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Voice over IP Codec at WiMAX Network (Analysis & Performance)

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ABSTRACT

WiMAX is a technology that comes up with multiple solutions. It's defined as Worldwide Interoperability for Microwave Access. It's based on IEEE 802.16 standard and is a Wireless Metropolitan Area Network (MAN) technology. It's designed to cover wide geographical areas serving a large number of users at low-cost and provides a wireless alternative to wired backhaul and last-mile deployments. This paper, will study setup of WiMAX using Voice over Internet Protocol (VoIP) applications to analyse and evaluate the performance of VoIP codecs using OPNET simulation. Relevant metrics is used to analyze the performance of the network and the ability to support VoIP calls. The main aim of this paper is to analyze the performance of VoIP codecs over the fixed WiMAX network to improve the quality of VoIP calls by using VoIP codecs such as (G.711, G.723, G.726, G.728, and G.729) in WiMAX network.

Keywords— Fixed WiMAX; VoIP Application; VoIP Codecs; OPNET.

I. INTRODUCTION

Along with the development of mobile communication and broadband technology, WiMAX has become a hot spot for global telecom operators and manufacturers. In 1998, a working group named 802.16 was formed by the Institute of Electrical and Electronics Engineers (IEEE), and their responsibility is to develop the specifications of broadband wireless access technology [1].

WiMAX is a telecommunications technology that provides wireless transmission of data using a variety of transmission modes such as radio spectrum to transmit tens of megabits per second in the bandwidth of (audio, data and video) from point to multipoint. It is a wireless industry coalition dedicated to the advancement of IEEE 802.16 standards for broadband wireless access (BWA) networks [2]. This technology addresses broadband wireless metropolitan access networks (MANs) that uses a point to multipoint architecture. The standard defines the use of bandwidth between the licensed 10GHz and 66GHz and between the 2GHz and 11GHz (licensed and unlicensed) frequency ranges [3].

The WiMAX provides worldwide area coverage and quality of service capabilities for application ranging from real-time, delay – sensitive voice over IP (VoIP) to real-time streaming download, ensuring that the subscribers obtained the performance they expect for all types of communications. It is a wireless digital communications system (IEEE 802.16) that it is intended for wireless "metropolitan area networks".

VoIP stands for Voice over Internet Protocol, a variety of methods for establishing two-way multi-media communications over the Internet or other IP-based packet switched networks. Although VoIP systems are capable of some unique functions (example: video conferencing, instant messaging, and multicasting), this appendix concentrates on the ways in which VoIP can use to replicate the voice conversation functionality of the public switched telephone network (PSTN). Voice over IP allows users to speak to each other using the Internet or an intranet as the transport network.

Today's telecommunication operators have seen the opportunities and the threats of VoIP thus, they implemented the VoIP gateways to allow communication between a PC and any telephone connected to a PSTN. Some operators even offer a transcending to VoIP for normal voice calls at a reduced cost [4]. VoIP developed at some a commercial institution and universities to diminish the cost of long-distance calls, for the price of decreased speech quality. Noticeably, the Internet has changed the way of thinking in telephony networks and will play an even more important role in the future. It is clear that the internet technologies will also influence the architecture and implementation of future mobile networks [5]. As a real-time digital application, VoIP requires a transmission system that includes low delay, jitter and packet loss rates, to ensure that the Quality of Service (QoS) is acceptable [4].

II. FIXED WIMAX

The IEEE 802.16d standard is also known as fixed WiMAX, which is shown in Figure 1. The fixed WiMAX supports only the application. It is very robust against multi-path propagation because it uses Orthogonal Frequency Division Multiplexing (OFDM). In OFDM, the subcarriers are selected in such a way that they are all orthogonal to each other, which decreases overlapping subcarrier channels.

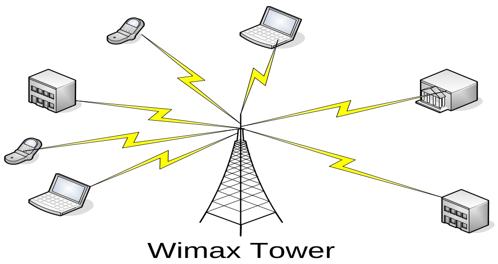

Figure.1. Fixed WiMAX

Fixed WiMAX can work as a point-to-multipoint with the transmission data rate of 1 Mbps to 75 Mbps at a transmission distance over 50km. It operates in 3.5 GHz and 5.8 GHz spectrum bands [6].The security mechanisms within the IEEE 802.16 standards are sufficient for fixed WiMAX access service [7].

III. VOIP APPLICATION

VoIP is a methodology and group of technologies for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet. Other terms commonly associated with VoIP are IP telephony, Internet telephony, broadband telephony, and broadband phone service. The primary advantages of moving voice over a data network increased efficiency and decrease cost. VoIP becomes available on smart phones and personal computers. VoIP applications used to carry voice over the Internet, also the cost is less compared to traditional telephone lines [8].VoIP application uses different varieties of signalling protocols such as H.323.

H.323 is a standard by the International Telecommunication Union-Telecommunication Standardization Sector (ITU-T) [9]. for transmission of real-time audio, video and data over packet switched networks. It specifies the components, protocols and procedures for multimedia application. The main goal of H.323 signalling protocol is to communicate and operate with other multimedia networks.

IV. VOIP CODECS

The CODEC is the analoge-to-digital-to-analoge converter. Although most of them are standardized, VoIP vendors implement proprietary codecs too. The type of Codec used is an important factor that affects the VoIP call quality, where the higher the compression, the lesser the size of data to be transmitted to the other side. Correspondingly, used codecs such as G.711, G.726, G.728, G.729 and G.723. Table [1] showing the description of VoIP codec for ITU standard:-

Table [1]: Description of VoIP codec

ITU Standard	Description
G.711	PCM
G.726	ADPCM
G.728	LD-CELP
G.729	CS-ACELP
G.723	Multi-rate CELP

V. BRIEFING OF VOIP CODECS

The codec is a technical term for the following variations, which essentially mean the same thing: compression-decompression / compressor-decompressor / Code Decode. In this paper, the researcher study several important codecs such as:-

G.711: It is an ITU (International Telecommunication Union) standard for narrow band speech codecs where it encodes the data into the stream of 8 bits at the rate of 8 kHz sampling rate, and with high bit rate, the audio codec of 64 kbps from the ITU [10].This codec is used for quality of voice calls. It employs logarithmic compression that compresses each 16-bit sample to 8-bit sample. It can be used for fax communication over IP networks [11].

G.723: It is an ITU-T standard codec was originally developed for videophones that delivers video and speech over regular phone lines (PSTN). It is a dual-rate speech codec designed for the ITU-T H.323 and H.324 audio and video conferencing/telephony standards. For the low bit rate, it uses the algorithm called as Algebraic Code Excited Linear Prediction (ACELP) [12].This codec is widely used in applications such as audio, video, fax and speech also used in VoIP applications.

G.726: It is an ITU-T standard codec. It is a waveform speech coder which uses Adaptive differential pulse-code modulation (ADPCM) having four bits rate: 40, 32, 24 or 16 kbps standard covering the transmission of voice. It was introduced to supersede both G.721, which covered ADPCM at 32 Kbit/s, and G.723, which described ADPCM for 24 and 40 Kbit/s. G.726 also introduced new 16 Kbit/s rate. The four-bit rates associated with G.726 are often referred to by the bit size of a sample, which are 2, 3, 4, and 5-bits respectively [13].

G.728: It is an ITU-T standard codec that operates at 16 kbps and is widely used for applications that requires a very low algorithmic delay. It's based on Low Delay Codebook Excited Linear Prediction (LDCELP) compression technique [14].

G.729: It is an ITU-T standard codec, Based on the Code Excited Linear Prediction (CELP) coding model. The G.729 codec delivers toll quality speech, similar in quality to 32-kbps ADPCM but at one quarter lower the bit rate. With the low rate of 8 kbps, G.729 is the lowest bit rate ITU-T standard with toll quality, offering opportunities for significant increases in bandwidth utilization in existing telephony and wireless applications. This codec is based on the Conjugate-Structure Algebraic Code Excited Linear Prediction (CS-ACELP) algorithm [12].

VI. METHODOLOGY

In this paper, the researcher will use OPNET modeller 14.5 simulators [15] which has a support of fixed WiMAX, to analyze the performance of VoIP codec under a given scenario. Figure 2 shows the planning of Network topology for simulation. Designed scenario below uses VoIP application server connected with two fixed WiMAX. Any WiMAX serves has many types of users like (offices, organizations, commercial institutes, privet mobiles and laptops). The total number of users in this experimentation is thirty tow users. The scenario this paper carry out several steps:

First, design and configuring WiMAX network Simulation and implementing VoIP call on the Simulation model. Second, analyze voice call quality over WiMAX network by comparing simulation results of codecs against each other. Finally, shows the results in the discussion.

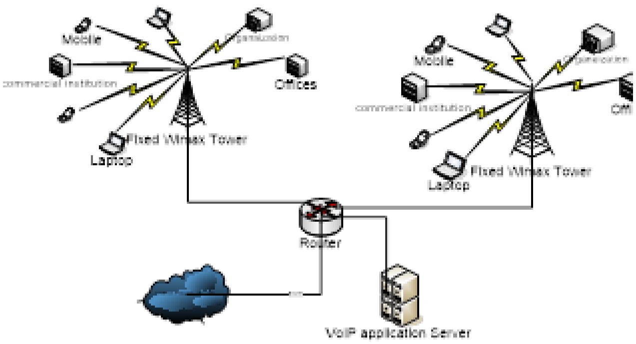

Figure 2. Planning Network topology

VII. *NETWORK DESIGN FOR SIMULATION*

OPNET modeller 14.5 simulators will use to analyze codecs to decide what the best performance of it, by using a VoIP application over WiMAX technology. Jitter, MOS, packet delay variation, packet end to end delay, and traffic received and traffic sent were also identified to find the most suitable codecs that will be used in comparison with each other. Figure.3 shows the network structure of:-

- Server for implementing communication over the Internet
- Router to connect fixed WiMAX antenna
- Two fixed WiMAX antenna
- Various groups of users to apply VoIP via WiMAX technology.

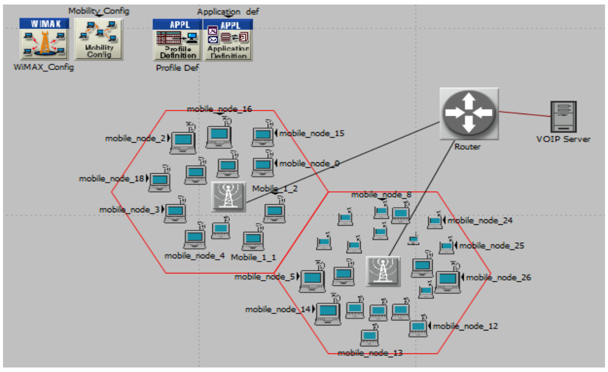

Figure.3Network Structure topology using OPNET for simulation

VIII. *SIMULATION: RESULT AND DISCUSSIONS*

A. *VOIP CODECS DISCUSSIONS AND RESULTS*

The result after analyzing VoIP codecs, shown in figure [4], are the comparative results of Jitter, MOS, packet delay variation, and packet end to end delay for codecs that are used in this experiment.

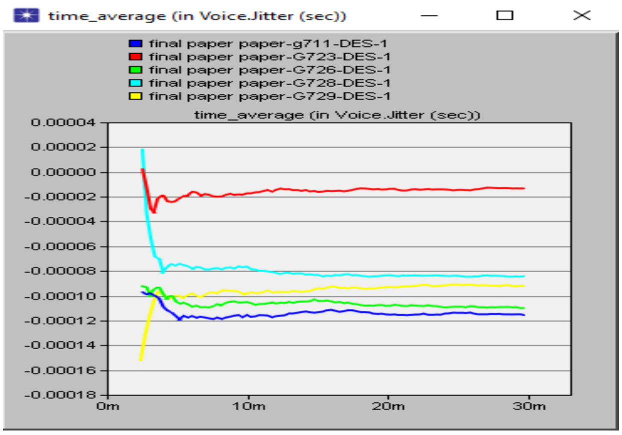

Figure. [4-1] Jitter (sec) value

Codecs Key	G.711, G.723, G.726,G.728, G.729

Jitter is the variance of packet arrival time. It defines the mean deviation of the packet spacing change between the sender and the receiver.

Fig 4-1 shows the comparative result of jitter for codecs that are used in this experiment. It can be seen that G.723 codec scheme has a large value of jitter variation of 0.0000133 but the G.711 codec scheme has a smaller value variation of -0.0001150. That means G.711 is a better codec. The negative value of jitter means that the time is the difference between the packets and the destination are less than the source.

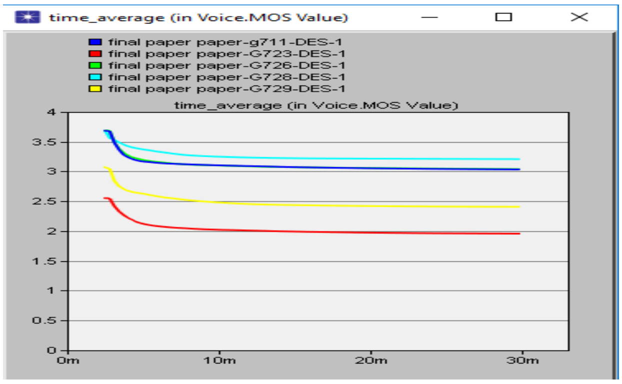

Figure. 4-2: MOS Value

Codecs Key:	G.711, G.723, G.726,G.728, G.729

Mean Opinion Score (MOS) is a common benchmark used to decide the quality of speech. Figure 4-2 shows comparative of Mean Opinion Score (MOS). Mean Opinion Score gives VoIP tests a number value as an indication of the perceived quality of receiving voice after being transmitted and compressed using codecs. The result G.728 codec has highest MOS variation of 3.209157, and the lowest MOS is G.723 codec variation of 1. 958112. That means the G.728 is a better codec.

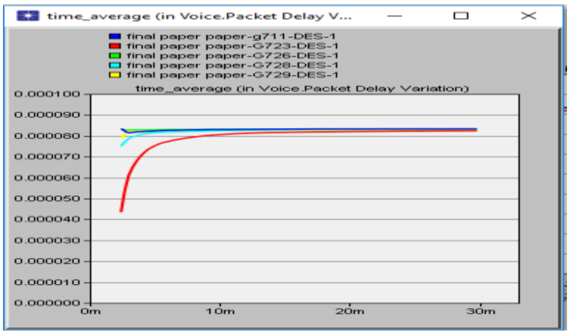

Figure. 4-3: Packet Delay Variation

Codecs Key:	G.711, G.723, G.726,G.728, G.729

Packet delay variation plays a crucial role in the network performance degradation and affects the users- perceptual quality. Figure 4-3 shows comparative results of packet delay variation, it can be seen from the graph and table [1] that the Packet delay variation is very high in the case of G.711 and the expected value of 0.0000833 Sec. G.723 codec has the lowest among the five codecs with the expected value 0.0000824 Sec.

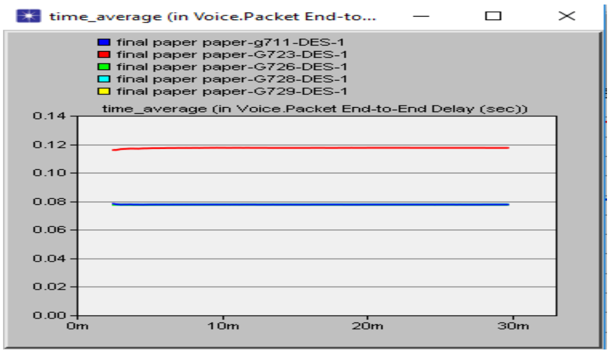

Figure. [4-4] Packet End-to-End Delay (sec)

Codecs Key:	G.711, G.723, G.726,G.728, G.729

Packet end-to-end delay in a VoIP network is caused by propagation delay and processing delay. Propagation delay is a characteristic of the transmission medium. Figure 4-4 shows comparative results Packet End-to-End Delay; it can be seen from the graph that G.723 codec scheme has the biggest value of 0.1174 but G.729 has the smallest value of 0.0775. That means G.729 is a better codec. Tables [3] bellow also showing the results of VoIP codecs.

Table [2] Result of VoIP Codecs

QoS	Jitter	MOS	Packet delay variation	Packet End-to-End Delay
G.711	-0.0001150 Low	3.035485	0.0000833 High	0.0777
G.723	-0.0000133 High	1.958112 -Low	0.0000824 Low	0.1174 High
G.726	-0.0001100	3.042107	0.0000832	0.0776
G.728	-0.0000845	3.209157 -High	0.0000832	0.0778
G.729	-0.0000922	2.409106	0.0000828	0.0775 Low

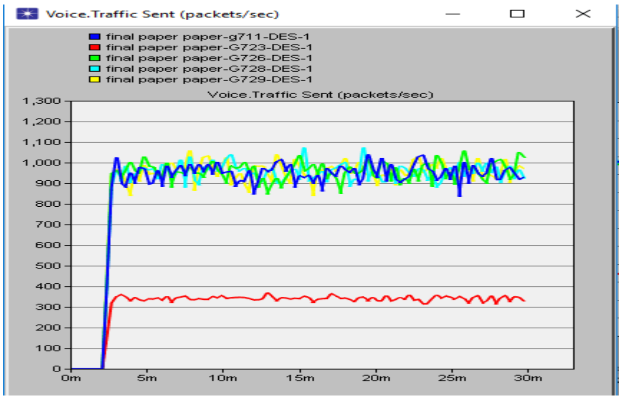

Figure. 4-5-a: Traffic by Second (sec)

Codecs Key:	G.711, G.723, G.726,G.728, G.729

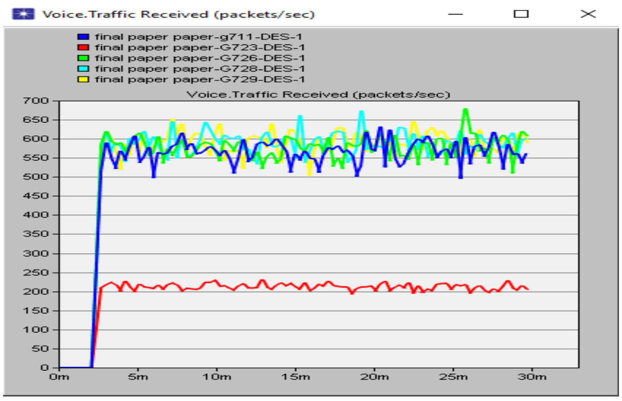

Fig. [4-5-b] Traffic received (sec)

Codecs Key:	G.711, G.723, G.726,G.728, G.729

Figure 4-5-a and Figure 4-5-b showing comparative results of Traffic send and traffic received. From the above graphs, calculating the differences between the traffic send and traffic received for all codecs will show what the best codecs are. The best codecs have lowest packet losses. It can be seen that G.728, G.729 and G.711 codec scheme in the sequence has a lowest packet lost value. That means G.728, G.729 and G.711 is a better codec of VOIP than G.723 and G.726.

B. WiMAX Discussions and Result

Delay or latency represents the time taken with a bit of data to reach from source to destination across the network. The main sources of delay can be categorized into propagation delay, source processing delay, Queuing delay, transmission delay and destination processing delay.

Fig 5-1 shows comparative results of WiMAX delay, it can be seen that the G.711 codec scheme has the highest value of 0.01685 but G.723 has the smallest value of 0.01661. But all delays haven't exceeded 0.1 Sec. From above results showing all delays it's less than 0.01Sec. That means WiMAX can provide better VoIP services.

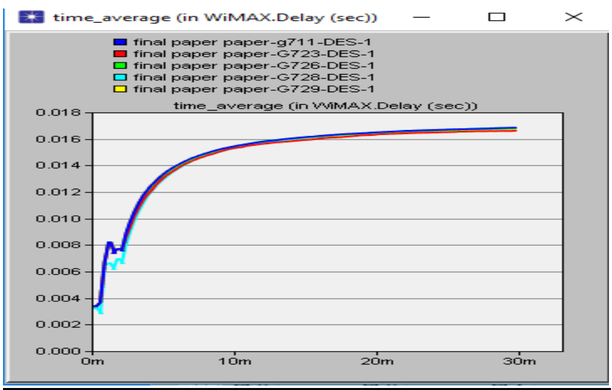

Figure 5-1: WiMAX Delay (sec)

Codecs Key:	G.711, G.723, G.726,G.728, G.729

Figure.5-2 shows comparative results of WiMAX Load, it can be seen that G.726 has highest value variation of 876.015556 packet/Sec. It Also G.723 has a lower value variation of 312.685 packet/Sec.

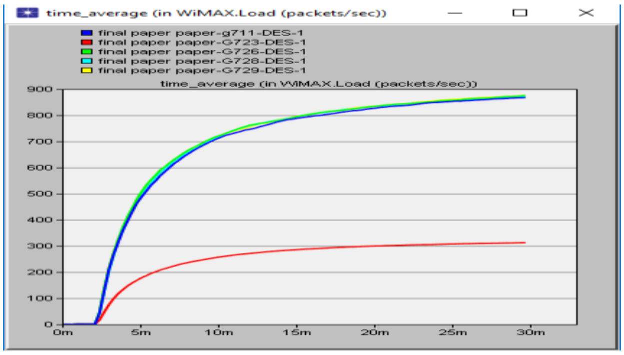

Figure 5-2: WiMAX Load (packet/sec)

Codecs Key:	G.711, G.723, G.726,G.728, G.729

Figure 5-3 shows comparative results of WiMAX throughput. When analyzing codecs using throughput measured by packet/Sec, it can be seen that the G.729 and G.728 have the highest throughput. It has value variation of 540.06 and 539.113333 packet/Sec. But the G.723 has lowest throughput value variation of 195.655556 packets/Sec. Table [3] bellow also shows the results of WiMAX.

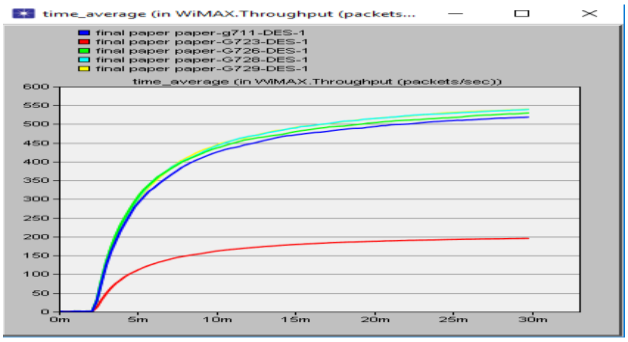

Figure 5-3: WiMAX Throughput (packet/sec)

Table 3. Result of WiMAX

Codecs Key:	G.711, G.723, G.726,G.728, G.729		
Qos	WiMAX Delay	WiMAX Load	WiMAX Throughput
Codecs	Delay (sec)	Load (Packet/sec)	Throughput (packet/sec)
G.711	0.01685- High	867.458333	518.469444
G.723	0.01661- Low	312.685- Low	195.655556- Low
G.726	0.01679	876.015556 High	529.72556
G.728	0.01675	871.278333	539.113333 High
G.729	0.01674	875.36	540.06- Highest

IX. CONCLUSION

This paper studied several important critical parameters that were analyzed such as: MOS, end-to-end delay, jitter and packet delay variation of VoIP. Also, analyzed delay load and throughput of WiMAX. The results obtained and indicated that VoIP codecs over WiMAX it has a high call quality. Simulation results also indicated that WiMAX can provide better VoIP services in terms of end-to-end packet delay. Also, the simulation indicated G.729 and G.728 have best codecs using for VoIP. The G.711 gives the better call quality for VoIP on the basis that it uses no compression at all, and as a result, the call quality sounds like using a regular ISDN phone. G.711 Codec is supported by most VoIP providers because it has a free license. G.729 and G.728 are considered to offer a good level of call quality at a low bit rate of kbps, which means that you would be able to get more calls through your bandwidth that if you were to use the G.711 Codec. G.729 and G.728 are supported by certain VoIP providers because the G.729 codec requires a product that does need a user to buy a license from the organization that re-sells licenses.

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