Performance Evaluation of voice Codecs in fixed and Mobile IPv6 networks

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Abstract- During the recent years, remarkable rise occurs into spreading of wireless networks since it offers flexibility, mobility and popularity due to the increase in mobile phones application users. Despite real time applications consume more bandwidth than others which is a big challenge for network designers. Newer VoIP applications start to use codecs that are used to compress and decompress speech signals in addition to coding and decoding. Codecs need to use high compression ratios for the VoIP implementation to generate maximum value of Quality of Service (QoS)to be Competitor to other traditional public switching networks. Also, Internet protocol version 6 is becoming more popular than version 4 since shortage of address space and additional features have been added to ipv6. In this paper, Different audio codecs G.711, G.729 and GSM that are widely utilized in VoIPv6 applications were studied and compared. These codecs were tested in WiMAX network through an IPv6 environment to detennine their perfonnance efficiency and find out the appropriate codec that suits VoIP transmission by observing various QoS parameters such as jitter, Delay, packet loss rate and Mean Opinion Score (MOS) value that affects the quality of voice

transmission. These transmissions had taken place while the nodes were stationary or moving with different speeds. All simulation scenarios are performed with OPNET modeller 14.5.

Index Terms- QoS, VoIP, G711 Codec, G729 Codec, GSM Codec, IPv6, WiMAX, Fixed Network, Mobile WiMAX Network

I. INTRODUCTION

Demand on voice transport increased via applications of internet protocol (VoIP) through the last two decades. This large demand refers to large decrease into costs of installation (set up) and operation of telecommunication via internet protocol compared to traditional telephone networks (PSTN). In addition, VoIP introduces huge and various group of services, such as Voice Call with low cost and the integration of data with possibility to combine technologies of voice, image and data spontaneously, also flexibility, graduation and simplicity of usage. These advantages make telecommunication via internet protocol more important into business field. Many VoIP applications such as Viber, telegram and WhatsApp messenger are available on the internet, which gives free, good quality calls [1, 2, 3]. VoIP signals utilize a combination of analog signals and digital signals. With VoIP, when you speak into a phone it's received as a digital signal. That digital signal is sent via the internet to the receiving phone. There, it is picked up and changed into an analog signal and your voice is heard. This may sound like a load of "who cares", but this small fact makes VoIP possible. Having your voice transmitted as a digital signal and then changed to an analog signal makes it possible for your VoIP phone to act like a normal phone. Meaning, if you call a landline, or cell phone, your phone will behave as any normal phone would. If VoIP phones were unable to change that digital file back into an analog file that would mean that VoIP phones could only communicate with other VoIP phones. Therefore, in order to make VoIP phone's practical, they need to be able to convert the digital signal into an analog signal [4]. Recently, the wireless communication technology has been evolved to meet the user's increasing demand of the network. [5] WiMAX wireless networks offer resilience to present real-time applications such as VoIP. Furthermore, it provides high data speed and large coverage distance as well. Therefore, WiMAX networks enable wireless connectivity on trains, buses and coverage to even difficult areas as large as cities. Consequently, WiMAX could be the perfect platform for VoIP application [6].

Decreases end-to-end delay and packet loss rate of acoustic packets should be maintained. As well as the jitter should be between the acoustic packets among an accepted limit. The jitter inhibits the sound quality and it makes calling Via internet protocol inconvenient for the user in many times, and buffers are utilized to hold obtained packets for short period before playing them on equal separated periods to decrease jitter [6,7]. VoIP services have to be equipped with mechanisms and algorithms to guarantee QoS in order to eliminate delay and packet loss.

VOIP advancement and applications are based on IP where progression in innovations could have more noteworthy affect in communications that are IP based as well as other related applications. When it comes to today's and future's voice communications, VOIP wi11 be the means of it due to its performance efficiency and cost efficiency as well. Since the voice communications are directly related to the IP based communication networks, the performance of VoIP applications will be affected by the aspects of IP and its linked orientations and characteristics. Due to the shortage of IPv4 address space, the researchers have been led to introduce new version, IPv6, which offers enough address space compared to IPv4.Added some features in the IPv6 header such as flow label field which improved The QoS of IP telephony, video/audio and interactive games applications are enhanced since the deployment of IPv6 is started. [8]. This paper is organized as follows: Section 2 introduces some previous work Section 3 describes VoIP Codecs. Section 4 shows Quality of service support in IPv6. Section 5 views **WiMAX** Networks Technology. Section 6 explains Methodology of building the scenario by OPNET. section 7 presents the results and some discussion Finally, Section 8 of this paper presents the main conclusions and summarizes the paper.

Research problem: Communication applications via internet protocol suffer some challenges such as packet lose rate and delay which will affect the quality of audio stream due to its sensitivity to packet delay and jitter, Delay might happen as result of different paths that are available for voice stream. Packets loss is also serious problem as it happens as result of queue overflow and congestion. Level of service quality goes down in VoIP session if there is no response for time requirements.

II. LITERATURE REVIEW

There have been studies focusing on the analyzing and evaluating the performance of audio codecs in wireless networks, concentrating on WiMAX networks.

The VoIP performance was analyzed and evaluated by the authors in [9] based on integrated wireless LAN / WAN in frame of voice codecs systems (G.711, G.723, G.729) the different parameters which indicate service quality were estimated and the network model was simulated by using OPNET modeler. The overall results indicated that codec G.729 offered great result into VoIP performance since G.729 has an accepted Mos value and is less deviation related to transmitted packets to received packets also End-to-End delay and voice jitter was lesser into codec G.729 compared to other two codecs indicated to it here. On the other hand, The above paper studied three audio codecs (G.7.11, G723,G.729) two of audio codecs (G711,G729) that are the same in this research however a new codec, GSM is not tested which will be used for Evaluation its performance in various QoS parameters that affect the quality of voice transmission , also in The above mentioned paper the transmissions taken place while the nodes are stationary through an IPv4 network. in this research the transmissions will take place while the nodes are stationary or moving with different speeds through an IPv6 network. In addition, In [1O] Authors studied the integration between wireless networks (Wi-Fi, WiMAX) in the field of VoIP codecs performance evaluation. Design of network scenario WiMAX / Wi-Fi integrated by using QualNet Program for application of communication via internet protocol. This integrated network has a dual radio interface for Wi-Fi and WiMAX network the service quality of voice was studied into this paper via for voice different codecs flows like G.726, G.711 and G.729 through service quality parameters measurements like average delay- average jitter - average Mos value which was used to evaluate codec performance. The results in this scientific paper indicated that codec 729 has the best performance compared to (G 711 - G 726) under heavy load. For delay, jitter, Mos value and packet loss. This paper also indicated that bandwidth requirements less for codec G 729. Therefore, this codec is mostly used in VoIP applications in the networks in which the preservation of bandwidth is crucial thing. The study in [1O] simulate the VoIP performance evaluation Via WiMAX network. Since it used different parameters such as Jitter, Mos, packet sent and received and packet delay for measuring of voice transmission performance Via WiMAX technology. Then it simulated three voice systems G.729, G.723 and G.711 in order to find out the appropriate codec to transmit voice Via WiMAX network the results showed the simulation into this paper that communication via internet protocol performs better under code G.711 compared to G. 723 and G.729. Research results also showed that VoIP applications might presented in better way under exponential traffic distribution. studied three audio codecs (G.7.11, G726, G.729) at least two of audio codecs (G71 l, G729) that are the same in this research however a new codec, GSM is not tested which will be used for Evaluation its performance in various QoS parameters that affect the quality of voice transmission. In addition, the transmissions taken place while the nodes are stationary through an IPv4 network is designed using QualNet 5.0.2. In this research the transmissions will take place while the nodes are stationary or moving with different speeds through an IPv6 network The experiments are done by means of simulation using OPNET.

In VoIPv4 , the performance of selected audio codecs that are widely used IPv4 has been studied [10]. Furthermore, G.711, G.723.1 and G.729A codecs were selected to study the VoIPv6. The performance efficiency of the IPv6 based voice communication network was studied by applying selected audio codecs to monitor various QoS parameters. G.711 showed better results in terms of vibration performance, with no random perturbation. However, the results showed good performance in the VoIPv6 scenarios, although further technical improvement is needed. Different parameters were taken in consideration to evaluate the performance of VoIP over WiMAX networks (Jitter, Mos value, packet end-to-end delay and packet lose ratio) [11]. The simulation result showed better performance for the G.711 codec as compared to the G.723 and G.729 codecs. Furthermore, under the exponential traffic distribution, the VoIP applications can perform better. The performance of WiMAX for supporting VoIP traffic for stationary nodes was evaluated under different parameters such as Mos, end-to-end delay and Jitter [12]. The simulation results showed that the average Mos was changed based on the distance between Base Station and nodes and the transmission power.

The above-mentioned papers study several audio codecs such as G.7.11, G723, and G.729. A new audio codec called GSM will be studied and evaluated in this paper. Moreover, the transmissions of the previous studies take place while the nodes are stationary through an IPv4 network. In this paper, the transmissions will take place while the nodes are stationary or moving with different speeds through an IPv6 network.

Ill.VOIIP CODEC

A term codec stands for coder-decoder, which basically transforms analog audio signals into compressed digital form and compresses digital into an uncompressed analog audio signal. The codec techniques have various functions and requirements. They are classified based on the implemented compression, data rate, and sampling rate they perform in different behaviors under different conditions, a codec with high bandwidth will give better voice quality Since every codec produces a specific quality of speech, the codec is chosen based on network conditions and its requirements. This section discusses the most popular and commonly used codecs on a large scale of voice communications Networks [13, 14].

A. G*.711 Codec*

G.711 is an ITU-T standard for audio commanding. It is principally utilized in telephony that provide toll-quality audio at 64 Kbit/s. In 1972, the standard, Pulse code modulation (PCM), was released for utilization. It is required in many technologies, ex. fax communication over IP networks [15]. Due to the algorithm simplest, G.711 requires lowest processing power and results very low processing overheads. Most G711 softphones are free (Like Xlite), the call quality sounds like using a regular ISDN phone. This Codec is supported by most VoIP providers [16,17]. The G.711 codec details are as shown in the table (1).

B. G.*729 Codec*

G.729 is formally defined as Coding of speech at 8 Kbit/s using code-excited linear prediction speech coding (CS-ACELP). G.729 is gives a good quality of audio stream at low bit rate, 8 Kbps, which allowing more calls than the G.711 Codec gives. In contrast, the G.729 Codec implementation requires more CPU processing time to transmit the audio stream. Therefore, many VoIP applications, that handle one call at a time, required to deal with G.729 due to its processing time. In addition, to use the G.729 codec, a licensed product is required [18]. The G.729 codec details are as shown in the table (1).

C. *GSMCodec*

GSM (Global System for Mobile communications) is the immediate telecommunication for mobile phones in Europe. It was the first digital speech coding standard used in GSM digital mobile phone systems [19] [20]. The coder has a bit rate of 13.2 kbps with an encoding frame length of 20ms and with sample size of 33 bytes. GSM-FR has flexibility which provides lots of advantages such as improving speech quality in both full-rate and half-rate modes. A speech quality and capacity are smoothly traded by adapting the channel and codec mode. The codec does not come ladened with a licensing requirement and offers outstanding performance with respect to the requirement of the CPU space [21].

Basic characteristic of standard codecs are illustrated in Table 1 explains compare CODECs with each other to get insight knowledge [22] [21].

Table 1: Codecs Types

"ORCodes & Bit Rate (Kbps)	Algorithm	Sample-rate (KHz)	Frame Size (ms)	Codec Sample Size (Bytes)	Codec Sample Interval (ms)	Voice payload Size (Bytes)	Packet Per Second (PPS)
G.711 (64Kbps)	Pulse code modulation (PCM)		Sampling 8	80	10	120	50
G.719 (8Kbps)	Conjugate Structure-Algebraic Code Excited Linear Prediction (ACELP)		10	10	10	20	50
GSM (13.2 kpbs)	Regular Pulse excitation long -Term prediction RPE-LTP	8	22.5	33	20	66	50

IV. QUALITY OF SERVICE (QOS) SUPPORT IN IPV6:

The IPv6 header includes a new field, Flow Label, which supports QoS. The instructions of QoS are included in the IPv6 packet header thus the queuing delay is reduced efficiently. In addition, the packet can be encrypted without QoS being affected which gives possibility to send streaming video or audio across the Internet, such as IPSec [23].

A Flow Label is uniquely specified by combination of the source address and of a non-zero flow label. This label is employed to keep up the sequential flow of the packets belonging to the identical stream. There are various benefits of recognizing the stream flow. The identical stream packets must be sent via same path with same flow label to avoid re-sorting of packets. [23] [24].

Flows are usually specified by an integration of sender and receiver addresses with their ports. However, in order for the sender and receiver ports to be determined, TCP or UDP headers must be looked into by the router. Essentially, many extension headers have to be tracked by an IPV6 router to find a transport-layer header. As a result, the complexity of the forwarding process will be increased and will affect the router performance. The packets of identical stream are handled in the same fashion. The IPv6 routers must handle the packets belonging to the same flow in a similar fashion. However, the information of the IPv6 data packets may be specified within the data packets themselves or it may be conveyed by a control protocol [25,26].

A flow can be recognized uniquely by sharing the sender address and non-zero flow label.

2. Several active flows can be occurred between the sender and the receiver along with different traffic.
3. if a packet is not belonging to a flow, the flow label is equal to 0.
4. The traffic should be sent with the same sender and receiver address and the non-zero flow label [24].

The main advantages of utilizing the flow label field are as follows [26]:

- The end-to-end delay is reduced.
- The problems caused by changing routing bath are reduced.
- The QoS mechanism is supported.

V. AN OVERVIEW OF WiMAX NETWORKS TECHNOLOGY

WiMAX stands for "World Interoperability for Microwave Access". It is normally a standard based on universal interoperability, including IEEE 802. l 6d-2004 for static, and 802. l 6e for mobile high-speed data. WiMAX is known as a wireless technology that delivers carrier-class, high speed wireless broadband at a much lower cost. Furthermore, it covers a large distance than WiFi technology and higher speeds over greater distances for a greater number of users. WiMAX can provide service in areas that are difficult for wire networks to do and overcome the physical limitations of traditional wire networks as well. It can handle high-quality of video and voice services. WiMAX can support access remotely up to 50 km for fixed stations and (5 - 15 km) for mobile stations [27, 28]. The description of standard IEEE 802.16 is as shown Table 2 [29].

Table 3: WiMAX QoS classes

QoS Class	Applications	Qos Specification parameters
UGS	Constant bit rate services such as VoIP	Maximum sustained rate ,Maximum latency tolerance, Jitter tolerance
rtPS	Streaming Audio or Video	Minimum reserved rate , Maximum sustained rate ,Traffic priority
ErtPS	Voice with activity detection	Minimum reserved rate , Maximum sustained rate, Jitter tolerance,

		Maximum latency tolerance, Traffic priority
nrtPS	File Transfer Protocol(FTP)	Minimum reserved rate , Maximum sustained rate, Traffic priority
BE	Data Transfer, Web Browsing	Maximum sustained rate, Traffic priority

A. WiMAX Quality of service architecture

It's the network's ability to provide good services for the client based on the requirements of different applications used by controlling certain network parameters is known as Quality of Service (QoS). With the increasing number of real time applications over the Internet, QoS is regarded as part of the network design and infrastructure. Currently, for multimedia applications, the appropriate solution should be considered to quickly address the QoS demand [30],[31]. QoS's essential objective is to guarantee the requirements of network parameters such as latency, packet loss rate, Jitter, and bandwidth.

Moreover, Priority could be provided to the same data stream or dedicated real time traffic bandwidth while preserving the usual data on the other side properly. Table 3 lists the scheduling service types in the QoS application of the WiMAX standard [30].

VI. METHODOLOGY OF BUILDING THE SCENARIO

The scenarios were performed and implemented in OPNET (Network simulator). It's the preferred tool due to its global recognition in terms of its consistency and reliability in producing results that are very close to actual scenarios. Each scenario consists of fixed scenario and Mobile scenarios. Both scenarios consist of 7 WiMAX cells; the length of each cell is 10000 meters. The cells are operating in the 802.16e standard; each cell has one Base station connected to the IP backbone cloud with PPP DS3 link (rate 44 Mbps).The network contains the WiMAX client (Mobile_l_l) Which initially will be used as a fixed node then will be used as a mobile node at a rate of IOm/s, 20m/s and 30 m/s. the IP Backbone (Core network) is connected to an IP Router (with PPP DS3) link then to an Ethernet Switch using 100Base-T link then the switch is connected to an Ethernet Workstation (VoIP Client) also using 100

Base-T Connection. The WiMAX configuration that has been done is shown in figure **(1).**

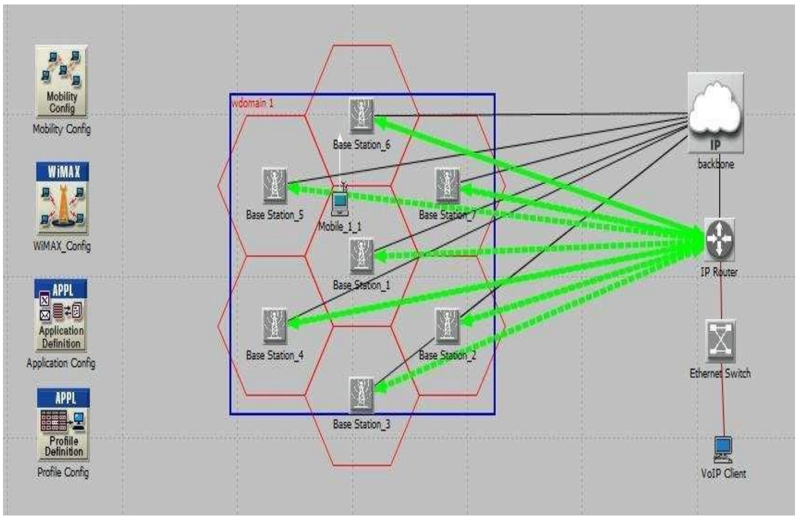


Figure 1: WiMAX Network Model

VII. SIMULATION RESULTS AND DISCUSSION

The following results gathered after collecting statistics by observing various QoS parameters.

A. Scenario: WiMAX Fixed Network

This section demonstrates results of comparison for three voice codec schemes (G.711, G.729, GSM) By using metrics of service quality (Packet Loss, E2E delay, Jitter, Mos Value) in fixed WiMAX network.

1. Packet Loss Performance

Figure 2 represents average of the sent and received traffic under various audio codecs. Cyan line and the Blue indicate to the data sent and received, respectively for audio codec G.711. Yellow line and the Red indicate to the data sent and received, respectively for audio codec G.729. Pink line and the green indicate to the data sent and received, respectively for GSM codec. To be more an efficiency network these two traffics must be equal as seen in figure 2 to all of the three-codec schemes to both voice traffic sent and received are identical to some and the packet loss is 0.0%.

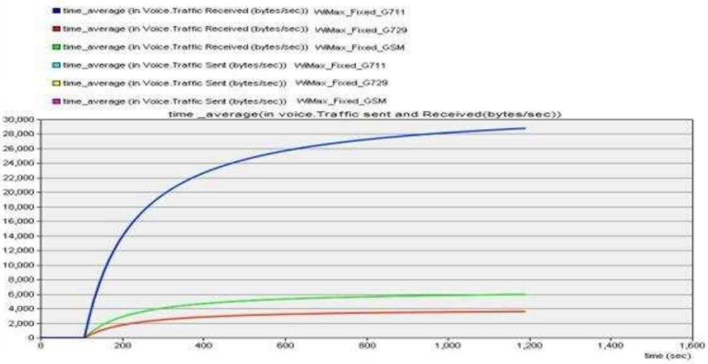


Figure 2: Average traffic sent and received

ii). End to End (ETE) Delay Performance

Figure 3 represent average packet ETE delay under various audio codecs. Packet end-to-end delay is considered as the most important performance metric in VoIP. G711 and G.729 presents the best performance and maintain a lower ETE delay level respectively on average. While GSM codec scheme yields the highest voice packet end-to-end delay.

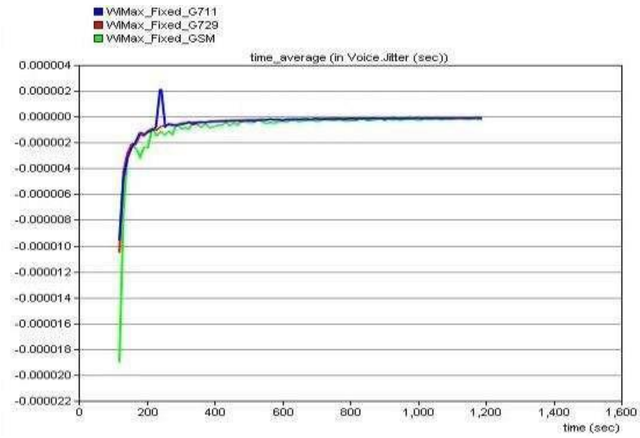


Figure 3: Average E-to-E Delay

2. Jitter Performance

Using different codecs, the average voice Jitter comparison is described in Figure 4. Through looking at average voice Jitter, sharply rose is noticed for Jitter with some negative delay variation which means that at the source, the time difference between the packets at the destination is less. Then settled to the end of the simulation and closed around O ms in average under various codecs that means that at fixed node configuration, the jitter effect is so small and the quality of the received voice traffic is so good.

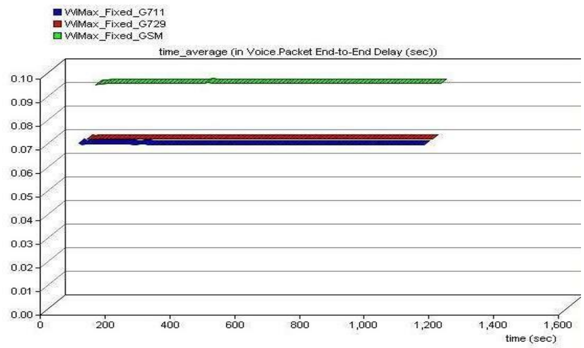


Figure 4 : average Voice jitter

3. MOS Value Performance

Figure 5 demonstrates the average Mos value under various audio codecs. The Mos is QoS metric used in VoIP applications that describe the voice perception quality. X and Y-axis illustrates the simulation time in seconds and the average Mos value in a scale defined of l to 5 respectively where (Excellent =5; Good=4; Fair=3; Poor=2; Bad=1). The observation shows stability of the average Mos value at most of simulation period, that indicated that G.711 had higher Mos value from GSM and G729 respectively.

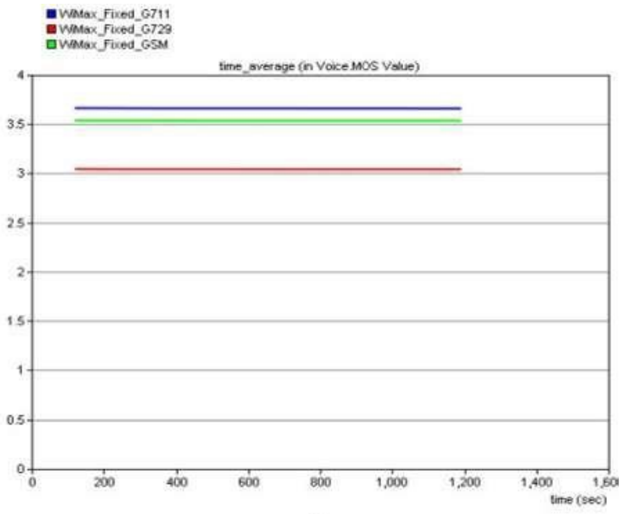


Figure 5: Average voice Mos

The findings obtained from the simulation results of three voice codec schemes over Fixed WiMAX Network under. Comparison of QoS parameters are depicted in table 4.

Table 4: Results for a Fixed WiMAX Network (Summary)

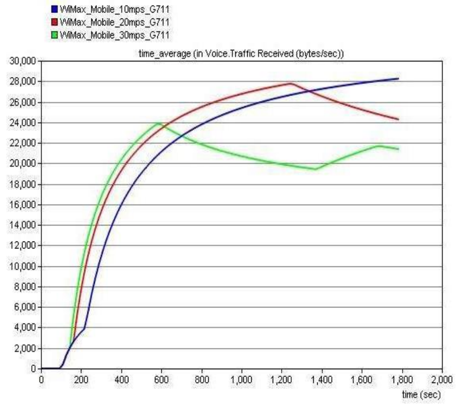
[illegible]	Fixed WiMax Network			
	Packet Loss	**ETE delay**	**Jitter**	**Mos**
G.711	**0%**	**70ms**	**0 ms**	**3.65**
G.729	**0%**	**71ms**	**0 ms**	**3.53**
GSM	**0%**	**90ms**	**0 ms**	**3.04**

B. Scenario: Mobile WiMAX Network

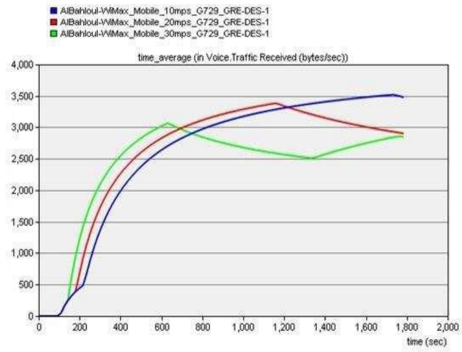
This section demonstrates results of comparison for three voice codec schemes (G.711, G.729, GSM) and the extent of mobility speed effect on (Packet Loss, ETE delay, Jitter, Mos Value) in Mobile WiMAX network. The mobile node is configured to move at a speed of 10rn/s, 20m/s, 30m/s.

1. Packet Loss Performance

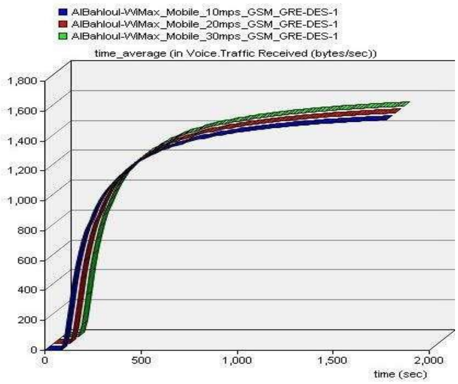
Figure 6 (a, b and c) show the voice traffic received in mobility state which were implemented under various audio codecs (G711, G729, and GSM) respectively with different speeds. The lines (Blue, red and green) indicate to the nodes with different mobility speeds (10, 20 and 30) mis respectively in all graphs. Through looking at curves
under different speeds for G.711 and G.729 codecs, As the speed of node movement increases, the average of voice traffic receive is decreased and vice versa. In the case of GSM codec, we find maintains the stability of the level of performance, and almost identical of the traffic receive ratio in all speeds as shown figure 6 (c).


WiMax_Mobile_10mps_G711
WiMax_Mobile_20mps_G711
WiMax_Mobile_30mps_G711
time_average (in Voice.Traffic Received (bytes/sec))
30,000
28,000
26,000
24,000
22,000
20,000
18,000
16,000
14,000
12,000
10,000
8,000
6,000
4,000
2,000
0
0 200 400 600 800 1,000 1,200 1,400 1,600 1,800 2,000
time (sec)

(a)


AlBahloul-WiMax_Mobile_10mps_G729_GRE-DES-1
AlBahloul-WiMax_Mobile_20mps_G729_GRE-DES-1
AlBahloul-WiMax_Mobile_30mps_G729_GRE-DES-1
time_average (in Voice.Traffic Received (bytes/sec))
4,000
3,500
3,000
2,500
2,000
1,500
1,000
500
0
0 200 400 600 800 1,000 1,200 1,400 1,600 1,800 2,000
time (sec)

(b)


AlBahloul-WiMax_Mobile_10mps_GSM_GRE-DES-1
AlBahloul-WiMax_Mobile_20mps_GSM_GRE-DES-1
AlBahloul-WiMax_Mobile_30mps_GSM_GRE-DES-1
time_average (in Voice.Traffic Received (bytes/sec))
1,800
1,600
1,400
1,200
1,000
800
600
400
200
0
0 500 1,000 1,500 2,000
time (sec)

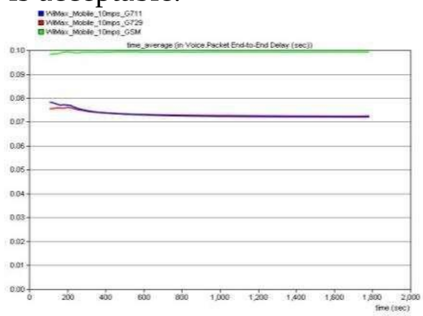
(c)

Figure 6 (a, b, c): Average voice traffic received under various audio codecs with different speeds in Mobile WiMAX

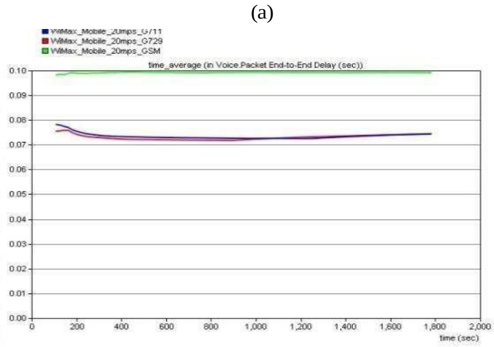
Figure 7 (a, b, c) shows Packet End-to-End Delay with different speeds (10, 20, and 30) mis respectively. To analysis of the performance based on ETE delay, the results have been compared with different results that have been obtained from different three mobile scenarios under various audio codecs.

As the figure 7(a, b, c) shown, the average voice packet ETE delay is almost stable which indeates that the effect of the increase in delay is so small or almost identical.

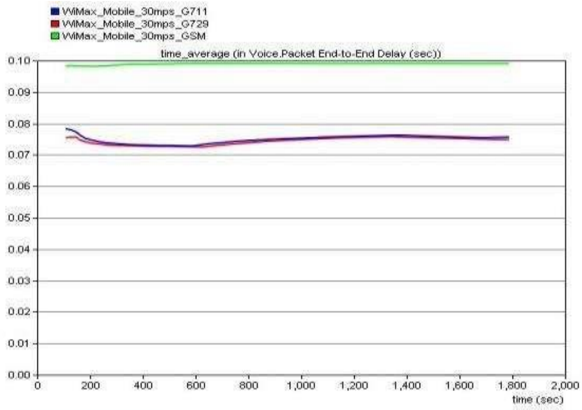
However, both 0711 and G.729 codec give best performance and maintain a lower ETE delay level in all speeds as compared to GSM. Therefore, Simulation result indicates that the coding schemes can provide VoIP services in terms of ETE packet delays where the delay less than 150 msec is acceptable.



(a)



(b)



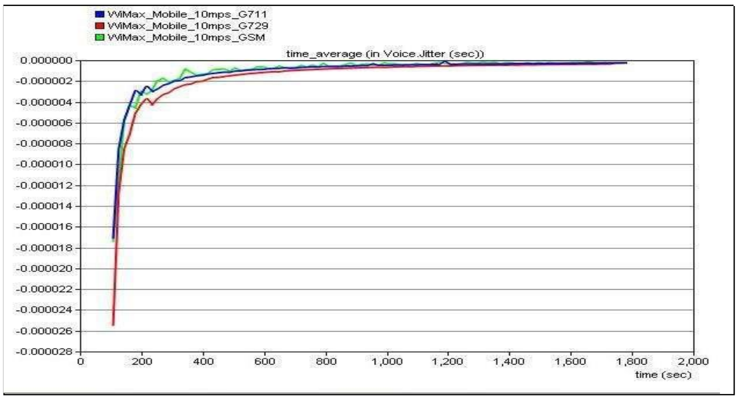
(C)

Figure 7 (a, b,c): Average packet ETE delay under various audio codecs with different speeds in Mobile WiMAX

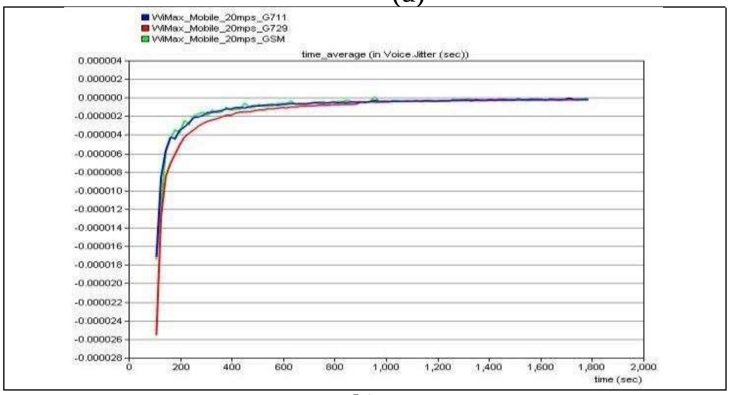
2. Jitter Performance

Figure 8 (a, b, c) shows average jitter delay of voice in different speeds (10, 20, and 30) mis respectively. To analysis of the performance based on jitter delay, the results have been compared with different results obtained from different three mobile scenarios under various audio codecs. the average jitter delay is rose sharply with some negative delay variation, then settled to the end of the simulation and closed nearly at O ms.

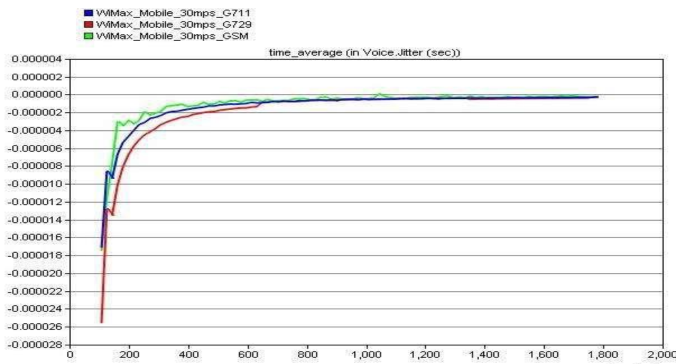
However, the results indicate similar performance for each codecs (0711, 0729, and OSM) which is stable and maintain low level in all speeds which is lower than The maximum allowable duration of jitter delay (40 ms).


WiMax_Mobile_10mps_G711
WiMax_Mobile_10mps_G729
WiMax_Mobile_10mps_GSM
time_average (in Voice.Jitter (sec))
time (sec)

(a)


WiMax_Mobile_20mps_G711
WiMax_Mobile_20mps_G729
WiMax_Mobile_20mps_GSM
time_average (in Voice.Jitter (sec))
time (sec)

(b)

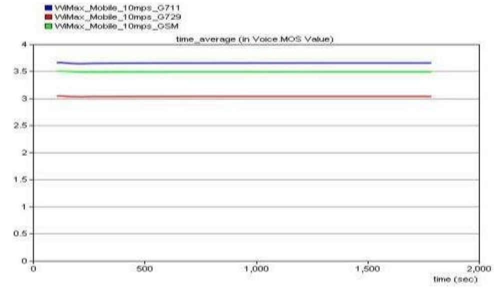

WiMax_Mobile_30mps_G711
WiMax_Mobile_30mps_G729
WiMax_Mobile_30mps_GSM
time_average (in Voice.Jitter (sec))

(C)

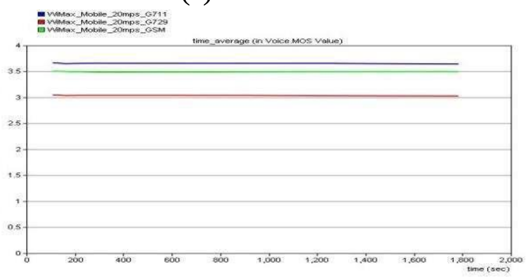
Figure 8 (a,b,c): average voice jitter under various audio codecs With different speeds in Mobile WiMAX

3. Mos Value Performance

Figure 9 (a,b,c) illustrates to average of Mos value under various speeds (10,20, 30) *mis* respectively. As mentioned before, the highest Mos value is the best performance codec. As seen in figure 9 (a, b, c), the average of Mos value is stable even though the node speed is increased. However, he G.711 codec has highest Mos value and offers the best performance for the applications of VoIP as compared to GSM and G.729 codec.



(a)



(b)

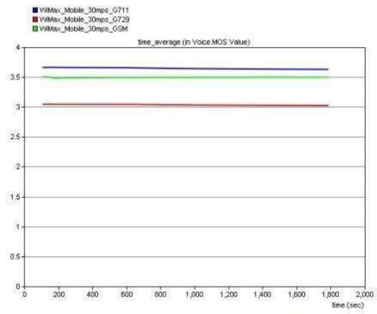


Figure 9 (a,b,c): average voiJci1:os value under various audio codecs With different speeds in Mobile WiMAX

CONCLUSION AND FUTURE WORK

In this paper, The QoS performance of VoIP traffic was analyzed and evaluated under various voice codecs G.711, G.729 and GSM for fixed and movement IPv6 Networks. The simulated results showed that VoIP for fixed network under G.711 and G.729 codec produces the best performance and maintain low ETE delay level. In contrast, all coding schemes provide a good performance and acceptable QoS without packet drop in all voice codes.

In concluded, the G.711 codec is the optimal codec and gives the best performance where the G.729 gives ETE delay and GSM shows better result in term of Mos due to its bit rate.

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