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A Comparative Study of VoIP over IEEE 802.11(b, g) and WiMax (UGS, ertPS) Wireless Network Technologies

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***Abstract*—This paper describes a comparative study of the performance of VoIP over wireless networks using OPNET tool. The simulation study is completed by running VoIP application in different network scenarios with IEEE 802.16 (UGS, ertPS) and IEEE 802.11 (b, g) with best effort service and interactive service The result clearly illustrated that the WiMax type ertPS has the best performance among all tested cases.**

***Keywords*— VoIP; Performance evaluation; WiMax, Wireless LAN.**

1. INTRODUCTION

Currently, very huge multimedia streams are transferred between millions of persons using different Internet social societies and networks across the world; making the world as a small village as a true reality. In addition, users are asking for more advanced multimedia applications' features. The multimedia applications can be classified into (i) streaming stored audio/video; (ii) interactive audio/video; and (iii) streaming live audio/video [1]. However, one of the emerging technologies that are currently evolving very rapidly as a carrier for the multimedia streaming is the wireless networks, which is motivated by the extensive use of mobile devices and Internet by hundreds millions of people all over the world. Hence, wireless networking is becoming very widespread in all countries, as well as there are many industrial companies developing different devices and environments for wireless technologies. In another hand, there are well-known technical organizations are continuously developing wireless standards such as IEEE. However, IEEE developed IEEE 802.11 (wireless LANs) and 802.16 *Worldwide Interoperability for MicroWave Access* (WiMax) standards that are designed to be used with the first and the second layers of the OSI model. Moreover, both the standards are used with a range of network layers such as IP [2] [3].

Voice over Internet Protocol (VoIP) is an example of real-time interactive audio/video application; it enables voice communication between two or more participants over the Internet. The VoIP procedure includes voice digitization, filtering of unwanted noise, compression using codecs, then packetization and sending through Internet. VoIP is supported by number of protocols such as session initiation protocol (SIP) or H.323, real-time transport protocol (RTP) and real-time transport control protocol (RTCP). Those protocols have been employed to initiate, maintain, and terminate the VoIP call. However, the call parties need to be identified by IP-address and sufficient bandwidth capacity needed to be available to finish the call successfully. This in turn, introduced the concept of the quality of service (QoS) in communication networks that are used to carry multimedia streaming such as VoIP. To quantitatively measure QoS several related features of the network service are often measured, such as end-to-end transmission delay, delay variation (jitter), packet loss, bandwidth, throughput, and network up-time. QoS means the ability of the network to deliver better service to certain network traffic by managing network resources to satisfy application requirements by controlling the end-to-end delay, delay jitter and packets drop. These requirements are required by some real-time and interactive traffic flows, while making sure that are not affecting other traffics bandwidth needs [1][4].

This paper is organized as follows: important definitions related to QoS are defined in section 2. The related work is briefly discussed in section 3. Section 4 introduces the reader to wireless LAN networks. WiMax technology is described in section 5. Section 6 outlines results of the simulation study produced in this paper. Finally, conclusion remarks are given in section 7.

2. DEFINITIONS

In this section, useful terms and definitions that are related to the paper context is presented [4].

Computer network it is a communication system that consists of tangible and intangible resources. Tangible resources are servers, routers, switches, and links. Intangible resources are like packets and frames. Hence, the QoS is the ability to manage network's tangible assets to meet the requirements of intangible assets in terms of end-to-end delay, delay jitter, and packets lose.

Congestion the lack of bandwidth (BW) will cause congestion. The congestion is happen when there are requirements of BW for the current stream that is greater than the available capacity or impulsive flood in loads or unexpected traffic flowing due to rerouting.

Flow is any unique stream of related packets such as single VoIP or video stream that resulted from a single user activity and needs the same QoS management.

Real-time application is defined as the application that needs the data in each packet arrived by a definite time and, if the data has not arrived at that time, it is then really useless. The real-time applications always do not work well across the Internet due to variable queuing delay and congestion losses. Hence, VoIP requires short delay and strict delay variation (jitter) of the packets.

Packet end-to-end delay is defined as the difference in the time at which the packet enters the network and the time at which it leaves the network; from synchronized sender to destination; including queuing and intermediate networking devices delay.

Delay variation (**Jitter**) is defined as the delay variation between two successive packets belonging to the same traffic stream. The delay jitter is often caused by queuing and rerouting and additional processing delays.

Bandwidth (BW) it is the ability of the network to deliver better service to particular network traffic within IP networks using effective QoS utilization which will result in a maximum use of available bandwidth.

Packet loss is defined as the number of packets that are lost during transmission process inside the network within a specified time interval. It is unavoidable under heavy traffic and usage of resources by different applications conditions. Packet loss may be caused by congestion, traffic rate limiting, physical layer errors, and network elements failures.

Throughput is defined in the network as the total number at which packets are transmitted from source to destination at a defined time period (e.g. packet/seconds).

3. RELATED WORK

There are a number of studies have been carried out by many researchers in this field. This section provides an overview of some significant research papers. In [5] authors evaluated the VoIP application on the wireless LAN (802.11) using OPNET, however, they used only four levels of priority to classify the traffic as QoS measure for a 2 minute time interval. In [7] authors have been evaluated VoIP over WiMax networks with different scenarios with different distances. In [8] authors have been studied and compared two options of comparison between WiFi and WiMax using *NS2* simulation tool. The VoIP measured as TCP flows that is different than OPNET tool, which generates VoIP traffic as calls with many parameters that shows much nearly real comparison with UDP/RTP protocols. In [9] authors reported a comparison study of VoIP over WiMax using different codecs only. In [10] authors evaluated the performance of three VoIP codecs over WiMax and WiFi networks. In [11] authors described a comparison between WiFi and WiMax with HTTP and FTP traffic. This paper differs with all mentioned above papers in such a way that it makes detailed comparison of VoIP over Wi-Fi 802.11 (b, g) and WiMa*x* (UGS and ertPS) using OPNET tool with four different scenarios each scenario with two services (best effort and interactive voice services) and measuring QoS with delay jitter, end-to-end delay, and Wireless LAN/WiMax delays.

4. IEEE WIRELESS LAN NETWORKS

The 802.11 standard refers to a family of specifications developed by the IEEE for wireless LAN technology that covers the physical and data link layers. The 802.11 standard specifies an over-the-air interface between a wireless client and a base station or between two wireless clients. The IEEE 802.11 family of standards is one of enabling technologies of wireless networks which use the Ethernet protocol and *carrier sense multiple accesses with collision avoidance* protocol (CSMA/CA). Wireless LANs have the following advantages: (i) ease of use with rich-developed different mobile devices; (ii) availability of different wireless public-mobile networks; (iii) increased productivity with the availability of public-mobile networks; (iv) quick deployment during the initial setup of an infrastructure-based wireless network; (v) expandability by accepting suddenly-increased number of clients with the existing equipment; (vi) the cost is acceptable. The wireless LANs have the following limitations: (i) security since the more commonly used encryption methods with WiFi are known with their weaknesses points; (ii) the typical range of this technology is a limitation; (iii) reliability since radio signals are subject to a wide variety of interference; (iv) speed of the WiFi is much less than wired networks[2].

There are several specifications in the 802.11 family of standards:

4.1. 802.11

The 802.11 applies to wireless LANs and provides 1 or 2 Mbps transmission in the 2.4 GHz band using either *frequency hopping spread spectrum* (FHSS) or *direct sequence spread spectrum* (DSSS). The FHSS is a very solid technology that has little influence from noises, reflections, other radio stations, or other environmental factors [2] [12]. Moreover, the number of simultaneously active systems in the same geographic area is considerably higher than the same number for DSSS systems. However, DSSS is very sensitive technology since it is influenced by many environmental factors such as reflection. Moreover, DSSS provides higher capacities than FHSS [2] [12].

4.2. 802.11a

It is an extension to 802.11 that applies to wireless LANs and provides up to 54 Mbps in the 5GHz band. 802.11a uses an *orthogonal frequency division multiplexing* (OFDM) encoding scheme rather than FHSS or DSSS. The 802.11a specification applies to wireless Automatic Transfer Machine (ATM) systems and is used in access hubs. The OFDM technique uses a large number of parallel narrow-band subcarriers instead of a single wide-band carrier to transport information. This feature makes OFDM easy and efficient in dealing with multipath and robust against narrow-band interference, which makes OFDM the suitable choice for high-speed data rate transmission [12].

4.3. 802.11b

It is also referred to as 802.11 high rate or Wi-Fi. It is an extension to 802.11 that applies to wireless LANs and provides

11 Mbps transmission (with a fallback to 5.5, 2 and 1 Mbps) in the 2.4 GHz band. 802.11b uses only DSSS. However 802.11b is a modification of the original 802.11 standard, allowing wireless functionality comparable to Ethernet [2] [12].

4.4. 802.11g

This type offers wireless transmission over relatively short distances at 20 – 54 Mbps in the 2.4 GHz band. 802.11g also uses the OFDM encoding scheme. However, 802.11g is always used for a last-mile solution because of its speed, capability to manage interference and interoperability with 802.11b-based devices [2] [12].

4.5. 802.11n

This type is an improvement to the previous 802.11 standards by adding multiple-input multiple-output antennas (MIMO). 802.11n uses OFDM and it operates at a maximum data rate from 54 Mbps to 600 Mbps in the 2.4 GHz band and 5 GHz. However, 802.11n had improved the network throughput over the 802.11a/g [2].

5. WIMAX NETWORKS

WiMax is an IP-based system that provides wireless high-speed Internet access to home and business subscribers, on metropolitan distances BS which can handle thousands of subscriber stations (SS) with QoS similar to cellular networks. WiMax is IEEE standard 802.16 based on OFDM technique. It is a wireless digital communications system, which ensures compatibility and interoperability between broadband wireless access equipment. WiMax supports user's mobility and broadband multimedia services delivery. WiMax supports: Data, voice systems such as VoIP, TCP/IP and video with different QoS. WiMax is less expensive than cable, much easier to extend to rural areas, broad coverage like cell phone networks [3] [6] [13].

WiMax has the following advantages: (i) covers wide areas with many users simultaneously; (ii) provides high-speed even over a large area; (iii) WiMax ranges from 2-to-10GHz ultra-wide band; (iv) offers better security with low cost. In another hand, WiMax has the following disadvantages: (i) the signal may be affected by weather conditions; (ii) line of sight is needed for longer connections; (iii) power-intensive technology; (iv) can be interfered by other wireless devices.

- In this paper we intend to discuss two WiMax classes:

(i) Unsolicited Grant Service(UGS)

This WiMax QoS class delivers fixed bandwidth allocation on periodic basis. UGS reserves bandwidth during setup time and it supports constant bit rate (CBR) services, such as T1/E1 emulation and VoIP without silence suppression. The scheduling in this type is static allocation with grant equal to *Maximum Sustained Traffic* (MST) rate. As quality parameters this type of service can tolerate maximum latency and delay jitter [3] [13].

(ii) Extended Real Time Polling Service(ertPS)

This WiMax QoS type is developed to support VoIP with silence suppression that has variable data rates with guaranteed data rate and delay requirements. The scheduling in this type is dynamic allocation, which reserves bandwidth during setup and allows bandwidth stealing [3] [13].

6. RESULTS

In this paper the OPNET simulation tool [14] is used to compare the performance of the VoIP over IEEE wireless LAN 802.11 with WiMax wireless IEEE 802.16. The model networks for both technologies consists of 15 wireless nodes distributed in 1 km × 1 km area as shown in figure (1) and figure (2) respectively.

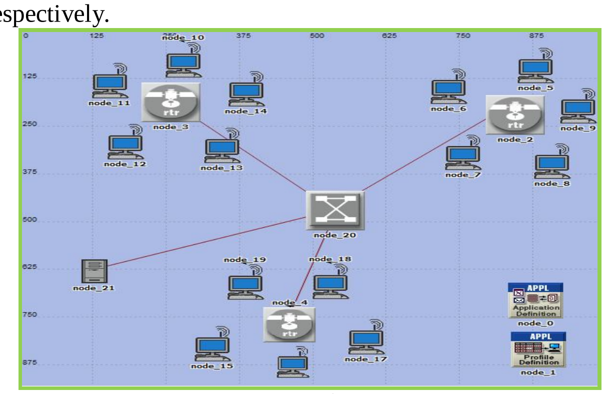

Figure 1: Topology of Wi-Fi case study

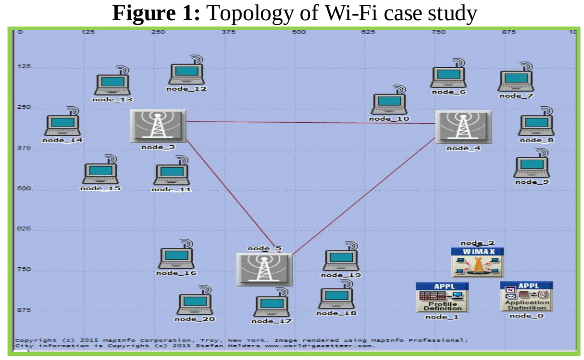

Figure 2: Topology of WiMax case study

However, the model details are as follows: IEEE 802.11b (direct sequence) standard 11 Mbps; IEEE802.11g (Extended Rate PHY) with 54Mbps data rate. The WiMax IEEE 802.16 UGS/ ertPS with 384 Kbps data rate.

- The First Wi-Fi scenario consists of three *Access Points* (AP) and 15 workstations, each five workstations connected to one AP as seen in figure (1). APs are connected to the switch and to server. Server is configured for one network application: VoIP application definition is setup for the same application as for the server. Addition profile definition is provided to enable the application over nodes.
- The WiMax case study model consists of three **BSs** connected by link (Base X 100_int), each **BS** connected to 5 workstations.
- The simulation run is lasted for 60 minutes (3600 seconds) period, the VoIP traffic has been configured between all nodes by using "create traffic flow" option in OPNET tool, with the following input parameters: call rate is 600 calls per hour, average call duration is 300s (5 min), voice flow duration is 3600s, the encoder scheme is G.711, traffic type is interactive voice with delay, throughput and reliability

including overhead (bytes) of RTP/UDP/IP. Therefore, the voice traffic after has been generated by the OPNET tool produced a huge data measured in hundreds of gigabytes flown in the networks.

- The experimental study included the VoIP application in different scenarios with combination of the following features:
  (i) IEEE 802.11b and IEEE 802.11g
  (ii) WiMax UGS and WiMax ertPS types.
  (iii) Type of service: best effort or interactive voice service.

6.1- Results of VoIP analysis for IEEE 802.11b with the comparison between best effort and interactive voice services

Figure (3) obviously shows that, the best effort service has higher jitter (worse) than the interactive voice service, while interactive voice has gained much less delay jitter. Figure (4) shows that the best effort service has higher (the worst) packet end-to-end delay values than the interactive voice service. Thus, interactive voice is better because it has less packet end-to-end delay. Figure (5) illustrates that the throughput for both cases is nearly the same at the beginning of the curve; but the throughput of the interactive voice service case has got the higher throughput rate at the second half of the curve.

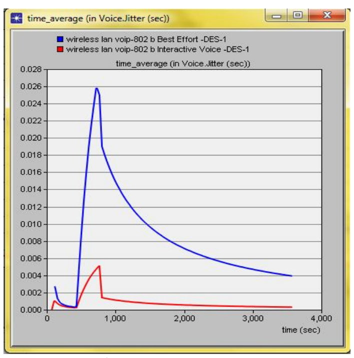

Figure 3: Voice jitter for IEEE 802.11b, using best effort and interactive voice services

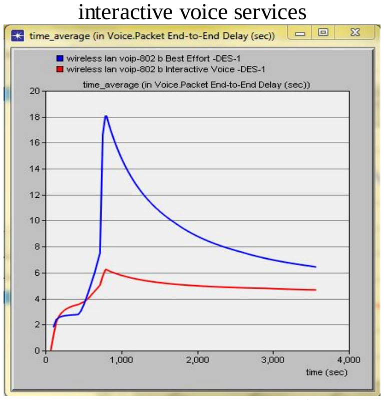

Figure 4: Voice packet end-to-end delay (sec) for IEEE 802.11b with best effort and interactive voice services

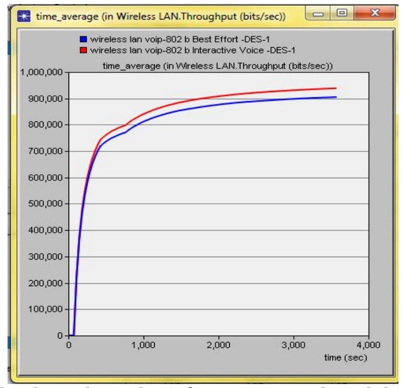

Figure 5: The throughput (bps) for IEEE 802.1b with best effort and interactive voice services

6.2- Results of VoIP analysis for IEEE802.11g with the comparison between best effort and interactive voice services

Figure (6) shows that best effort has higher jitter values than the interactive voice service. Hence, best effort service seems to be worse because it has more delay jitter than IEEE802.11g traffic with interactive voice. Figure (7) illustrates that the best effort has higher (worse) end-to-end delay value than that of interactive voice service. Hence, the interactive voice service is better because it has less packet end-to-end delay. Figure (8) describes the best service has got higher delay (worse) than interactive service for VoIP over wireless networks. Therefore, the interactive voice service is better because it has less wireless LAN delay.

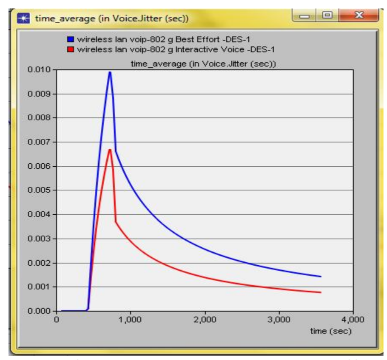

Figure 6: Voice jitter for IEEE802.11g, using best effort and interactive voice service

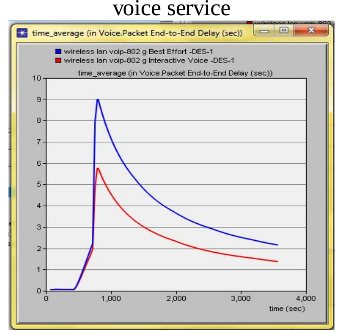

Figure 7: Packet end-to-end delay (sec) for IEEE802.11g, with best effort and interactive voice services

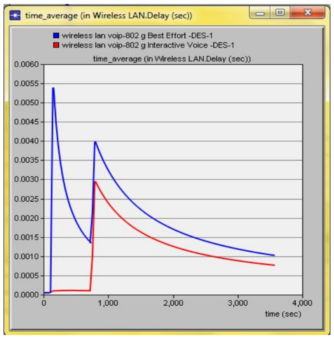

Figure 8: Wireless LAN delay for IEEE802.11g with best effort and interactive voice services

6.3- Results of VoIP analysis between both Wi-Fi networks IEEE802.11b and IEEE802.11g protocols using best effort service

Figure (9) shows that, IEEE802.11b has higher delay jitter than the IEEE802g; hence, IEEE802.11b seems to be worse because it has more delay jitter than IEEE802.11g. Figure (10) illustrates that IEEE802.11b has higher (the worst) end-to-end delay assessment than that of IEEE802.11g. Thus, the IEEE802.11g is better than the IEEE802.11b. Figure (11) shows the wireless LAN delay, it is clear that the wireless LAN delay of IEEE 802.11b is higher than that of IEEE 802.11g for the same best effort service; which makes 802.11g is better than IEEE 802.11g for VoIP application.

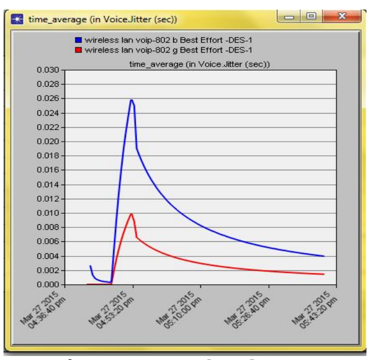

Figure 9: Voice jitter for IEEE802.11b and IEEE802g using best effort service

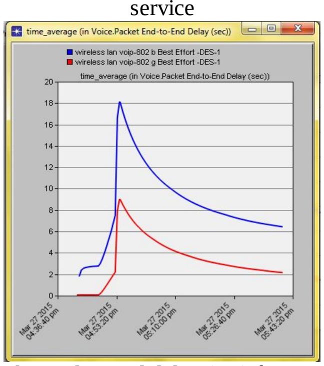

Figure 10: Packet end-to-end delay (sec) for IEEE802.11b and IEEE802.11g, with best effort service

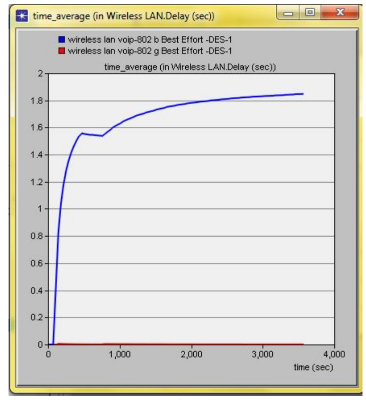

Figure 11: Wireless LAN delay for IEEE802.11b and IEEE802.11g with best effort service

6.4- Results of VoIP analysis between both Wi-Fi networks IEEE802.11b and IEEE802.11g protocols using interactive voice service

Figure (12) displays that IEEE802.11g has higher delay jitter (worst) than the IEEE 802.11b. However, IEEE 802.11b seems to be better in this measurement than IEEE 802.11g. Figure (13) describes that IEEE 802.11b has much higher (worse) end-to-end delay value than that of IEEE 802.11g. Hence, the IEEE802.11g is considered as the best candidate for VoIP service than the IEEE 802.11b. Figure (14) presents the results of the wireless LAN delay; however, the IEEE 802.11b has got higher wireless LAN delay (worse) than IEEE 802.11g for carrying VoIP over wireless networks.

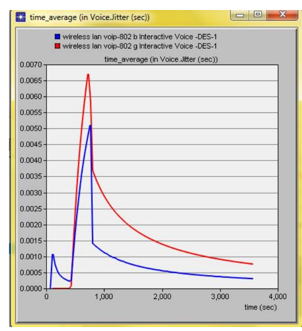

Figure 12: Voice jitter for IEEE 802.11b and IEEE 802.11g, using interactive voice service

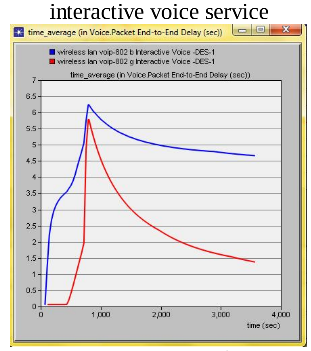

Figure 13: Packet end-to-end delay (sec) for IEEE 802.11b and IEEE 802.11g with interactive voice service

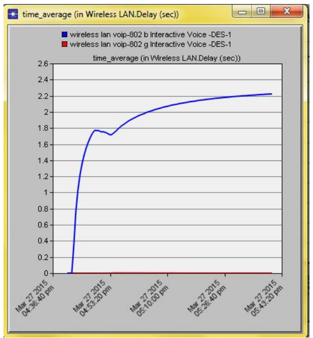

Figure 14: Wireless LAN delay for IEEE 802.11b and IEEE 802.11g with interactive voice service

6.5-Results of VoIP analysis between WiMax networks UGS and ertPS using interactive voice service

Figure (15) shows that WiMax has higher voice jitter with interactive voice service with ertPS while UGS has the less jitter value which makes UGS type WiMax better in jitter performance value.

Figure (16) describes the voice packet end-to-end delay (sec) for UGS and ertPS WiMax technologies with interactive service. The UGS has the highest value (worst). The figure shows that end-to-end packet delay is increasing linearly. Figure (17) illustrates the WiMax delay (sec) using interactive voice service, with ertPS and UGS WiMax types. The comparison shows that UGS has much less WiMax delay than the ertPS.

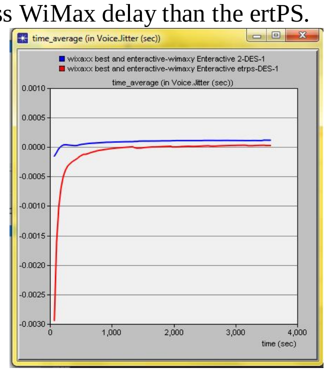

Figure 15: Voice jitter with interactive voice service with ertPS and UGS WiMax networks

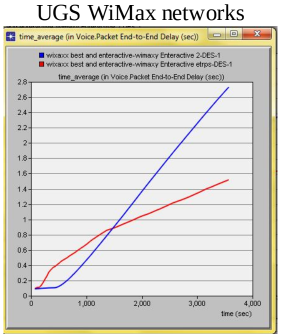

Figure 16: Packet end-to-end delay (sec) for WiMax UGS and ertPS with interactive service

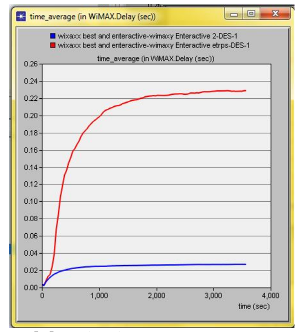

Figure 17: WiMax delay (sec) using interactive voice service, with ertPS and UGS WiMax types

6.6- Results of VoIP analysis between WiMax networks UGS and ertPS using best effort service

Figure (18) illustrates that UGS WiMax has the highest voice jitter values with best effort service while ertPS has the lowest (better in this case).

Figure (19) shows the results of voice packet end-to-end delay (sec) with best effort service for WiMax technologies (UGS, ertPS). It is clear that both have end-to-end delay increasing in linear fashion with time. But UGS has much higher delay (worst) than ertPS. Figure (20) describes the WiMax delay for UGS and ertPS technologies. It is clear that ertPS has got much less delay (better in performance) than UGS when using best effort service.

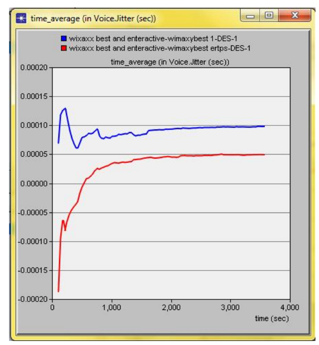

Figure 18: Voice jitter of ertPS and UGS WiMax with best effort service

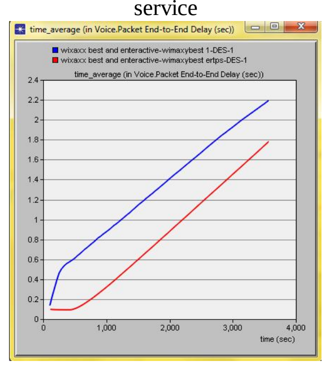

Figure 19: Packet end-to-end delay (sec) with best effort service for WiMax (UGS, ertPS)

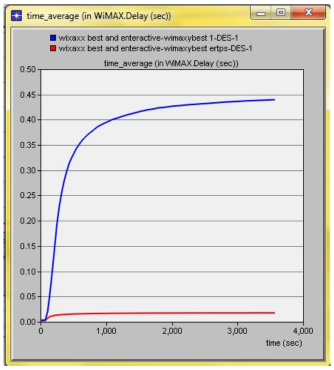

Figure 20: WiMax delay for UGS and ertPS technologies using best effort service

6.7- VoIP analysis results of the comparison between Wi-Fi (b, g) and WiMax (UGS, ertPS) *with best effort service*

Figure (21) shows the voice jitter results for Wi-Fi (b, g) and WiMax (UGS, ertPS). The Wi-Fi network IEEE 802.11b and IEEE 802.11g has the higher voice jitter compared with WiMax networks ertPS and UGS for best effort service. Both WiMax network, have better results because it has lowest and constant Voice jitter around zero value. Figure (22) illustrates voice packet end-to-end delay for Wi-Fi (b, g) and WiMax (UGS, ertPS) wireless networks. The result clearly says that WiMax ertPS has got the lowest end-to-end delay which makes it the best candidate for VoIP among others for best effort service.

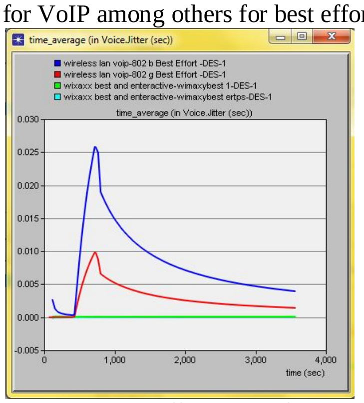

Figure 21: Voice jitter with best effort service for Wi-Fi network (b, g) and WiMax (ertPS**,** UGS)

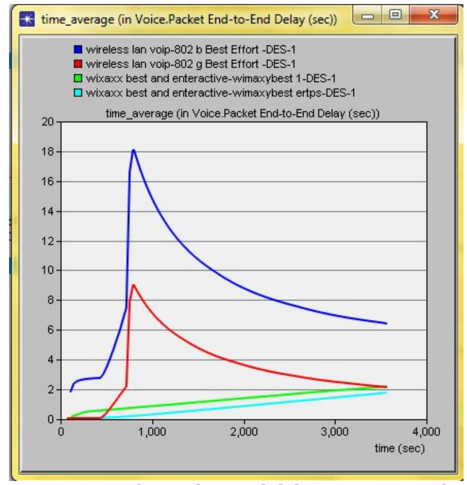

Figure 22: Voice packet end-to-end delay on Wi-Fi and WiMAX networks with best effort service

6.8- VoIP analysis results of the comparison between Wi-Fi (b, g) and WiMax (UGS, ertPS) *with interactive service*

Figure (23) summarizes the voice jitter results for Wi-Fi (b, g) and WiMax (UGS, ertPS). The Wi-Fi network IEEE 802.11b and IEEE 802.11g has the higher (worse) voice jitter compared with WiMax networks ertPS and UGS for interactive voice service. Both WiMax networks have better results; i.e. lowest voice jitter values. Figure (24) illustrates voice packet end-to-end delay for Wi-Fi (b, g) and WiMax (UGS, ertPS) wireless networks. The result noticeably states that WiMax ertPS has got the lowest end-to-end packet delay which makes it the best candidate for VoIP among others for interactive service.

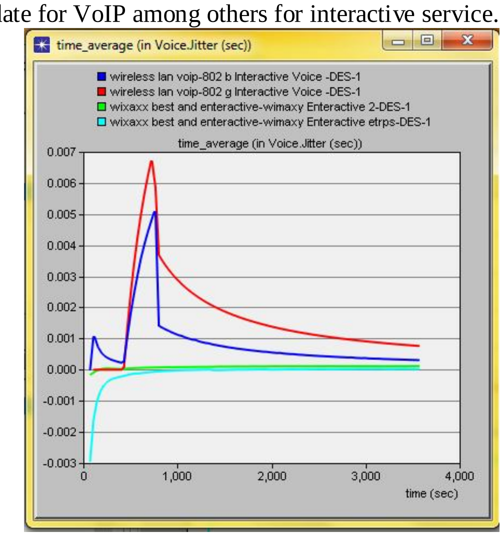

Figure 23: Voice jitter on Wi-Fi and WiMAX networks with interactive service

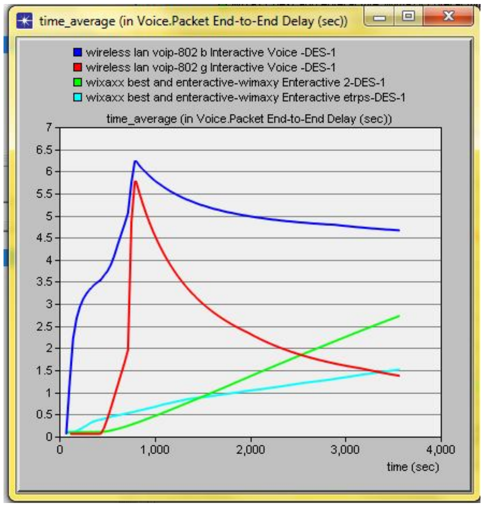

Figure 24: Voice packet end-to-end delay on Wi-Fi and WiMAX networks with interactive service

7. CONCLUSION

In this paper a comparative study of VoIP over the wireless networks IEEE 802.11 (b and g) and WiMax 802.16 (UGS, ertPS) standards has been realized by building network models with different scenarios using the OPNET tool.

The results clearly demonstrated that the wireless networks with IEEE 802.11g protocol have better performance for using VoIP than wireless networks with IEEE 802.11b protocol for both best effort and interactive services considering the packet end-to-end delay and wireless LAN delay. Moreover, the results evidently showed that the WiMax type ertPS has better performance for using VoIP than WiMax type UGS for both best effort and interactive services considering end-to-end delay with delay jitter as major factors. The result clearly presented that the WiMax type ertPS has the best performance among all tested cases.

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