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Investigating the QoS of Voice over IP using WiMAX Access Networks in a Campus Network

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Abstract

VoIP is a very rapid evolving communication technology which supports transportation of voice data via IP based networks. In parallel, IEEE 802.16e standard based WiMAX is a new emerging access technology and the first generation of 4G broadband access wireless technology with an enhanced in-built quality of service (QoS) provision with many benefits including cost reduction, high quality as well as other value added network service solutions especially for communications Service Providers with emphasis on real time services. WiMAX promises manifold benefits in terms of optimal network performance across a long distance in contrast to other wireless technologies such as Wi-Fi and 3G cellular technologies. Hence, this research attempts to identify some of the network performance parameters that Service Providers will focus on to develop a VoIP over WiMAX communication tool that will serve as a voice communication broadband replacement technology to old circuit-switch voice communication. This study adopted a simulation-based network performance analysis to investigate the effects of the application of different voice encoder schemes on QoS of VoIP system deployed with IEEE 802.16e standard WiMAX network. Through different network simulation experiments using realistic network scenarios in OPNET environment, this research provided an in-depth network performance comparative analysis of VoIP over WiMAX using performance parameters which indicate QoS such as voice jitter, voice packet ETE delay, packet-sent-packet-received, WiMAX network delay, voice packet delay variation and throughput. The obtained simulation experiment results indicated that choice of suitable codec scheme can affect the QoS of VoIP traffic over WiMAX network. The results also indicated that the choice of suitable voice encoder scheme with a small number of voice frame-size per packet have a significant impact over VoIP traffic performance when deployed with WiMAX access technology.

Keywords: WiMAX, QoS, End-to-End delay, Jitter, IEEE 802.16e, PSTN, OPNET, Simulation, Wi-Fi, Codec

1. Introduction

VoIP as a communication technology supports transportation of voice data via Internet Protocol (IP) based networks. This communication technology seems to have edge over circuit-switched PSTN (Public Switched Telephone Network) (Alshomrani *et al*, 2012; Dudman & Backhouse, 2006); as a result of its effectiveness in voice transportation in the form of digital IP packets via the TCP/IP based Internet. This technology enables the transmission of telephone calls through Internet or Intranet as opposed to PSTN by sending packetized voice signal via Internet Protocol (IP). Voice over IP entails that VoIP is based upon IP; hence, the transmission technology is basically in digital form (Gibson & Wei, 2004), this enables the application of complex encoding and decoding algorithms called codecs to digitise and split up voice data into packets. International Telecommunication Union – the Telecommunication Division (ITU-T) has clearly defined most of the codec algorithms for use in compression/decompression of digitised voice packets such as ITU G.711, ITU G.722, ITU G.723, ITU G.726, ITU G.728, ITU G.729, etc. Many notable Internet Protocol-based data access networks namely Internet, Fiber optic, Ethernet or wireless technologies like Third Generation (3G), Worldwide Interoperability for Microwave Access (WiMAX), and Wireless Fidelity (Wi-Fi) could be used for Voice over IP deployment.

In parallel with VoIP, WiMAX as an emerging technology promises a lot in terms of performance across a long distance more than any other existing wireless technologies such as Wi-Fi and 3G cellular technologies. WiMAX Networks if properly designed, configured and implemented will yield data transmission speeds not lesser than 70mbps spanning across a metropolitan area (that is, distance of over 30 miles or even more) in contrast to Wi-Fi and 3G with data transmission speeds over a distance of about one hundred (100) feet or less and 2mbps to 3mbps over half a mile respectively.

Wireless access technologies such as IEEE 802.11 based wireless LAN (WLAN) (Crow *et al*. 2007) as well as

3G cellular networks (Inamura *et al*, 2003) widely in use by IP networks have been excessively demanded for real-time services like voice, video and numerous related multimedia applications. Sengupta, Chatterjee & Ganguly (2008) argues that the recent achievement of IEEE 802.16 standard in terms of performance for mobile WiMAX in the metropolitan areas has positioned it as the possible alternative solution to resolve the issue of excessive demand of the previous mentioned wireless access technologies like WLAN and 3G cellular networks. This WiMAX is an emerging access technology that offers wireless data transmission in several forms ranging from point-to-point links to mobile cellular access networking. WiMAX uses the fundamental principles of IEEE 802.16 standard which offers wireless broadband access as a substitute to our conventional Cable and DSL; hence, it uses mobile broadband data access networks to provide IP connectivity to the subscribing users (Alshomrani *et al*, 2012). These authors through their different researches have been able to show the greater potential of WiMAX Networks over other mentioned wireless technologies especially when deployed with VoIP application.

The quality of real-time applications like VoIP is commonly affected by several performance issues like voice packets end-to-end delays, voice packet delay variation and jitters (Haghani *et al*, 2008; Pentikousis *et al*, 2008) as they are easily noticed irrespective of their percentage occurrence during voice/video data transmission. The Quality of Service (QoS) features embedded in WiMAX enables it to provide optimum transmission and delivery of interactive and real-time voice and video packets in any properly designed and configured access network.

In the light of the above, a significant growth is been witnessed nowadays in the deployment of IEEE 802.16e standard based WiMAX Networks. Generally, voice over IP via WiMAX and broadband wireless appears very promising in the near future; hence, the several speculations from various schools of thought that VoIP over Wireless will soon totally overwrite the circuit-switched voice communication technology. Consequently, the deployment of VoIP over WiMAX technology demands that communication Service Providers should pay more strict attention into issues of Quality of Service with regard to voice, video, high speed data/access and mobility over an IP based networks. Hence, the dire need to intensively investigate, identify, analyse and evaluate the performance issues in this evolving technology by designing and simulating network scenarios using network simulator like OPNET Modeler in order to position VoIP over WiMAX as a voice replacement technology in the near future. Because VoIP over WiMAX is still in its very tender stage in the commercial market today, this research through the study of the simulated network scenarios, will attempt to discover the performance (or QoS) weaknesses associated with this emerging technology and thereby propose necessary network add-on performance parameters that will give VoIP over WiMAX an edge over circuit-switched voice communication technology.

2. Literature Review

There have been an increase in global demand for wireless data services as well as real-time applications like VoIP, audio and video streaming (Sengupta, Chatterjee & Ganguly, 2008); this increasing demands is as a result of rapid growth which has been massively witnessed in several wireless technologies recently. Countless researches are on-going in areas of wireless technologies deployment (especially WiMAX) using Voice over IP based network system, all in a bid to come up with a communication system that will be able to provide optimal wireless services so as to meet the increasing user demands. As self-reliant units, holons have a degree of independence and handle circumstances and problems on their particular levels of existence without reaching higher level holons for assistance. The self-reliant characteristic ensures that holons are stable, able to survive disturbances.

In their respective researches in (Salah, 2009) and (Yanfeng & Aiqun, 2006), the authors argue that it is necessary that the capabilities of a network to support VoIP applications be measured prior to its deployment with such network. According to them, the network's readiness to support deployment with VoIP system could be investigated by using network modeling and simulation approaches, measuring for voice packet end-to-end delay, voice packet delay variation, throughput and voice jitter after injecting real time (VoIP) traffic into the network. The author's argument if adhered to, will help in solving a great deal of problem as it will save both time and resources instead of just deploying real-time applications such as VoIP with just any wireless access technology without prior investigation of whether such network has any real-time application support capabilities or not.

With reference to Halepovic, Ghaderi & Williamson, (2009), VoIP system has become increasingly popular more than ever even as WiMAX Networks are been deployed in several countries across the globe. Hence, many researchers in recent years as well as currently have focused extensively on different features of VoIP services over WiMAX networks, all focused on investigating and identifying network add-on performance criteria that will enhance the quality of service delivery of VoIP system over WiMAX networks. In (Flizikowski, Majewski, & Przybyszewski, 2010), the authors have investigated to a remarkable extent the audio, data and video support features in WiMAX Networks. Their research was focused on examining the QoS deployment over WiMAX Networks and comparison of the performance achieved using WiMAX service classes like Unsolicited Grant Service (UGS) and Extended real time Polling Service (ertPS). The studies carried out by these authors have confirmed that WiMAX Networks supports real-time application more compared to other wireless access technologies like WLAN and 3G.

A traffic-aware scheduling algorithm for the deployment of VoIP applications over WiMAX Networks have been proposed in (Ansari & Haghani, 2008), the authors critically examined the performance of the proposed method in comparison with various notable conventional methods. They further explained how the efficiency of VoIP over WiMAX networks performance can be improved upon by the application of their proposed scheduling methods. But their proposed algorithm was not investigated under known performance metrics to ascertain and establish its robustness in QoS supports.

The authors in (Shrivastava & Vannithamby, 2009) maintain that though WiMAX Networks are efficient in supporting data traffic, the capacity of VoIP when deployed over IEEE 802.16e WiMAX system is not impressive as a result of overwhelming MAP overhead always generated by dynamic scheduling of VoIP traffic. In their work, they adopted persistent scheduling as a mechanism in IEEE 802.16e WiMAX system in order to minimise MAP overhead occurrence. The only deficiency inherent in their proposed persistent/group scheduling mechanism is that it creates sort of "resource hole" in the frame at the data allocation region which leads to inefficient resource allocation.

Majority of the VoIP QoS investigations have been conducted on Ethernet LAN, Wireless LAN as opposed to WiMAX access networks. In most of the occasions where it has actually been done with the deployment of WiMAX networks, the researchers have failed to look at some notable complex codec algorithms/schemes with reduced value of voice frame size per packet that can be applied on voice/video calls/conference to enhance the quality of VoIP performance when deployed over WiMAX Networks. These complex codec schemes provide tremendous compression efficiency that saves network bandwidth essentially in wireless technologies like WiMAX networks.

Moreover, majority of the researchers that have done some related work on investigation of VoIP system performance when deployed over WiMAX networks used other network simulators such as NS-2, E-modeling, NetSim, etc for designing, modeling and simulation of the network which most computer networking students and professional industrial practitioners are not quite very much conversant with. This research will use OPNET Network Simulator which is rather universal among IT students and communication professional to look at the performance criteria as well as some codec algorithms approved and specified by ITU-T for voice and video conferencing using VoIP over IEEE 802.16e standard WiMAX networks in a campus network.

2.1 VoIP compression Schemes

The quality of voice and video services in VoIP system deployed over WiMAX network are affected by many factors ranging from hardware, software, bandwidth, broadband connection and even the technology we use (Muntean, Otesteanu, & Muntean, 2010). These factors are all under our control as we can change, replace or even improve on them; but the VoIP system voice quality over wireless technologies such as Wi-Fi, WLAN, 3G and WiMAX are not under the users' control. Hence, this research intends to use network simulation methodology to identify the best data compression algorithms and other relevant network performance add-on parameters that will improve the QoS of VoIP services deployed over WiMAX network.

VoIP data compression according to Cignoni *et al* (2008) is a process through which voice data are rendered less unnecessarily large using compression software (also known as codecs) for easy transmission over IP-based networks as well as to enhance the voice quality upon reception. These codecs encode and transform the voice analog signals into digital data that are further compressed into much lighter packets which are therefore transferred over the Intranet or Internet. Decompression (or decoding) process is used at the destination point to decompress the packets and realise the original analog voice signals which the user (receiver) can hear. Given that analog voice and video signals cannot be transmitted over IP-based networks, they are encoded before transmission using codec schemes. The efficiency and overall quality of a chosen codec scheme therefore has a very high impact over the voice and video quality of VoIP conversation over WiMAX network. There are various codecs (compression software) for textual materials, fax, audio (voice) and video data. The most generally applied codecs for voice over IP networks as approved and specified by ITU-T include G.711, G.722, G.723, G.726, G.728, G.729 and many others.

These are also part of the issues that this study will try to resolve by using OPNET Network Simulator to investigate the necessary performance add-on criteria (QoS) with notable complex codec algorithms as listed above. We used OPNET Network Simulator in designing, modeling and simulating VoIP system over WiMAX network in realistic networking scenarios in order to provide a thorough investigation on how different complex codec algorithms with varying frame size per packet affect VoIP overall QoS when deployed over WiMAX Networks. Hence, the results of the above proposed network simulation experiments will enable us to suggest the best codecs and other relevant performance criteria that will make VoIP over WiMAX a viable replacement for circuit switched PSTN voice communication.

3. Simulation Setup

In carrying out the network performance evaluation of voice over IP using WiMAX access network, we designed and simulated close-to-real-life network scenarios to investigate the QoS of VoIP real-time applications deployed over IEEE 802.16e standard based WiMAX access technology in a campus network using OPNET Modeller 16.1 network simulator. The simulation only considered VoIP supported services and applications. Hence, background traffics such as http, e-mail and ftp are not considered in this simulation. Fig. 4.1 depicts the IEEE 802.16e standard based WiMAX network topology designed and used for this network research simulation. The WiMAX access network model is made up of four (4) cells with 0.3km radius and an Internet Protocol (IP) backbone. One WiMAX Base Station (BS) and ten (10) Subscriber Stations (SS) are contained in each cell. The configuration of both the BS and SS performance parameters are depicted in Figure 2 and Figure 3 below. We used a server backbone with one voice server. In this simulation setup, we performed the following experiments:

Experiment 1: here we used scenario 1 simulation to study the effect of different codecs on VoIP services over WiMAX networks. The encoder schemes used for the investigation include ITU-T G.711 (default encoder scheme), G.723 and G.729 with voice frame size used per packet set to “10”.

Experiment 2: here we used scenario2 simulation to study the VoIP performances over WiMAX using different codec schemes with voice frame size used per packet set to ‘4’ with the same encoder schemes used in scenario1.

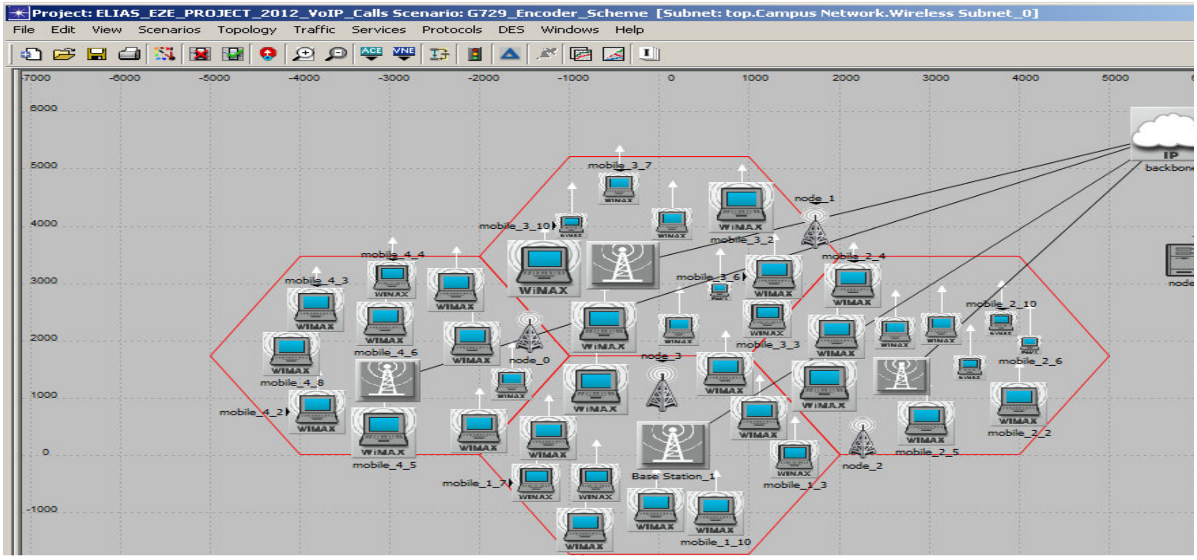
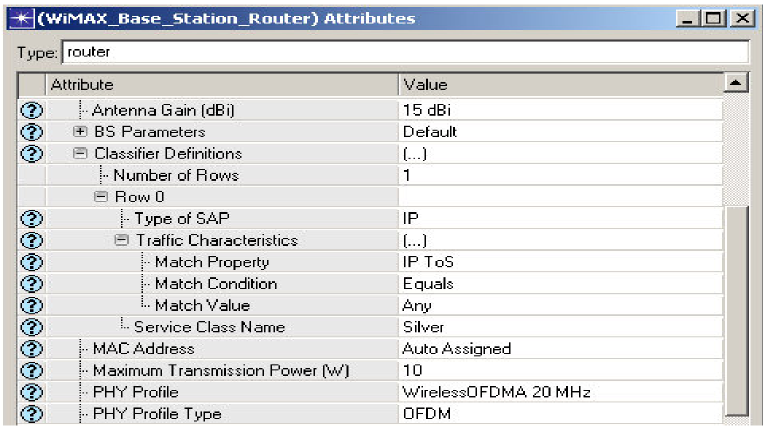


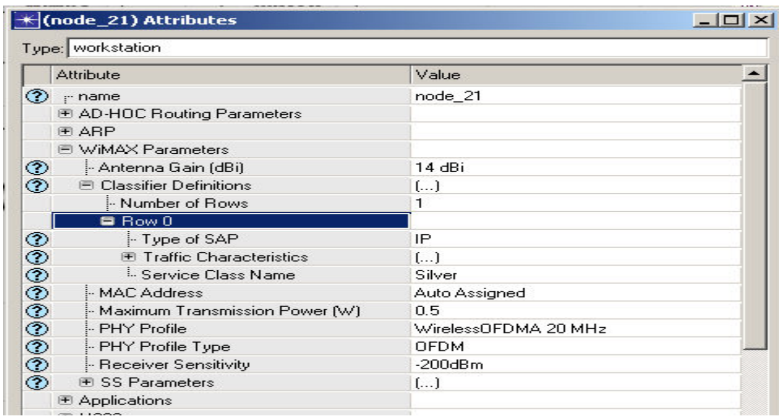
Figure 1: WiMAX Access Networks Model


(WiMAX_Base_Station_Router) Attributes

Type: router

Attribute	Value
Antenna Gain (dBi)	15 dBi
BS Parameters	Default
Classifier Definitions	(...)
Number of Rows	1
Row 0	
Type of SAP	IP
Traffic Characteristics	(...)
Match Property	IP ToS
Match Condition	Equals
Match Value	Any
Service Class Name	Silver
MAC Address	Auto Assigned
Maximum Transmission Power (W)	10
PHY Profile	WirelessOFDMA 20 MHz
PHY Profile Type	OFDM

Figure 2: WiMAX Base Station (BS) Parameters configuration


(node_21) Attributes

Type: workstation

Attribute	Value
name	node_21
AD-HOC Routing Parameters	
ARP	
WiMAX Parameters	
Antenna Gain (dBi)	14 dBi
Classifier Definitions	(...)
Number of Rows	1
Row 0	
Type of SAP	IP
Traffic Characteristics	(...)
Service Class Name	Silver
MAC Address	Auto Assigned
Maximum Transmission Power (W)	0.5
PHY Profile	WirelessOFDMA 20 MHz
PHY Profile Type	OFDM
Receiver Sensitivity	-200dBm
SS Parameters	(...)
Applications	

Figure 3: WiMAX Subscriber Workstation (WiMAX SS) Parameters

3.1 Configuration/settings for VoIP traffic mode Encoder Schemes

Both scenario1 and scenario2 were configured to use ITU-Ts' G.711 (default codec scheme), G.723, G.729.

Given that scenario2 is used to simulate and study the effects of applying different encoder algorithms to VoIP system over WiMAX network traffics with the aim of determining the best encoder scheme that will improve voice quality in call sessions over WiMAX. Figure 4, Figure 5 and Figure 6 below show the configuration/settings for the three different types of encoder schemes simulated.

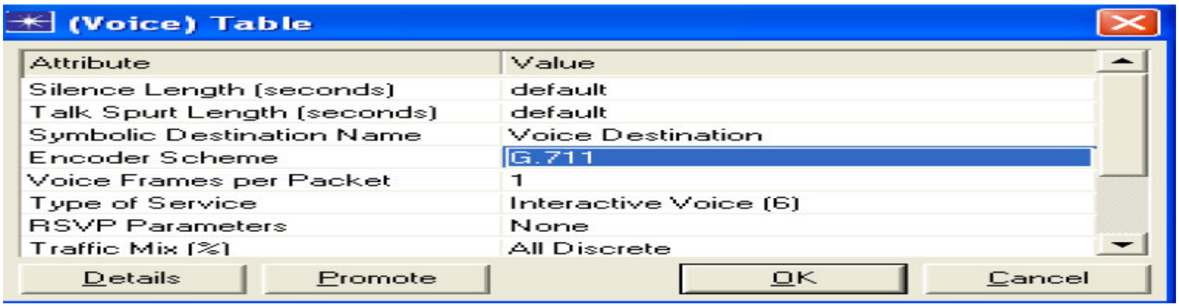


Figure 4: configurations for ITU-Ts' G.711 encoder schemes

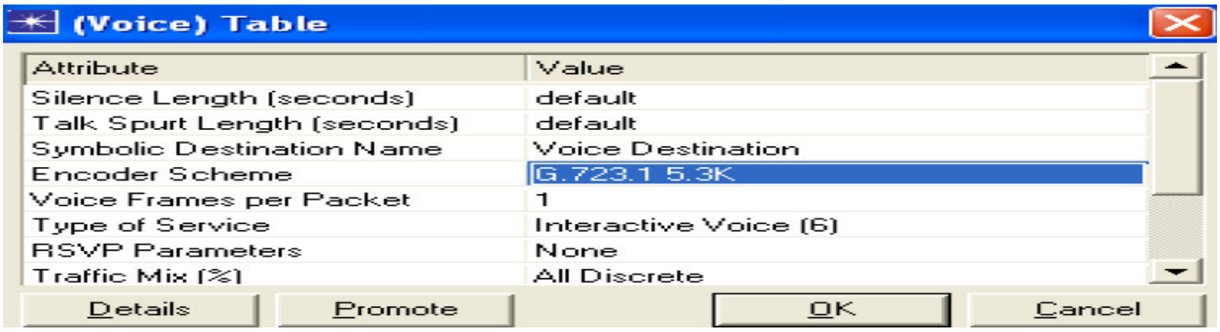


Figure 5: configurations for ITU-Ts' G.723 encoder schemes

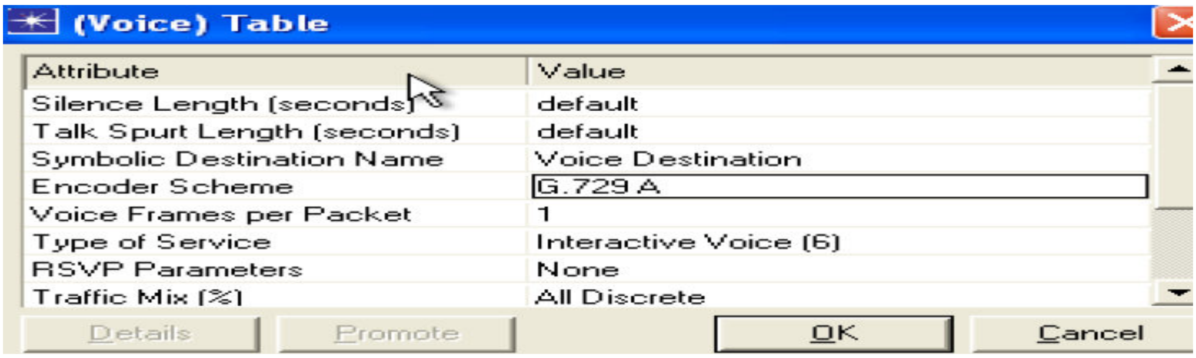


Figure 6: configurations for ITU-Ts' G.729 encoder schemes

4. Simulation Results and Comparative Analysis

Here we carried out a detailed comparative analysis of the modelled network performance of voice over IP using WiMAX network via extensive network research simulation methodology. The results of the OPNET simulation experiments are presented in the following sections.

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4.1 Senario1: Application of Different Codec Schemes

This scenario was simulated to study the effect of different codecs on VoIP services over WiMAX networks. The encoder schemes used for the investigation include ITU-T G.711 (default encoder scheme), G.723 and G.729 with different voice frame sizes used per packet. Many related researches with reference to those cited in our related literature review have carried out a similar simulation experiment but they did not consider the fact that reduction in the value of voice frame sizes per packet used with a given codec can have a significant effect on the VoIP system QoS over WiMAX networks. The results obtained from the simulation experiments are shown below in Figure 7(a) through Figure 7(f) and Figure 8(a) through Figure 8(f).

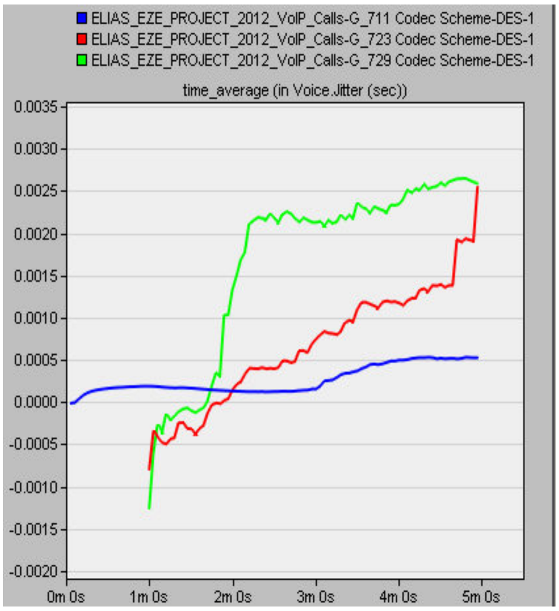


Figure 7(a): Voice Jitter.

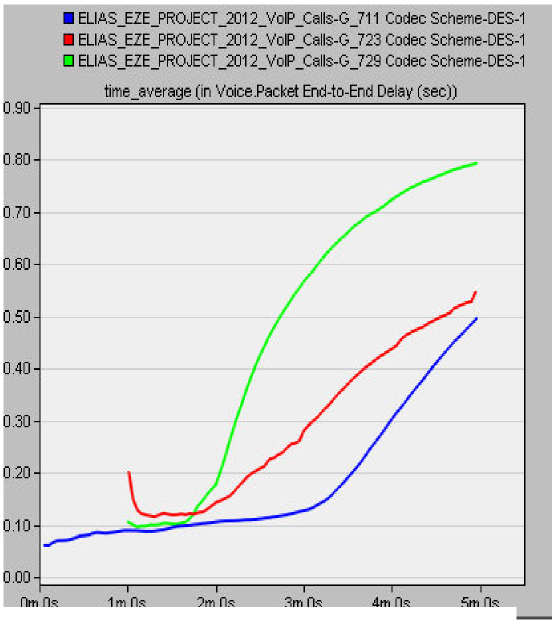


Figure 7 (b):

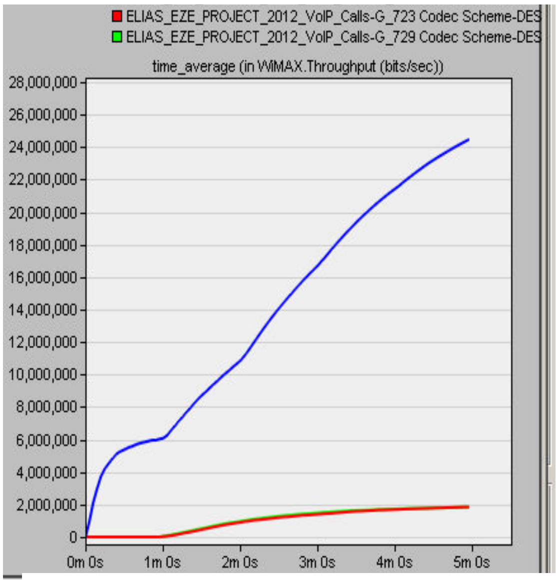


Figure 7 (c): Network Throughput.

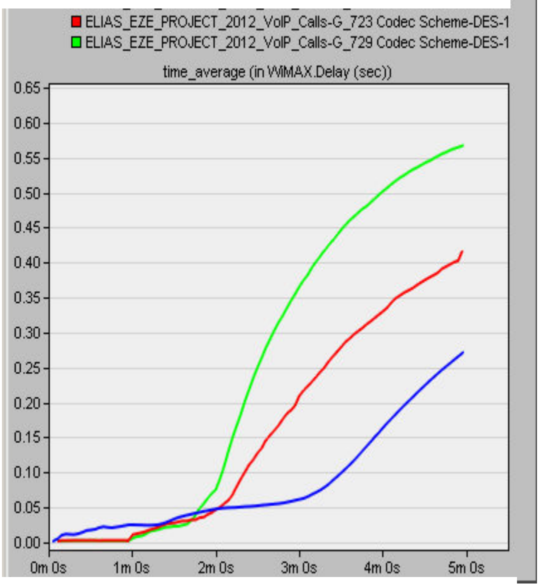


Figure 7 (d):

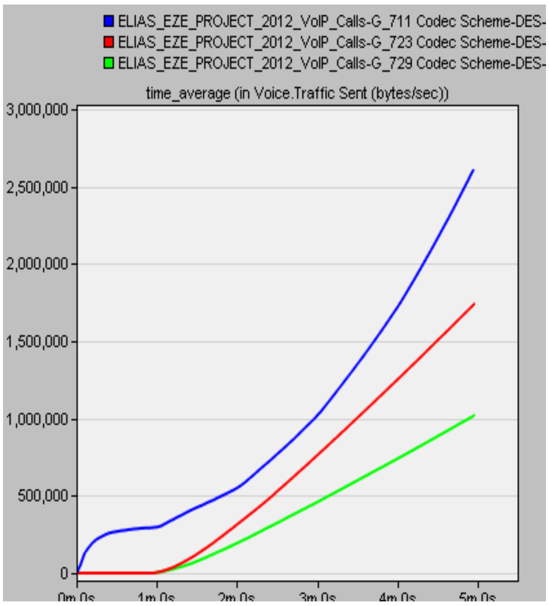


Figure 7(e): Voice Traffic Sent.

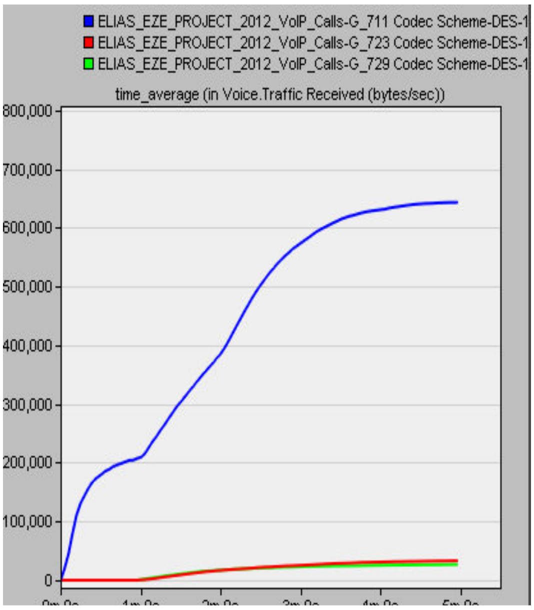


Figure 7 (f): Voice Traffic Received.

Figure 7(a) – Figure 7(f): VoIP performances over different codec schemes with Voice frame size per packet set to '10'.

Figure 7(a) and Figure 8(a) compare voice jitter levels when different codecs are applied. While value of '10' was set for the voice frame size per packet in Figure 7(a), G.711 encoder yielded the lowest voice jitter of 6.0ms (0.0006 seconds) and G.729 with the highest value of 27ms (0.0027 seconds). In Figure 8(a) where voice frame size per packet was set to a value of '4', the jitter value of G.729 encoder decreased to 25ms while G.711 yielded yet least voice jitter value of 2ms (0.0002 seconds). In all the three encoder schemes applied and studied, the voice jitter values are within the range of acceptable voice jitter threshold but the results indicated that the less the value of voice frame size per packet the lesser the voice jitter value witnessed.

The VoIP traffic suffered high packet End-to-End (ETE) delay as clearly witnessed in both Figure 7(b) and Figure 8(b) with G.729 encoder producing the highest packet ETE delay of 8.0s and G.711 encoder yielding the lowest packet ETE delay of 5.0s. These range of delay adversely affected the general network QoS as the acceptable voice packet ETE delay is always below 200ms. Surprisingly, after reducing the value of voice frame size per packet to '4', the range of packet ETE delay values obtained drastically reduced towards the acceptable packet ETE delay threshold given that G.711 had a packet ETE delay value of 1.1s, G.723 had 5.5s and G.729 had 7.9s as shown in Figure 8(b). These results show that the use of G.711 encoder as well as the reduction in the value of voice frame size per packet promotes improved performance of VoIP applications over WiMAX Networks.

In scenario2 simulation experiment, it has been discovered and established that QoS of VoIP call session improves with a decrease in value of voice frame size per packet. In addition, the simulation results also shows that G.711 encoder yields the highest network throughput as seen in Figure 7(c) and Figure 8(c) with total throughput rates of 24,490,000 bits per second and 23,530,000 bits per second respectively. Therefore, the use of G.711 encoding scheme yields the highest network traffic throughput compared to G.723 and G.729 codecs which were investigated in scenario2. Hence, the reason for its wide acceptance and common application in transmitting voice over IP-based network services. Both G723 and G.729 codecs yielded minimal average throughputs of 1,998,000 bits per second and 2,000,000 bits per second respectively which are quite negligible compared to the result with G.711 codec.

Figure 7(d) and Figure 8(d) presented the average WiMAX network delay in VoIP traffic over different encoder schemes. In Figure 7(d), G.711 encoder had the least WiMAX network delay of 2.6s while G.729 recorded the highest network delay of 5.7s. After reducing the value of Voice frame size per packet to '4', the average

WiMAX network delay recorded with the use of G.723 encoder decreased to 3.8s in Figure 8(d) as opposed to 4.2s in Figure 7(d). These ranges of network delay can render the transmitted voice signals totally incomprehensible thereby adversely affecting the QoS of VoIP call sessions over WiMAX networks except the result in the case of G.711 (Figure 7 (d)).

According to Figure 7(e) and Figure 8(e), G.711 encoder recorded the highest amount of voice packet transmitted at the rate of 2,700,000 bytes/second while G.729 had the lowest voice packet transmission rate of 1,100,000 bytes/second. After setting the traffics' voice frame size per packet to '4', G.729 encoder maintains voice packet sent value of 1,250,000 bytes/second. Hence, G.711 codec still proves the best suitable voice codec scheme for optimal QoS in VoIP applications over WiMAX networks.

The average rates of voice traffics received are shown in the graphs contained in Figure 7(f) and Figure 8(f). Once more, the highest rate of voice packet-sent-packet-received was achieved with G.711 voice codec as it yielded a total average of 650,000 bytes/second as opposed to minimal rate of 20,000 bytes/second and 19,000 bytes/second that were recorded by G.723 and G.729 voice codec respectively. Figure 7(f) shows that the value of voice frame size per packet has a significant impact over the encoder scheme applied in VoIP deployed with WiMAX Networks. Hence, the general network performance increases with a corresponding decrease in voice frame size per packet.

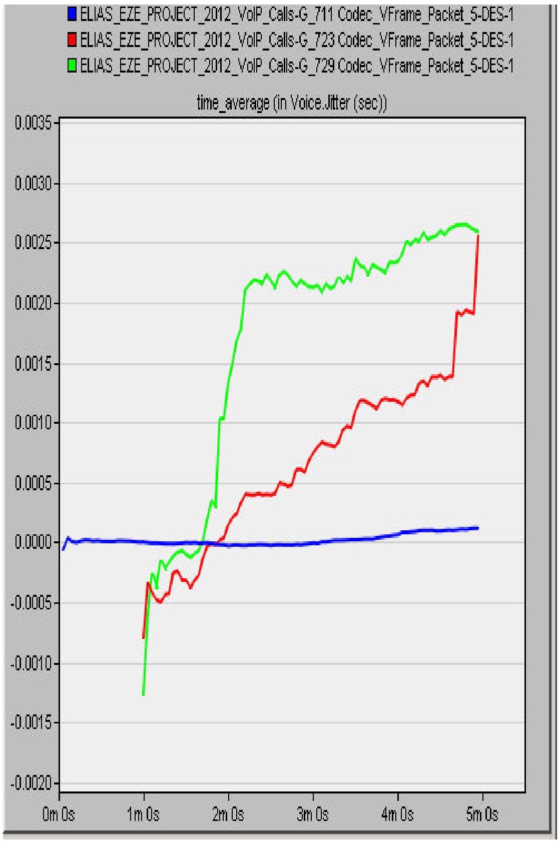


Figure 8 (a): Voice Jitter.

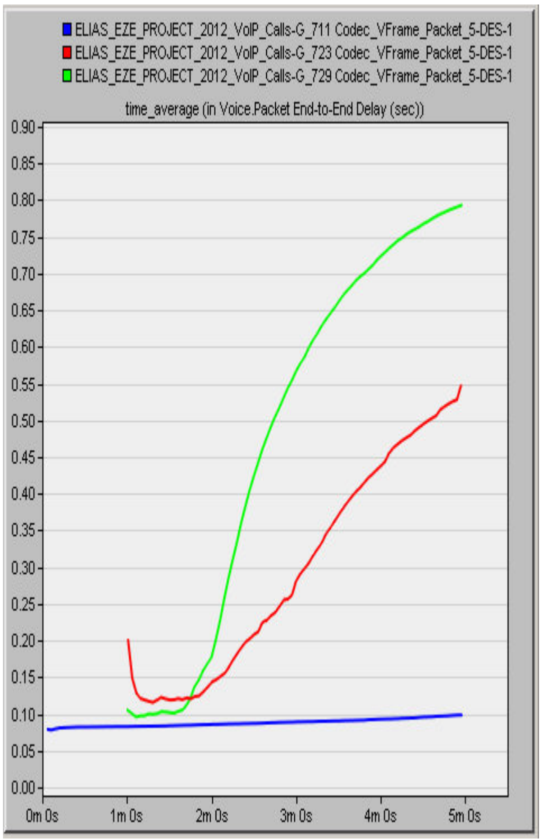
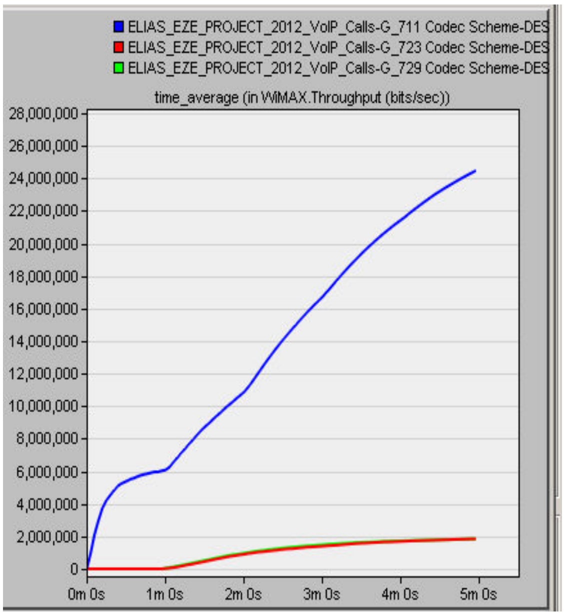


Figure 8 (b): Voice Packet ETE



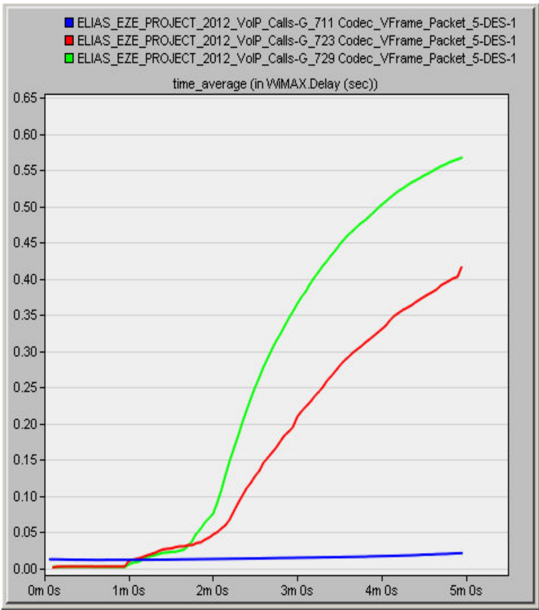
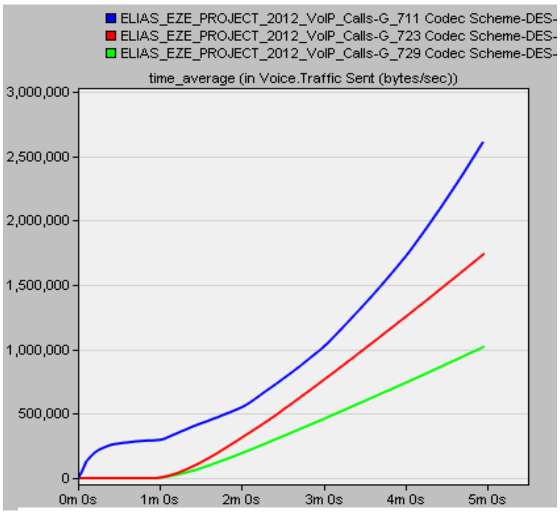


Figure 8 (c): Network Throughput.

Figure 8 (d):



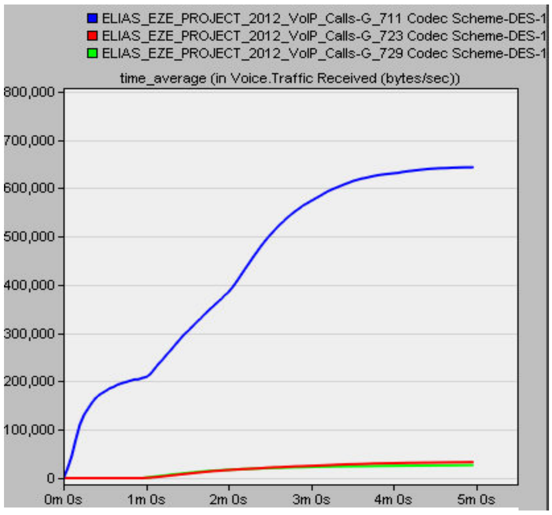


Figure 8(e): Voice Traffic Sent.

Figure 8(f): Voice Traffic

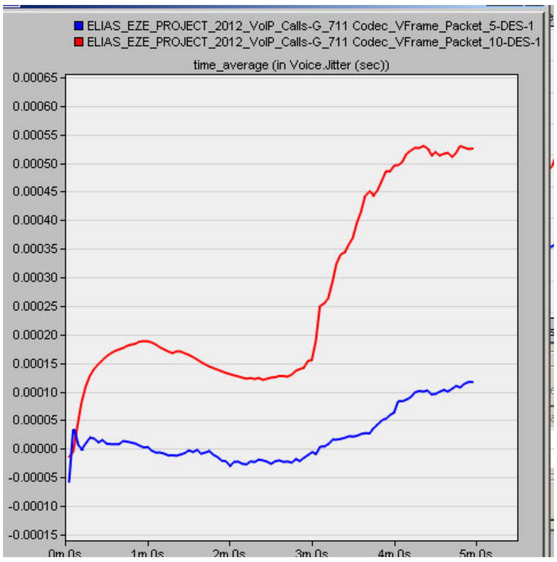
Figure 8(a) – Figure 8(f): VoIP performances over different codec schemes with Voice frame size used per packet set to '4'.

Table 1 below presents the summarized experimental result acquired after running the simulation with different codecs using varying voice frame sizes used per packet.

Voice Codecs		**Voice Jitter(s)**	**Packet delay variation(s)**	**Packet ETE delay(s)**	**WiMAX Network delay(s)**	**Voice traffic Sent(pps)**	**Voice traffic received(pps)**	**Packet loss rate(pps)**
Voice Frame size/ packet= '10'	G.711	0.002	0.006	1.00	0.80	10,348	9,898	3.50%
	G.723	0.003	0.259	1.74	1.54	20,036	19,899	0.34%
	G.729	0.006	0.779	3.51	3.00	10,256	9,897	3.50%
Voice Frame size/ packet= '4'	G.711	0.005	0.081	1.41	2.41	12,898	11,898	0.00%
	G.723	0.007	0.239	2.49	1.21	44,519	44,519	0.00%
	G.729	0.003	0.259	1.79	1.71	20,035	19,798	0.73%

Table 1: Summarized results of the simulation experiments with different codecs using varying voice frame sizes used per pack.

More so, Figure 9 (a) and Figure 9 (b) further proved that a reduction in the value of voice frame size per packet used with the codecs has an astronomical impact on VoIP traffic QoS; hence, the same G.711 codec under different voice frame sizes per packet '10' and '5' generated a wide difference in voice jitter and voice packet ETE delay as clearly shown below.



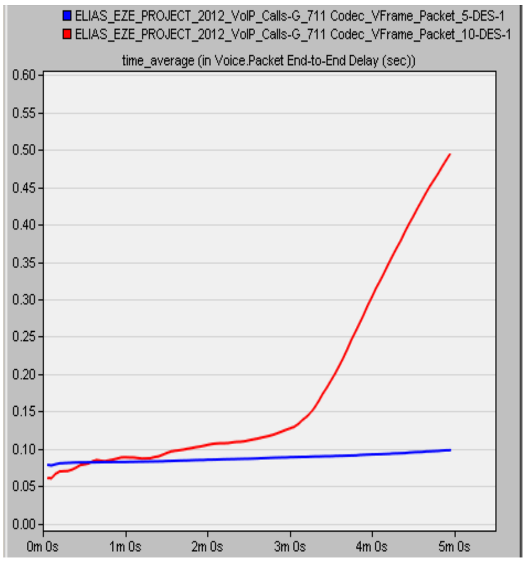


Figure 9 (a): Voice Jitter. Figure 9 (b): Voice Packet

Figure 9 (a) – Figure 9 (b): VoIP system performance over WiMAX networks with the same G.711 codec scheme under different voice frame sizes per packet (Voice Frame size per packet set to '10' and '5').

5. Conclusions

In this research, we used OPNET modeler 16.1 network simulation tool to investigate and study VoIP applications performance over WiMAX broadband access technology in a campus network. Several network performance parameters including voice packet ETE delay, voice jitter, WiMAX network delay, packet delay variation, throughput and so on, were used to determine the quality of VoIP calls that can be guaranteed with good QoS over WiMAX network and the most suitable codec scheme that will produce the best voice quality for VoIP systems deployed with IEEE 802.16e standard based WiMAX access technology. The research study investigated VoIP system performance under different codec schemes. Three different voice codec schemes (G.711, G.723 & G.729) were studied using simulation methodology in order to determine the best suitable voice codec scheme that will yield optimal voice quality for VoIP systems over IEEE 802.16e standard WiMAX network. The results of our network research simulation exercise proved that VoIP system yields best performance (improved QoS) under G.711 codec scheme as opposed to G.723 as well as G.729 codec schemes as were clearly shown in Figure 7 (a) through Figure 7 (f) in section four (4) above. The obtained simulation experiment results indicated that choice of suitable codec scheme can affect the QoS of VoIP traffic over WiMAX network. The results also indicated that the choice of suitable voice encoder scheme with a small number of voice frame-size per packet have a significant impact over VoIP traffic performance when deployed with WiMAX access technology.

Future work could focus on VoIP system security through the implementation of IPSec Utilities via VoIP gateways components in order to establish control over VoIP call connections, routing and management of both Terminal and Multipoint Control Units (MCU). These IPSec Utilities if implemented on VoIP applications over WiMAX networks can go a long way to reduce call interference and check against unauthorized calls.

6. References

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