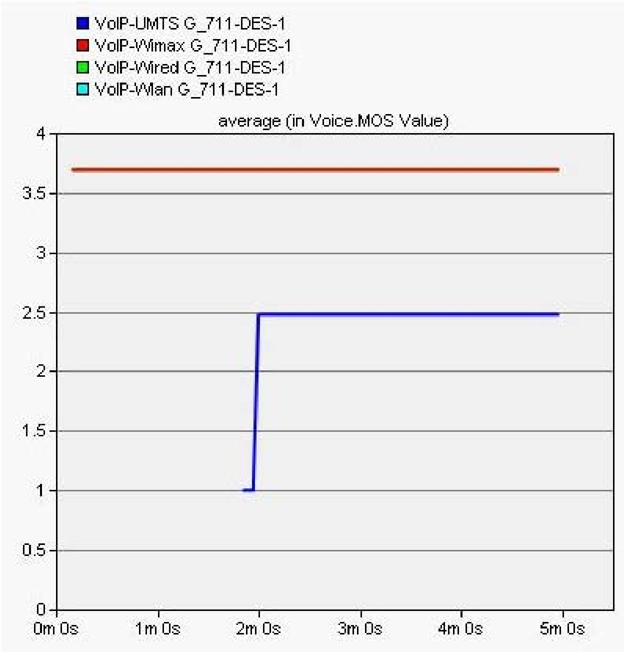

Figure 5: MOS for G.711

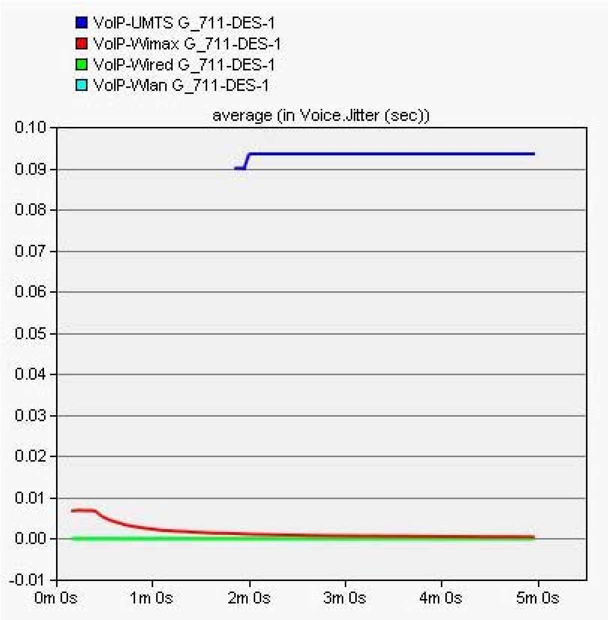

Figure 6: Jitter for G.711

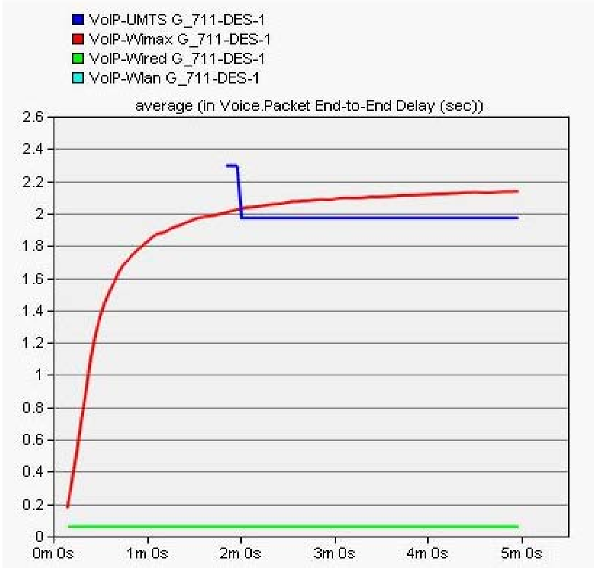

Figure 7: Packet end to end delay for G.711

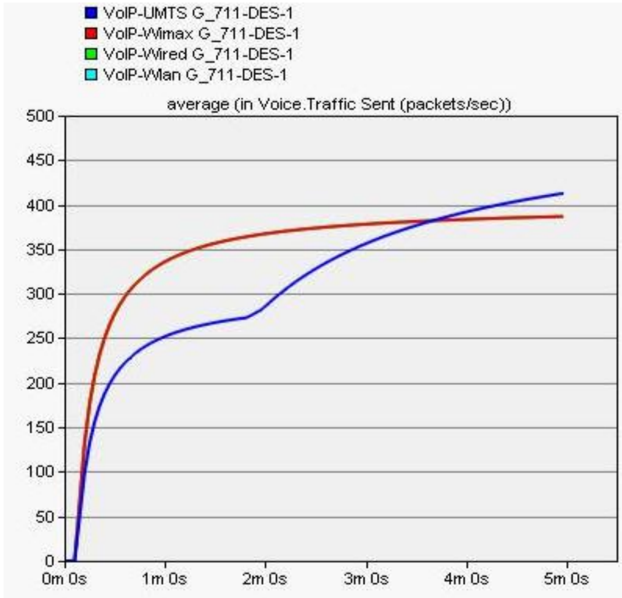

Figure 8: Traffic sent for G.711

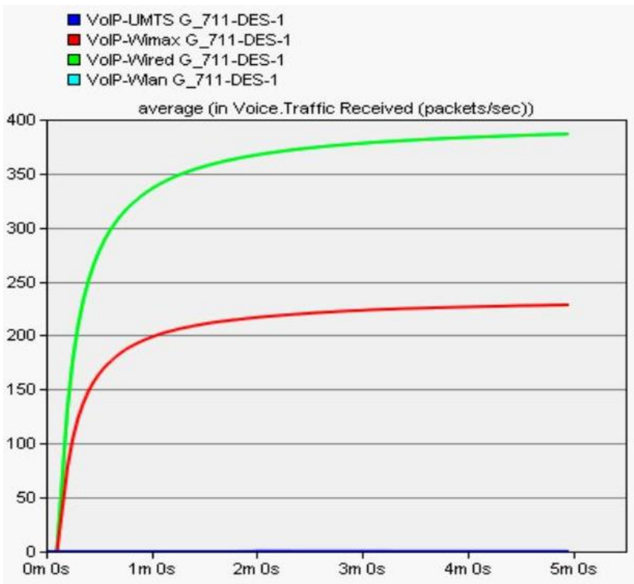

Figure 9: Traffic Received for G.711

4.2 Codec G.723.1

This simulation is performed for the G.723.1 codec in different networks. The results shown in figure 10-14 are used to evaluate the performance of G.723.1. Figure 10 show that the value of MOS is 2.5 in wired, WLAN and WiMAX models. Comparatively, MOS value for UMTS model is 2 showing the bad quality of speech. However, when it comes to jitter and end to end delay in figures 11 and 12, WLAN and UMTS models along with the wired model have minimum lateness in packets and attain zero jitter. In WiMAX jitter and the amount of delay is very small hence providing the good quality of VoIP. Traffic sent and received is almost same in all the models except WiMAX model which loses a few packets as appeared in figures 13 and 14. Jitter, delay and full reception of packets in wired, WLAN and UMTS models allow using the G.723.1 codec with low MOS. The performance of wired, WLAN, WiMAX and UMTS models is not quite effective but the jitter and delay are reduced using the G.723.1 codec.

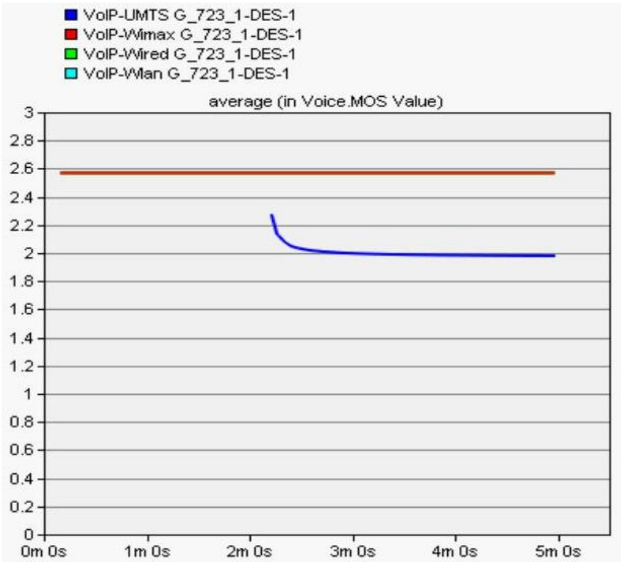

Figure 10: MOS for G.723.1

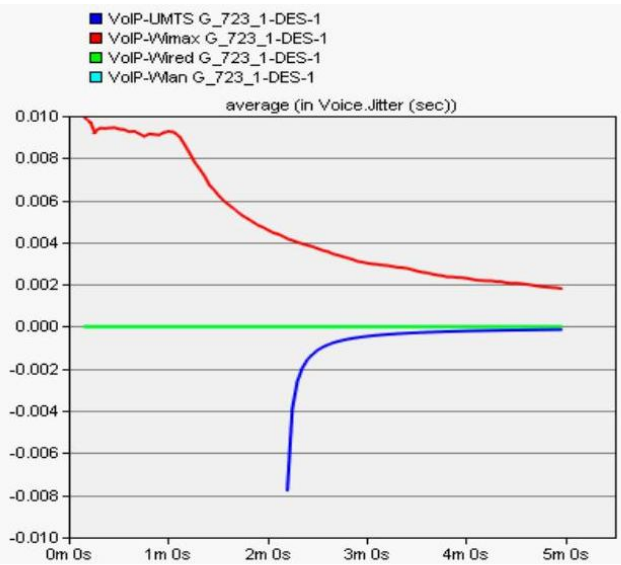

Figure 11: Jitter for G.723.1

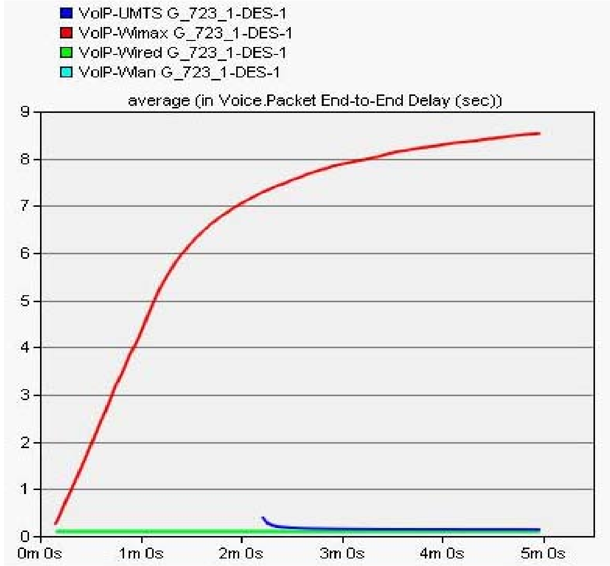

Figure 12: Packet end to end delay for G.723.1

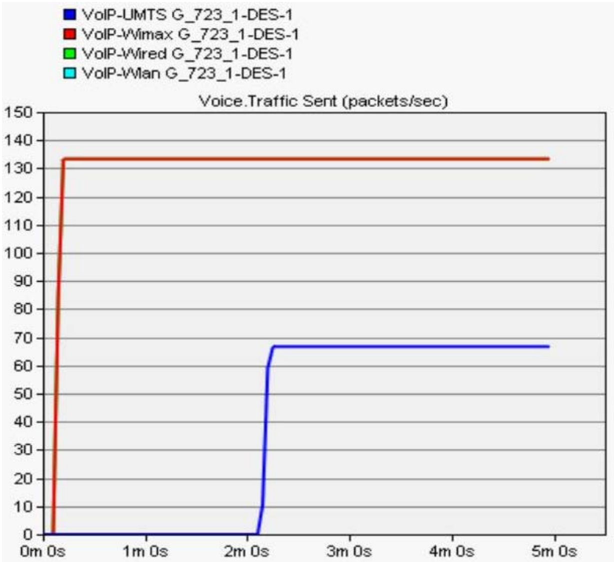

Figure 13: Traffic sent for G.723.1

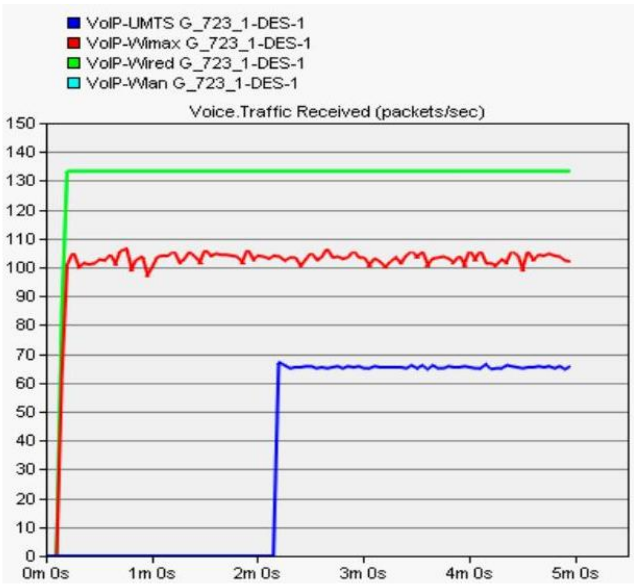

Figure 14: Traffic Received for G.723.1

4.3 Codec G.729A

This simulation is performed for G.729A codec in different networks. The results shown in figure 15-19 are used to evaluate the performance of G.729A. Figure 15 show that the value of MOS is 3.2 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 16 and 17 shows 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 same in wired and WiMAX models while there is some loss of packets in WLAN and UMTS networks as shown in figure 23 and 24. Jitter, delay and full reception of packets in WiMAX model represent that it gives the best quality of voice while using G.729A. The performance of WLAN and UMTS models is not effective as there is a delay and packets are lost.

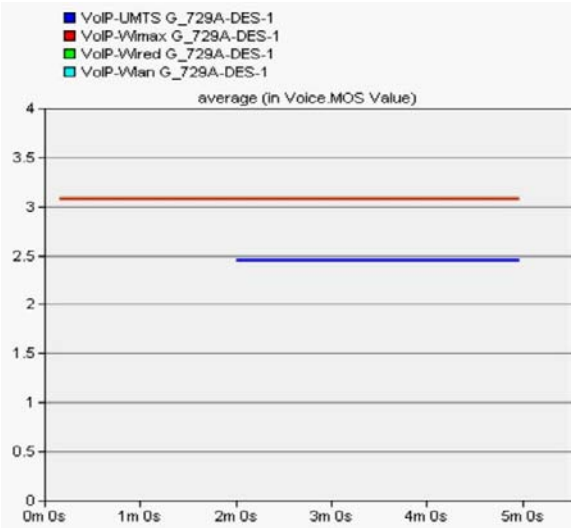

Figure 15: MOS for G.729A

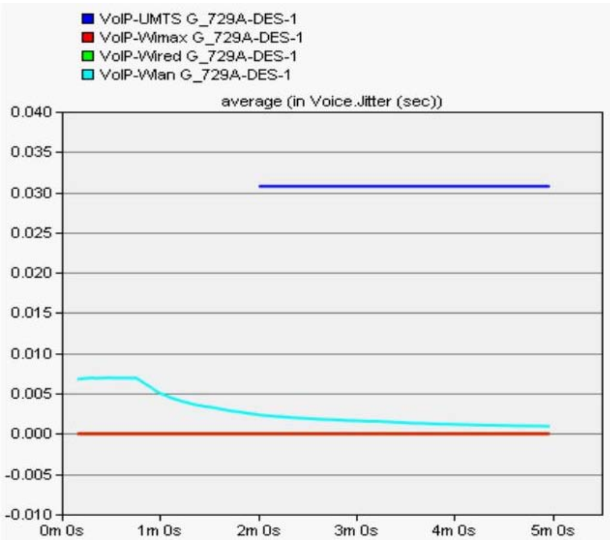

Figure 16: Jitter for.729A

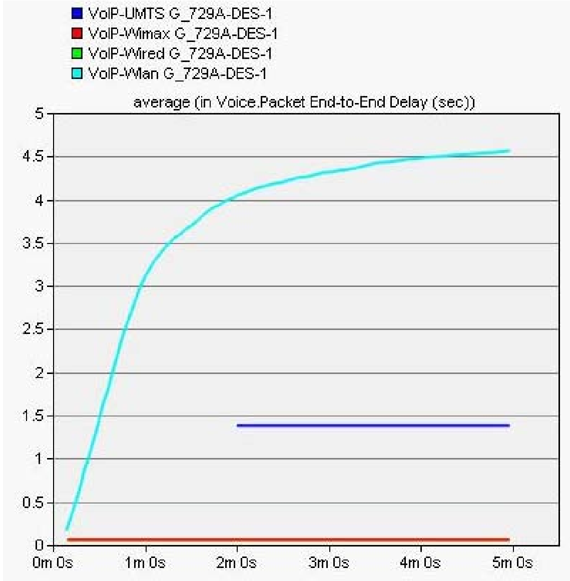

Figure 17: Packet end to end delay for G.729A

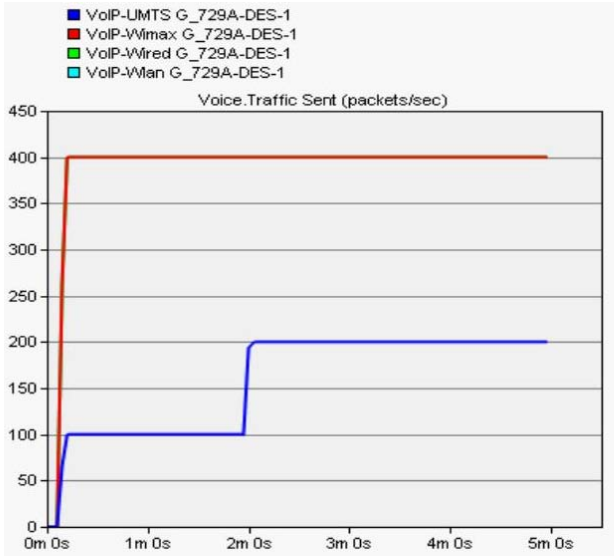

Figure 18: Traffic sent for G.729A

5. Conclusion

The performance of several VoIP codecs in different networks is analyzed using the OPNET Modeler. A variety of simulations is carried out to get the most effective and efficient results. On the basis of results achieved, the conclusion for the selection of VoIP codecs in different networks is made. Depending on the results it is concluded that wired network performs well irrespective of the VoIP codec being used. G.711 and GSM-FR can be selected for VoIP communications in WLAN network. For WiMAX network, G.729A codec is the most effective one. The quality of G.723.1 codec is observed low as it is a low- quality codec. Hence it can be used in all the networks depending on the environment and users density.

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