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Performance Evaluation of G.711 VIOP Coder Using Wired and Wireless Communication Network

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Abstract: ***Voice over Internet Protocol (VoIP) has been an interesting topic of research in the last decade. The engrossing increase in the use of VoIP services is resulting in the enormous growth of broadband network. The main objective of this paper isto evaluate the performance of G.711 VIOPcoder for different 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 like end-to-end delay, MOS, throughput and jitter. The VoIP codecs used in the measurements of QoS are: G.711. Simulations showed that G.711 isthe best schemes that provide high quality of voice in Wireless Local Area Network (WLAN) communicationsThe 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: VoIP; WLAN; WIMAX; UMTS; Codec; QoS

1. Introduction

Voice over Internet Protocol (VoIP) practices is potentially mounting day by day resulting in the demand of rapid improvements in the networks. There is a demand of decreasing the difference between the qualities of voice and increasing the available bandwidth to provide the best VoIP services comparative to the traditional circuit switched telephony [1]. VoIP has almost replaced the conventional Public Switched Telephone Network (PSTN) due to its cost effectiveness and the features being provided [2]. The wired Internet Protocol (IP) networks provide better VoIP services as compared to the wireless IP network as wireless networks have their own characteristics and impairments [3]. The unsolved issues caused by the wireless network in this area still needs some dedicated work spotlighting 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 delay 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.

This paper is to provide good quality of VoIP services in every network and analysis is done using different codecs mentioned in table1.VoIP packets are analyzed focusing all the major parameters like end-to-end delay, MOS, throughput and jitter over Wired, Wireless, UMTS and WiMAX networks using the OPNET Modeler.

2. VoIP and Codecs

The demand for mobile and broadband services is rising day by day. The last decade has seen the ever-increasing VoIP users with the demand of reliable and quality services. VoIP is an emerging technology for voice communication used these days. The services are not only being used for long distance calls but also for the short distant communications. The devices like IP phones and the VoIP enabled desktop systems are cost effective and also 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.

2.1 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 research work are listed in the table I below

Table 1: Characteristics of VoIP G.711 coder

codec	Coding algorithm	Sampling rate
G.711	PCM	64 kbps

3. 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 osses/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 codecs. The reason for utilizing this modeling method is to allow performance of the codecs to be analyzed in an improved manner.

3.1 Wired

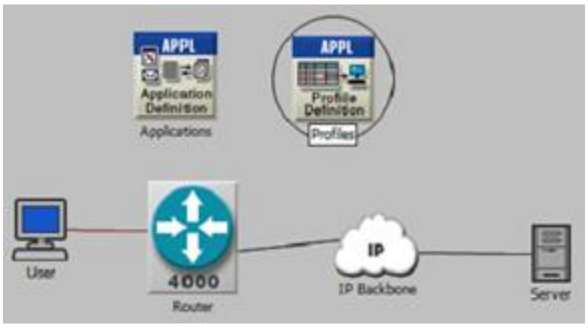

Figure 1: Wired Model

3.2 WLAN(wireless)

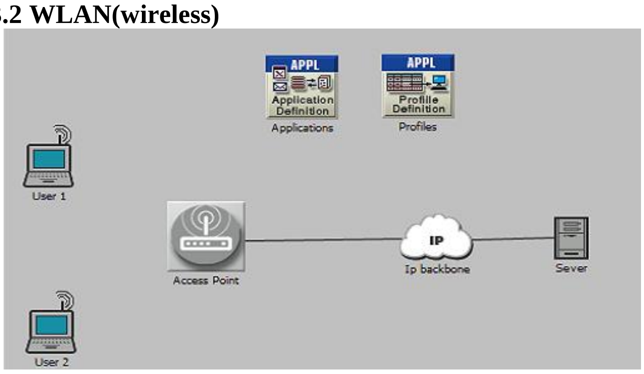

Figure 2: WLAN Model

3.3 UMTS

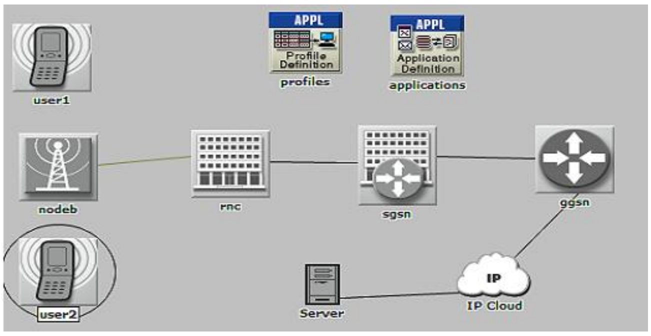

Figure 3: UMTS Model

3.4 WiMAX

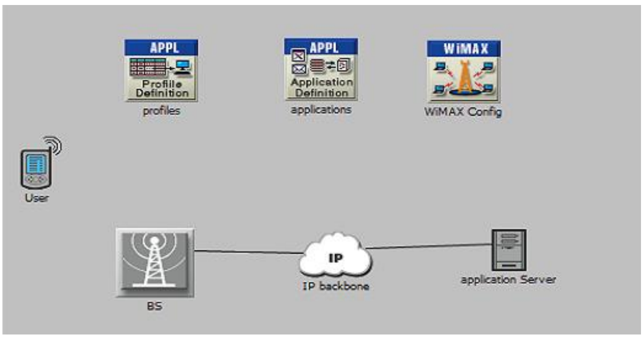

Figure 4: Wimax Model

4. Results and Analysis

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

4.1 Analysis of Codec G.711

This simulation is performed for G.711 codec in different networks. The results shown below are used to evaluate 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 is 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 loss in the packets as compared to wired and WLAN networks. Jitter, delay and less reception of packets in UMTS model represent that it gives 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 best performance while using G.711.

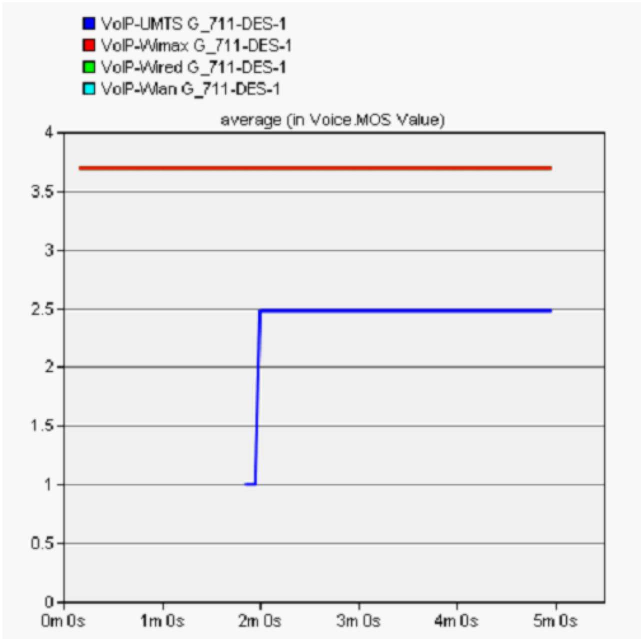

Figure 5: MOS for G.711

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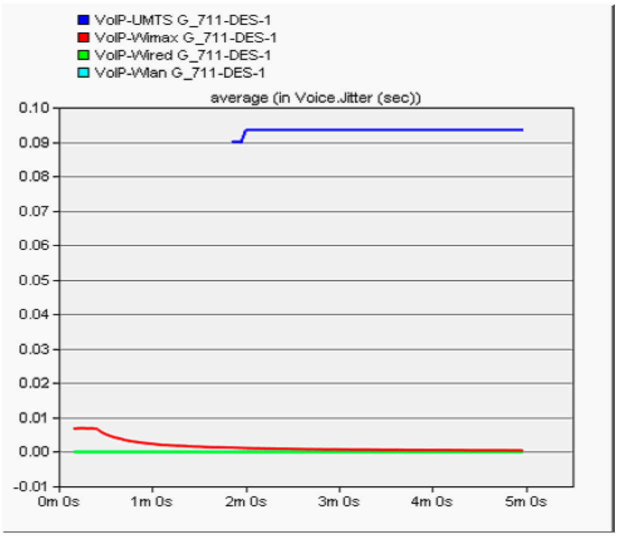

Figure 6: Jitter for G.711

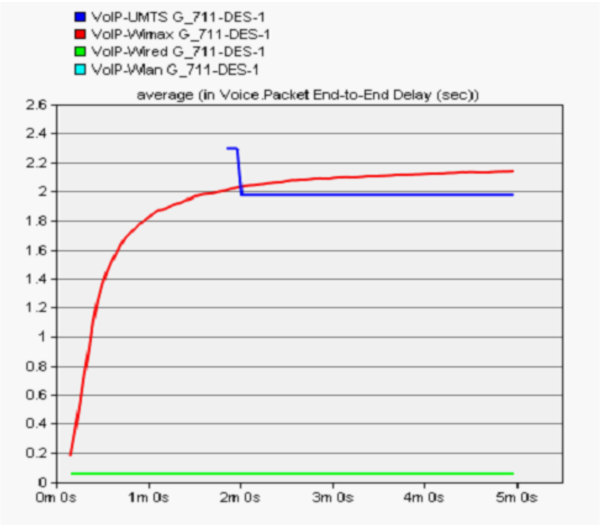

Figure 7: Packet end to end delay for G.711

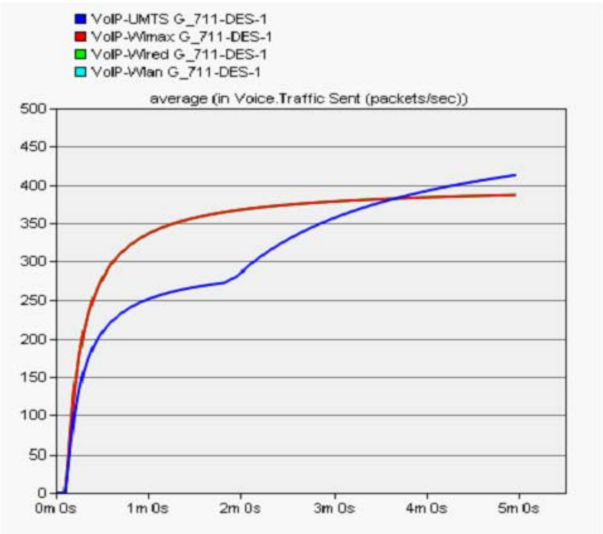

Figure 8: Traffic sent for G.711

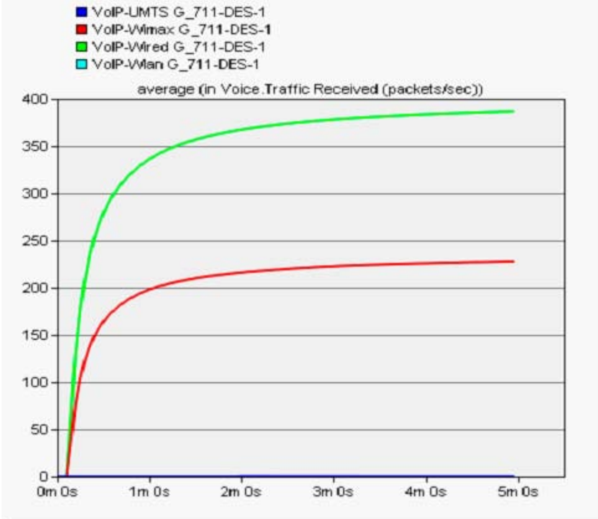

Figure 9: Traffic Received for G.711

5. Conclusion

Performance of VoIP codecs using G.711 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 of results attained, conclusion for the selection of VoIP codecs in different networks is made. Depending on the results it isconcluded that wired network performs well irrespective of the VoIP codec being used. G.711 can be selected for VoIP communications in WLAN network.

References

[1] T. Kwok, "Residential broadband Internet services and applications requirements" Communications Magazine, IEEE Volume 35, Issue 6, June 1997 Page(s):76 - 83
[2] T. Wallingford, "Switching to VoIP", Publisher: O'Reilly, ISBN: 0-596-00868-6, Pub Date: June 2005
[3] A. Samukic, “UMTS Universal Mobile Telecommunications System Development of Standards for the Third Generation“, 1998
[4] IEEE Std 802.11, “Wireless LAN Medium Access Control(MAC) and Physical Layer (PHY) Specifications,” ANSI/IEEE Std 802.11, 1999.
[5] Per Beming, Lars Frid, Göran Hall, Peter Malm, Thomas Noren, Magnus Olsson and Göran Rune, “LTE-SAE architecture and performance,” Ericsson Review No. 3, 2007
[6] Naveed Iqbal, Ajmal Khan, Malik Ahsan Ali, UzmaAnwar, BurhanUllah, M. FaizanSabir, ” Performance Analysis of Soft and Hard Handovers based on UMTS QoS Traffic Classes,” ICIIT, 2010.