Improving the QoS of VoIP over WiMAX Networks Using OPNET Modeler

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Summary
As a rapidly growing technology, voice over Internet Protocol (VoIP) enables the transmission of voice data over Internet Protocol (IP)-based networks. VoIP has become an alternative solution for the public switched telephone network given its capability to transmit voice data in the form of digital IP packets over TCP/IP-based Internet, particularly with the significant deployment that occurred in the standard of Worldwide Interoperability for Microwave Access (WiMAX, IEEE 802.16) networks. This study investigates the performance of VoIP traffic over WiMAX networks by using various voice codec schemes and by considering several realistic networking scenarios. The Optimized Network Engineering Tool (modular 14.5) was used to measure and analyze a set of quality of service parameters, including mean opinion score packet loss, jitter, and delay. Simulations results indicate that the improved selection of voice codecs and statistical distribution significantly affect VoIP performance in WiMAX networks..

Key words:
BE, ertPS, nrtPS, QoS, rtPS, UGS, WiMAX, VoIP Codecs.

1. Introduction

The demand for digital services with high data transfer rates and increasing bandwidth capabilities is rapidly growing. Examples of these services include multimedia services, mobile television, videophone, and video conferences. Thererefore, continuously developing new telecommunications infrastructure that can support these services has become necessary. This objective can be realized by providing fast Internet access, along with secure and large data transfer rates, and by implementing voice over Internet Protocol (VoIP) and multimedia protocols.

VoIP services have continuously evolved and are widely implemented in all types of wireless and mobile networks. However, VoIP has high bandwidth requirements. Worldwide Interoperability for Microwave Access (WiMAX) technology, which has high bandwidth and wide transmission range, can be considered suitable for supporting VoIP services for many end users (fixed mobile or cellular users). WiMAX is a high-capacity wireless transmission technology that is based on IEEE 802.16 standards. The expansion of wireless broadband services to “anytime and anywhere” is the main objective of WiMAX. For example, a single WiMAX base station can provide services to hundreds of people [1]. Furthermore, WiMAX facilitates the development of Broadband Wireless Access (BWA) service. The use of WiMAX integrated with a wireless local area network (WLAN, IEEE 802.11) increases transmission range by up to 30 miles at speeds of up to 70 Mbps [2], whereas independent WLAN supports only a few hundred meters with a bandwidth of 11 Mbps; hence, WLAN supports highly restricted VoIP connections [3].

VoIP in WiMAX networks is one of the important applications in IEEE 802.16 that offers multimedia application services [4], [5]. VoIP codecs generate diverse data rates; furthermore, a trade-off is observed among voice quality, generated data rate, and codec complexity [6]. Five types of service class are included in IEEE 802.16, namely, Unsolicited Grant Service (UGS), real-time Polling Service (rtPS), extended real-time Polling Service (ertPS), non-real-time Polling Service (nrtPS), and Best Effort Service (BE) [7]. These previous advantages are the motivations for studying VoIP quality of service (QoS) in IEEE 802.16 WiMAX networks.

This paper begins by presenting an up-to-date literature review about VoIP over WiMax networks. Then, the challenges faced by VoIP QoS in IEEE 802.16 WiMAX networks are discussed. The rest of this paper is organized as follows. Section 1 provides important definitions related to the QoS of VoIP over WiMAX networks. Section 2 briefly discusses related works. An overview of VoIP is presented in Section 3. Section 4 describes the QoS in IEEE 802.16. Experimental work of the simulation study is discussed in Section 5. Finally, concluding remarks are provided in Section 6, in addition to a list of references.

2. Literature Review

A number of up-to-date related works for the QoS of VoIP over WiMAX networks are described in this section. With the increasing sophistication of technologies in the communications revolution, many approaches have been proposed to improve data transmission through wireless communication. Kaur et al. [8] presented a comparative study on four common service classes, namely, rtPS, ertPS,

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nrtPS, and BE. In addition, their study evaluated the performance of the Internet Protocol (IP) backbone links DS3, E3, and SONET48. A backbone link was used to connect each subnet, and three common evaluation measures were adopted to evaluate QoS, namely, throughput, jitter, and delay. The Optimized Network Engineering Tool (OPNET) 14.5 simulator was used for the design models. The results indicated that the best performance was attained when SONET48 was used as the IP backbone link. VoIP traffic can be served best with ertPS flow.

Sharma and Panjeta [9] conducted another comparative study in 2016. They investigated six scheduling algorithms: first-in, first-out; priority queue; weight fair queuing (WFQ); round robin (RR); deficit RR (DRR); and modified DRR (MDRR). The result showed that the best scheduling algorithms used in the assessment were resolved to rely on two evaluation measures, namely, the jitter and throughput. Furthermore, the most extreme activity was obtained for a specific application and for each servicing class.

In 2016, Li and Yang [10] first proposed one part of the dual-trigger handover algorithm. They selected a suitable value of handover threshold hysteresis based on signal-to-noise ratio. Furthermore, the traffic sent and received rate throughput was used to compare the performance of WiMAX and LTE in this study

In [11], a new model to improve the QoS of VoIP over WiMAX networks is presented. The performance of VoIP was evaluated based on WiMAX networks with different distances, modulations, and power values. The authors used several WiMAX parameters, such as connection statistics (load and throughput) and VoIP connection statistics (jitter, mean opinion score (MOS), and end-to-end delay). QoS was analyzed based on the long-distance data transfer between two sites with VoIP over a WiMAX network. The analysis was performed using OPNET Modeler 14.5 as the network simulator.

In [12], the authors identified network performance parameters that provided VoIP over WiMAX communication. A simulation-based network performance analysis was conducted to investigate the effects of applying different voice encoder schemes on the QoS of a VoIP system. The authors used the OPNET modeler 16.1 network simulation tool to investigate the performance of VoIP applications over WiMAX broadband access technology in a campus network. The obtained performance was compared with the analysis result of VoIP over WiMAX using performance parameters such as voice packet ETE delay, voice jitter, packet-sent packet-received, network delay, voice packet delay variation, and throughput. These parameters were used to determine the quality of VoIP calls that could be guaranteed with good QoS over WiMAX networks. The results indicated that the choice of suitable codec schemes could affect the QoS of VoIP traffic over WiMAX networks. A VoIP system exhibits the best performance under the G.711 codec scheme. Furthermore, the selection of a suitable voice encoder scheme with a small value of voice frame size per packet significantly affects VoIP traffic performance with WiMAX access technology.

In [13], the authors analyzed the performance of different VoIP codecs, such as packet end-to-end delay, MOS, and jitter, over a WiMAX network. These measures were used to evaluate the performance of various VoIP codecs. The results indicated that more delays occurred in cases with stationary nodes than those with mobile nodes. The performance of three VoIP codecs over Wi-Fi and WiMAX networks was evaluated in [14]. OPNET simulation models were designed to generate and evaluate performance metrics, such as MOS, average end-to-end delay, and jitter. The required QoS for VoIP applications in Wi-Fi and WiMAX technologies was examined. VoIP performance was simulated via six simulation scenarios using OPNET 16.0.A. MOS, average end-to-end delay, and jitter were used as performance parameters to define VoIP QoS. The G.711 codec achieved the best performance for VoIP over Wi-Fi networks, as indicated by the results.

A methodology for choosing the suitable neighbor base station (NBS) was proposed in [15]; the NBS was considered the target base station (TBS). This process helps reduce the scanning time for TBS. The results were reviewed using OPNET Modeler. The proposed methodology can reduce the waste of resources due to scanning time, as indicated by the simulation outcomes. Thus, the overall handover performance is improved. The simulation and evaluation of three networks with different sizes (i.e., small, medium, and large) were conducted in [16]. The study used networks of 15, 25, and 40 mobile workstations. In each network, a group of five WiMAX workstations connects and calls one another via one WiMAX base station (BS) for 1000 sec. QoS is indicated by the performance of parameters, such as initial ranging activity, delay, total transmission power, and physical layer path loss.

3. Background

Broadband wireless systems based on IEEE protocol 802.16 (known worldwide as WiMAX technology) are considered among the best alternatives for providing fourth- generation (4G) digital services. The objective of WiMAX is to satisfy modern communication requirements not only at home but also in schools, hospitals, airports, government agencies, industries, offices, research facilities, and particularly, when an individual moves from one place to another. WiMAX has large coverage and high transmission capacity, regardless of the time, place, mobility type, and connection time.

3.1. Overview of VoIP

Voice is an analog signal that should to be encoded on the sender side before the transmission process. Consequently, the encoding and decoding processes must be performed on the receiver side. Both processes are performed via voice codecs. Compression voice techniques are used to decrease the bandwidth consumption level [17]. Voice compression via a codec adds an algorithmic delay. Thus, a codec is anticipated to offer good voice quality even after compression, with reduced delay.

Different codecs can support voice. The most common codecs are discussed in this section. G.711, which has a stable data rate at 64 kbps, is the most widely used codec for encoding IP telephony. This narrowband codec supports frequencies ranging from 300 Hz to 3,400 Hz, which provides good quality but consumes considerable bandwidth because compression is not applied. In video conferencing applications, G.723 is used; the compression ratio of this codec exhibits the highest efficiency. G.729 is an industry standard with high bandwidth utilization for quality voice calls; this second most supported codec provides approximately the same quality as that of G.711. G.729 can be compressed to be eight times smaller than G.711, which is its main advantage over the latter [18]. These codecs are summarized in Table 1.

Table 1: Bandwidth requirement of common codecs and their description [19].

Codec	Bandwidth	Method	Algorithm Delay(ms)	Quality (MOS)	Packets Per Second	IP Packet Size (bytes)
G.711	64 kbps	PCM	0 .375	4.0	100	120
G.729	8 kbps	ACELP	35	3.9	100	50
G.723	48 to 64 kbps	CS-ACELP	97.5	3.6	33	60

In this study, we applied a G.711 encoder to a workstation and compared it with other encoders, namely, G.723 and G.729, thereby obtaining excellent results [20] [21].

3.2. QoS in IEEE 802.16

The demand for high-speed wireless Internet access has recently increased, particularly with the development of new short-range (IEEE 802.11) and long-range (IEEE 802.16) wireless technologies. Furthermore, the demand for BWA networks has increased significantly among customers who require high QoS. Certain features are included in the current standard. VoIP is a real-time application that is sensitive to packet loss, jitter, and delay. Therefore, VoIP design is important in effectively ensuring the delivery of packets in real time. The performance of VoIP in WiMax networks will be vulnerable to deterioration because these networks use broadband technology.

In any WiMAX model, five types of service class, namely, BE, UGS, rtPS, ertPS, and nrtPS, are proposed to ensure QoS. In the current study, we designed a WiMAX network that uses several scenarios, with each scenario depending on specific parameters. Then, we compared these scenarios for identifying and selecting parameters that could achieve the best QoS for voice over this type of network [22].

Service classes have two main categories. The first category includes nrtPS and BE, whereas the second category includes UGS, rtPS, and ertPS. In the first category, WFQ and the alteration of RR may be applied as a common scheme, in which it is not affected by delay and jitter. Moreover, real-time services place numerous constraints on the parameters, thereby making scheduling difficult in achieving delay limits and tolerating delay and jitter with maximum throughput [23].

- UGS

Constant bandwidth division on the periodic base is provided by this service class. Only one established connectivity is required. The use of UGS aims to develop real-time data streams that contain constant-sized data packets at periodic intervals, such as VoIP without silence suppression. No contention request is made in UGS, and no clear bandwidth request is made by the subscriber station (SS). The BS provides constant-sized access slots at periodic intervals to UGS flows. However, the UGS bandwidth is wasted during inactive voice calls [24].

- rtPS

This service class is used to develop real-time data streams, including variable-sized data packets that run at periodic intervals, such as MPEG videos. The advantage of rtPS is reducing reserved traffic rates. The flows of rtPS are voted by BS via unicast request voting, which occurs sufficiently frequently to meet the delay requirements of service flows [25].

- ertPS

This QoS type supports VoIP along silence suppression. No traffic transmission will occur during the silent period. The QoS parameters are the same as those for the UGS service type. The application of ertPS is provided in Table 2. This service class is similar to UGS in that the BS assigns MST during active mode and no BW is allocated during the silent period. In this study, the BS needs to poll mobile subscribers to determine whether the silent period has ended [26].

- nrtPS

This type of service class is used for non-real-time variable bit rate traffic. It supports no delay guarantee, and minimum rate is ensured. This service class develops delay-tolerant real-time data streams for the required variable-sized data packets with minimum bit rate. In addition, nrtPS flows act as rtPS flows and are voted via transmission request voting. Therefore, rtPS flows can also receive a few request voting opportunities during network overcrowding. Furthermore, contention requests are allowed to be used [27].

- BE

This service class is designed to support data streams that do not require a minimum service level. The flows of BE are allowed to use contention request opportunities. The BE service class uses any remaining bandwidth after the required bandwidths of all the other classes have been allocated. This service is used in many applications, such as in emails and the File Transfer Protocol (FTP). These applications do not require strict latency.
The QoS classes are summarized in Table 2. In this study, we use UGS and ertPS, as discussed in the subsequent section [28].

3.3. Simulation Module: OPNET

Procedure simulation is commonly used as a method to analyze and evaluate the performance of wireless networks. In this study, we used OPNET Modeler version 15.4A to measure and analyze the performance of a WiMAX network while implementing VoIP services. OPNET offers an extensive area for modeling different communication networks and service systems with significant simulation accuracy [29]. OPNET Modeler is an analytical model that can simulate devices, protocols, and behavior; it has approximately 400 special-purpose modeling functions. The OPNET environment provides three main operational phases: data collection, model design, and data analysis. OPNET provides high leverage for modeling distributed systems. It includes a set of graphical editors that can be used to design the details of networks, nodes, processes, and models [30].

Table 2: QoS classes in IEEE 802.16.

Service class	Description	Application
UGS	Maximum continuing for an extended period or without interruption rate. Maximum latency Tolerance Jitter Tolerance	Video VoIP Conference
ertPS	Maximum reserved rate Maximum continuing for an extended period or without interruption rate Maximum latency Tolerance Traffic priority	VoIP with silence suppression
rtPS	Maximum reserved rate Maximum continuing for an extended period or without interruption rate Maximum latency Tolerance Traffic priority Jitter rate	Streaming Audio and video
nrtPS	Maximum reserved rate Maximum continuing for an extended period or without interruption rate Maximum latency Tolerance Traffic priority	FTP
BE	Maximum continuing for an extended period or without interruption rate Traffic priority	HHTP Web browsing

3.4. Experimental Parameters

The performance of the system was measured using different QoS parameters, including throughput, delay, jitter, data loss, and MOS. These parameters and their minimum QoS requirements are summarized in Table 3.

Table 3: Measured parameters with QoS

Parameter	Definition	Measurement Unit	QoS requirement
Throughput	Total data transferred from one node to another	Bytes/second	n/a
End- to –End delay	Total delay for data transfer to occur	Millisecond	<=140 ms
Percent data loss	(Packets sent –packets received)/packet sent	Percent	<1%
Jitter	Variation in packet arrival	Millisecond	<=0.5 ms

- Throughput:

Throughput is a measure for computing the actual speed of sending data through a network. This measure is defined by the following equation:

$$\text{Throughput} = \frac{\text{Number of delivered packet} * \text{packet size} * 8\ \text{bits}}{\text{The total duration of the simulation}}$$

- Delay (latency):

Delay is a measure for computing the time consumed in transferring an entire message to its destination, starting from the time that the first bit is sent out from the source.

$$\text{Delay} = \text{propagation time} + \text{tranmission time} + \text{queuing time} + \text{processing delay}$$

- Jitter:

Jitter is a variation of the delay time of sequent packets. Jitter significantly affects real-time, delay-sensitive applications, such as voice and video. A small amount of jitter may be acceptable, but if its amount increases, then delay-sensitive applications will be adversely affected and will become useless. Jitter is measured by computing the difference in the delay of packets over a certain period.

$$\boldsymbol{Jitter} = \boldsymbol{Max}_{\boldsymbol{1 \le i \le n}} \{[\boldsymbol{t'(n)} - \boldsymbol{t'(n-1)}] - [\boldsymbol{t(n)} - \boldsymbol{t(n-1)}]\}$$

Where: t(i) is the time transmitted by the transmitter, and
t'(i) is the time received at the receiver.

- MOS:

MOS is a measure for computing voice quality in VoIP with a value that ranges from 1 to 5. As shown in Table 4, a value of “1” indicates the worst quality, whereas a value of “5” indicates the best quality.

Table 4: MOS.

Quality scale	Score	Listening effort scale
Excellent	5	No effort required
Good	4	No perceivable effort required
Fair	3	Moderate effort required
Poor	2	the large effort required
Bad	1	No meaning understood with effort

$$MOS = 1 + 0.35 * R + 7 * 10 - 6[R(R-60)(100-R)]$$

where R = 100 – Is – Ie – Id + A

Is: effect of impairments that occur with the voice signal;

Is: impairments produced by different types of loss;

Id: impairments produced by delay, particularly mouth-to-ear delay.

The guidelines for voice quality measurement for both end-to-end delay and jitter are presented in Table 5. These measures are provided by the Telecommunications Standardization Sector of the International Telecommunications Union. To produce a good-quality voice call, the range of delay must be from 0 to 150 ms, with a jitter between 0 and 20 ms. Voice quality is considered poor when the delay is longer than 300 ms or jitter is higher than 50 ms, as described in Table 5. Voice quality can be acceptable if the delay is between 150 ms and 300 ms or jitter is between 20 ms and 50 ms [31].

Table 5: Voice quality guidelines.

Network parameter	Good	acceptable	Poor
Delay(ms)	0-150	150-300	>300
Jitter(ms)	0-20	20-50	>50

4. Proposed Model

4.1. WiMAX Modulation Techniques

The main objective of this work is to study and analyze the performance of VoIP over WiMAX networks. A second objective is to study the WiMAX (IEEE 802.16e) standard with QoS features. Accordingly, we briefly reviewed VoIP technology and its main features to identify the best methodology for designing a model. Furthermore, we simulated the WiMAX network model by using different scenarios that addressed QoS classes, number of users, and voice codecs.

4.2. Experimental Network Configuration

The configuration details of the simulation components are described in this section. These components include the WiMAX network model, BSs, and SSs. The WiMAX network model used in the simulation study is presented in Figure 1. The network consists of seven coverage cells that are controlled by a WiMAX node. Each cell contains one BS and five SSs, as indicated in Table 6. The BSs are linked to the core network via an IP backbone. In turn, the IP backbone is linked to the server backbone via an access service network (ASN) gateway that supports mobility in the WiMAX network. The IP backbone, server backbone, and ASN gateway comprise the service provider network.

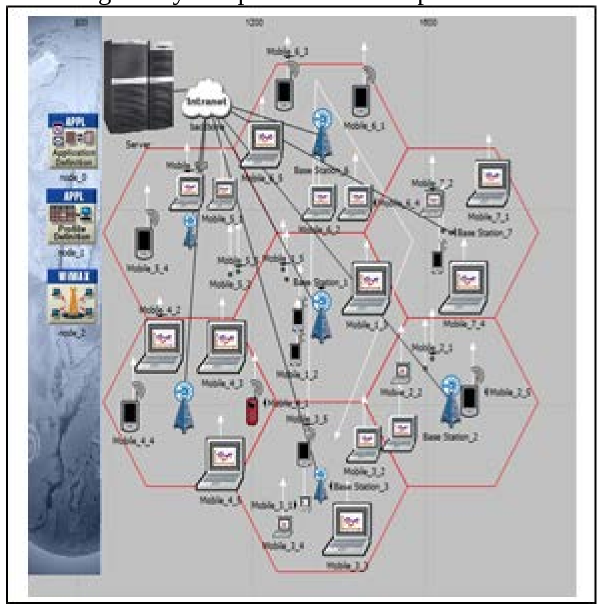

Fig. 1 Network Model For WIMAX.

The cell radius in the simulated network is 200 m and voice calls with public switched telephone network quality are configured among mobile nodes. Transmission power is set to 0.5 W, with reference to the subscriber node. Experimental work was performed by using a PC with the software and hardware specifications provided in Table 7.

Table 6.Network Configuration Details.

Component	Value
Network	7 cells
Cell Radius	0.2 km
No. of Base Stations	7 station
Simulation time	240 s
No. of Subscriber Stations per BS	5 station
Base Station Model	wimax_bs_ethernet4_slip4_router
Subscriber Station Model	wimax_ss_wkstn
IP Backbone Model	Router_slip64_dc
Voice Server Model	ppp_server
Link Model (BS-Backbone)	ppp_adv
Link Model (server – Backbone)	ppp _sonet_oc12

Table 7. PC software and hardware specifications

	Component	Specification
Hardware Setup	Processor	Intel Core i5-2450M CPU @ 2.50 GHz
	RAM	6.00 GB
	Hard disk	500 GB
Software Setup	Operating System	Microsoft Windows7 operating system (64- bit)
	Simulation modeler	OPNET Modeler (OPNET version 14.5.A).

5. Results Analysis and Discussion

The analysis results of VoIP QoS in WiMAX networks are presented in this section. Four simulation scenarios were built (Default, 1, 2, and 3). Each scenario used different IEEE 802.16 QoS classes, modulation techniques, and initial QoS parameters. The performance results of VoIP in WiMAX networks were categorized based on simulation scenarios and compared in terms of three voice codecs, namely, G.711, G.723, and G.729.

The major challenges that are experienced by current WiMAX networks include available bandwidth, network capacity and load, QoS classes, and data traffic. We built three scenarios to analyze different aspects of a WiMAX network that affect VoIP traffic. The simulated WiMAX network selected in the previous section was used to generate four simulation scenarios to analyze the performance of VoIP over a mobile WiMAX network.

5.1. Default scenario

Three voice codecs (G.711, G.723, and G.729) were used to generate different voice traffic over the WiMAX network. An important capability of a WiMAX network is throughput, which represents the total data traffic in bits/s forwarded from the WiMAX layer to higher layers in all the nodes of the network [32]. The following result compared all the aforementioned voice codecs to evaluate their performance. Codec G.723 achieved the best performance based on throughput, MOS, and jitter as shown in figure 2 and figure 3. In addition, G.723 is better than G.711 and G.729 because it has the shortest delay and jitter and the highest MOS and throughput.

The average value of throughput (in packets per second) for VoIP application in WiMAX networks is shown in Figure 2. As shown in the figure, the highest value of voice codec occurs in G.723.

Fig. 2 Relative Throughput of Voice Codecs (packet /s)

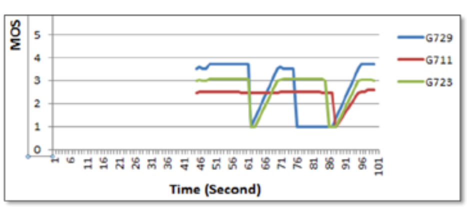

Fig. 3 Average MOS value of voice codecs (bits/s)

Figure 4 shows the average value of throughput in packet per second for VoIP application in WiMAX network. As shown in Figure 4, the highest value of voice codec occurs in G723.

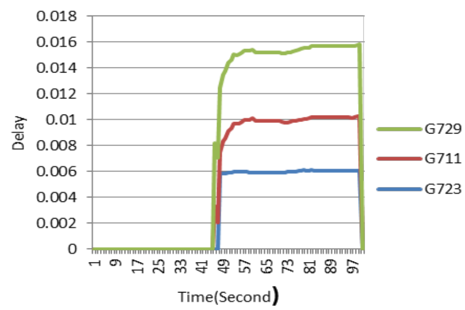

Fig. 4 (a)

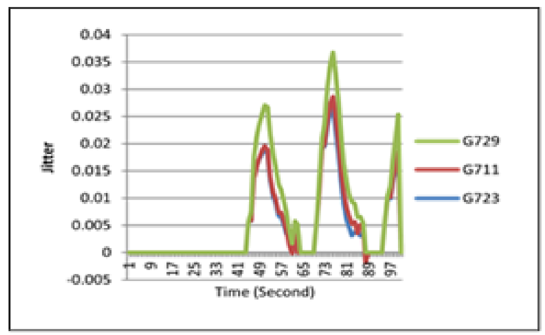

Fig. 4 Jitter and Delay in WiMAX connection using default scenario (packet/s)

MOS is one of the most important performance metrics in VoIP. The comparative results of the average MOS values for the three codecs used in the default scenario are provided in Figure 5. The MOS value is the highest for the voice codec G.723, thereby indicating that this codec will provide better speech quality compared with the other two codec schemes (G.711 and G.729).

The comparative results of voice jitter and delay for the three codecs (G.711, G.723, and G.729) are illustrated in Figure 6. The jitter value in G.711 and G.723 traffic is acceptable because its average remains less than 30; this value is classified as good based on the standard values of the performance metrics described in Table 5. The delay value is the highest in voice codec G.723, thereby showing that this codec will provide the smallest value compared with the other two codecs schemes (G.711 and G.729). Table 5 shows that G.729 obtained the worst value.

The simulation results indicated that G.723 was dominant and outperformed the other two codecs in terms of throughput, delay, and MOS performance parameters. Thus, G.723 is considered more desirable for VoIP applications. We assumed only one codec, i.e., G.723, for

Scenarios 1, 2, and 3 because this codec has been widely used in previous studies.

5.2. First Simulation Scenario (scenario1)

In Scenario 1, VoIP performance was studied with regard to WiMAX QoS configurations. To identify the best class for carrying VoIP, several WiMAX service classes, namely, BE, rtPS, and UGS, were verified. The number of stations (MS) is 35 nodes, as described in the configuration network (Table 6). This scenario used only one codec, namely, G.723, and compared its result with the findings of previous studies that had adopted the same environment.
Scenario 1 used ertPS, nrtPS, BE, rtPS, and UGS for uplink and downlink directions. Service class was set to gold, and the bit rate was 64 kbps. The initial modulation used was quadrature phase-shift keying with an initial coding rate of (½

5.3. Second Simulation Scenario (scenario2)

This scenario adopted the same five service classes used in Scenario 1. However, Scenario 2 improved voice by resizing the rate of the gold service class to 64 kbps, in which the initial modulation mechanism was 64–quadrature amplitude modulation (QAM) ¾. As shown in Figure 7, the throughput of VoIP traffic that uses G.723 is described through Scenarios 1, 2, and 3.

5.4. Third Simulation scenario (scenario3)

This scenario is a duplication of the previous scenario (i.e., Scenario 2). However, the value of service flows changed to (96000). The initial modulation mechanism was 64–QAM ¾, which is similar to that in Scenario 2. Through this modification, traffic in this scenario became more resilient and delay was reduced. The throughput result after Scenario 3 was used increased traffic resilience and decreased delay.

- Results of improved Scenarios

The simulation results were analyzed based on four major factors: throughput, delay, jitter, and MOS. The following results describe the previous major factors for Scenario 1, and consequently, for Scenarios 2 and 3.
Scenario 1 has been designed to determine the best WiMAX service class that can attain the highest performance. To compare the different service classes, the data gathered from the five service classes are presented in the same figure. Figures 5, 6, and 7 show the comparative results based on the proposed design in Scenario 1, as described in Section 5.2.
The throughput for UGS flow is the highest among the five service classes because this service class is designed by considering constant bit rate traffic. The throughput for rtPS traffic is better than that for BE traffic. ertPS and nrtPS achieved the same result as rtPS. These values do not appear in the plot shown in Figure 5.

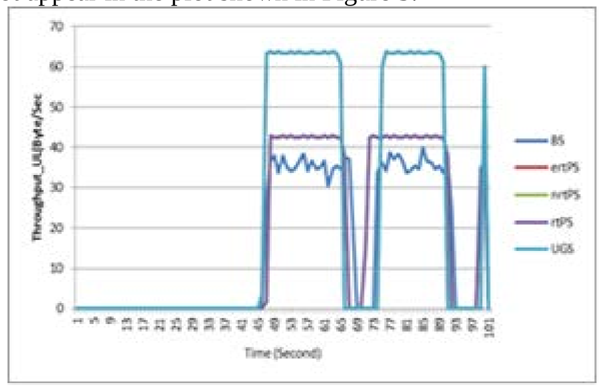

Fig. 5 Throughput in WiMAX connection using scenario1

Figure 6 shows the MOS values for all five service classes. UGS achieved the highest MOS value. The result of rtPS is close to that of UGS, which is slightly better than that of BE. The MOS values for rtPS, ertPS, and nrtPS are the same, and thus, they do not appear in the plot.
Figure 7 presents the average jitter and delay for the five service classes. The figure shows that the rtPS service class exhibits the highest jitter. The average jitter for UGS attains the lowest result, with an extremely small value. UGS has low jitter values, which are very close to those of nrtPS. By contrast, the delay for UGS achieves the highest result, whereas that for rtPS obtains the lowest result.

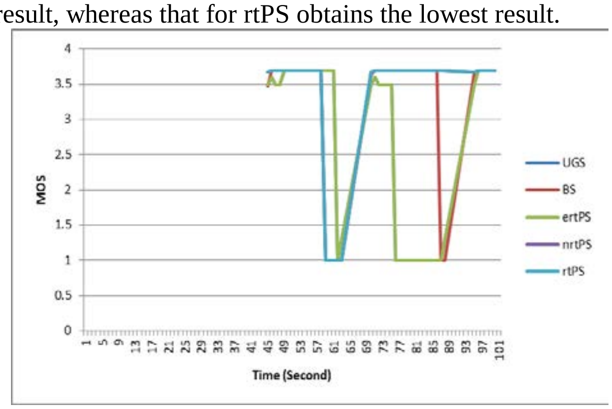

Fig. 6 MOS in WiMAX connection using scenario1

For Scenario 2, the throughput for UGS flow is the highest among the five service classes because this service class is designed by considering constant bit rate traffic. The throughput for rtPS traffic is better than that for nrtPS. Lastly, ertPS and BE attain the worst results, as shown in Figure 8.

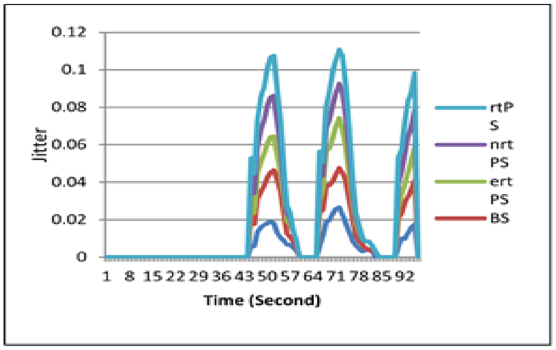

Fig. 7 Jitter and Delay in WiMAX connection using scenario1 (packet/sec)

Figure 9 shows the MOS values for the five service classes. BE scored the highest MOS value. The result for rtPS is close to those for the remaining service classes. BE and rtPS have slightly better MOS values than UGS. Furthermore, the MOS values for rtPS, ertPS, and nrtPS are the same. Therefore, these results do not appear in the plot.

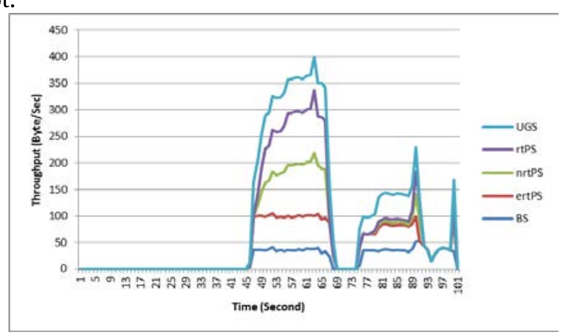

Fig.8 Throughput in WiMAX connection using scenario2

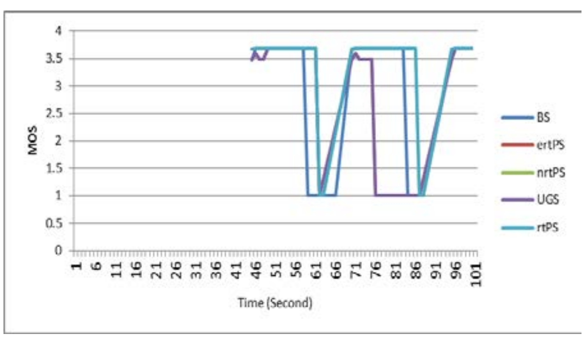

Fig.9 MOS in WiMAX connection using scenario2.

Figure 10 shows the average jitter and delay for the five service classes. The figure shows that rtPS achieves the highest jitter. The average jitter for BE obtains the lowest result, with an extremely small value. UGS has high jitter values, which are extremely close to those for rtPS. By contrast, the delay results for UGS are the highest, whereas those for BE are the lowest.

For Scenario 3, the throughput for UGS flow is the highest among the five service classes. The throughput for rtPS traffic is better than those for the remaining service classes. ertPS and BE obtain the worst results, as shown in Figure 11.

Figure 12 shows the MOS values for the five service classes. nrtPS, rtPS, and BE achieve the same results and have scored the highest values. The result for ertPS is close to, but slightly lower, than that for UGS. Furthermore, the MOS values for rtPS, BE, and nrtPS are the same, and thus, these values do not appear in the plot.

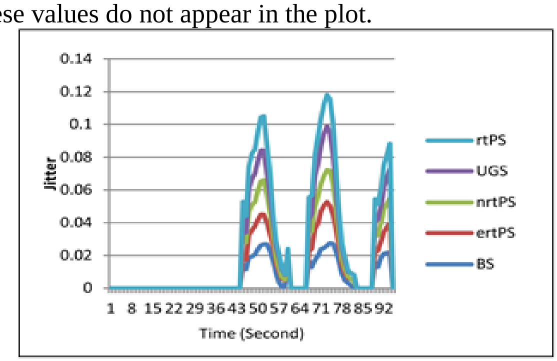

Fig. 10.a

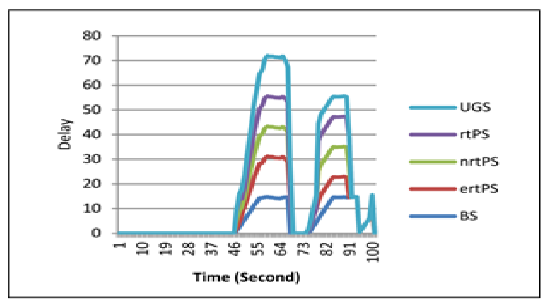

Fig.10.b Jitter and Delay in WiMAX connection using scenario2

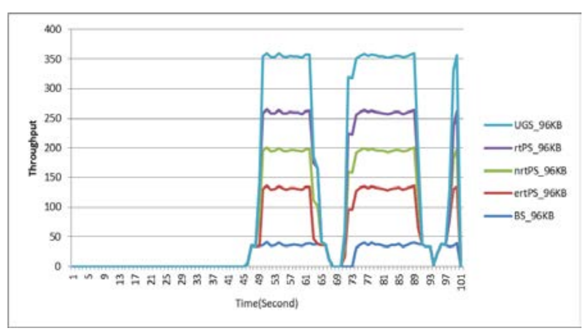

Figure.11: average throughput in WiMAX connection using scenario3

Figure 13 presents the results of jitter and delay for all five service classes. The figure shows that rtPS and UGS have the lowest jitter. The average jitter for BE obtains the worst result given its high value. Moreover, UGS has low jitter values, which are extremely close to those for rtPS. By contrast, UGS and BE have the lowest delay, whereas ertPS achieves the maximum delay.

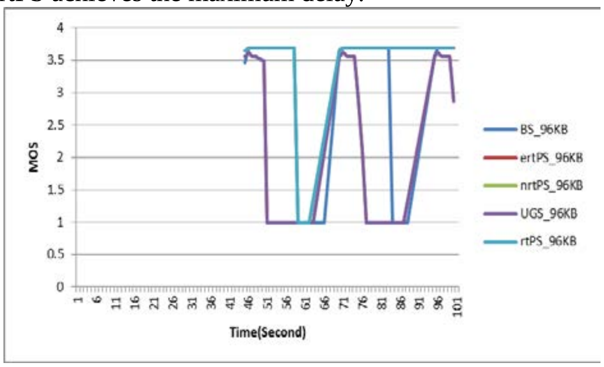

Fig.12 Average MOS in WIMAX connection for scenario3

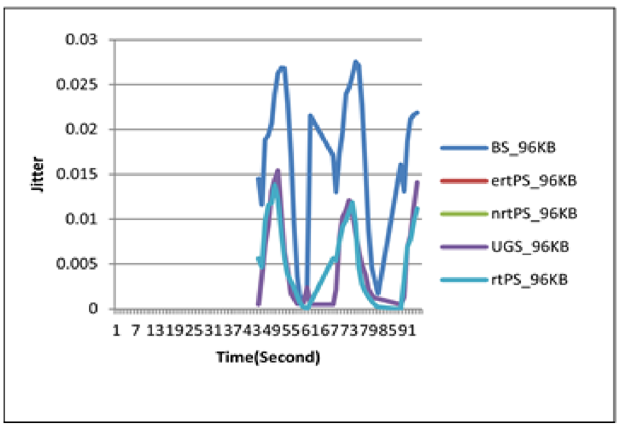

Fig.13.a

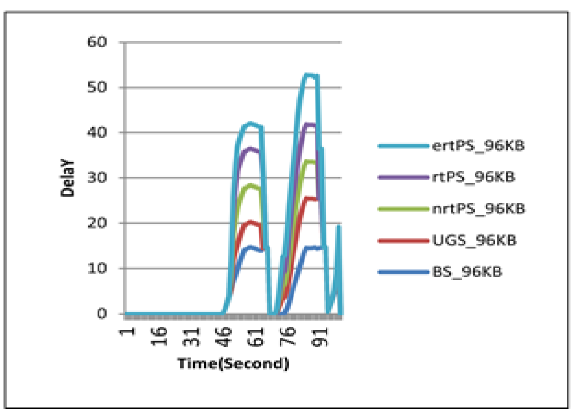

Fig.13.b Jitter and Delay in WiMAX connection using scenario3.

Finally, the proposed method achieved acceptable results when these results were compared with those of previous studies on the QoS of VoIP over WiMAX networks. Table 19 provides a comprehensive comparison among studies that did not use numerous methods to improve the QoS of VoIP over WiMAX networks.

Table 8. Comparison between previous studies and the present study.

#	Reference	Approach	VoIP performance metrics	Best result	Simulation module	Network environment
1	[8]	Effects of different IP backbone links and service flows on QoS parameters, in addition to the service flow of ertPS and the bandwidth in nrtPS	Throughput Utilization Delay Jitter	SONET48 achieves the best value compared with those of DS3 and E3.	OPNET	Seven cells are in each subnet with ten nodes in each cell.
2	[33]	Simulation and analysis of service classes using different VoIP codecs in terms of throughput, average jitter, and average delay	Throughput Average delay Jitter	UGS has the best performance parameters for serving VoIP.	NS-2	Number of mobile nodes with VoIP traffic varies from 2, 4, 6, 8, and 10.
3	[34]	Assessment of the best effort of WiMAX service flow for different VoIP codecs by considering various performance parameters	Throughput Delay Jitter	Throughput of G.723 is low compared with the throughput values of G.729 and G.711. Average delay and average jitter are nearly zero for G.723.	NS-2	Number of mobile nodes with VoIP traffic varies from 2, 4, and 6.
4	[35]	Evaluation of the performance of WiMAX for supporting VoIP traffic and analysis of important critical parameters for various codecs	MOS End-to-end delay Jitter Packet delay	G.723 is better than G.711, G.726, G.728, and G.729 because it has lower delay and higher MOS, received traffic, and throughput.	OPNET	Not available
5	[36]	Evaluation of the performance of VoIP over WiMAX networks. Different parameters, such as jitter, MOS value, packet end-to-end delay, and sent and received packets, were used to measure performance.	Throughput Delay Jitter	Results show that VoIP performs best under G.711 compared with under G.723 and G.729.	OPNET	WiMAX network consists of seven cells with a radius of 0.2 km and five SSs.
6	Present study	Improvement of the QoS of VoIP over WiMAX networks using OPNET Modeler	Throughput MOS Jitter Delay	Research outcomes present that G723 is the best codec used, and VoIP applications can perform better under UGS service class.	OPNET	Network consists of seven cells with a radius of 0.2 km; every cell has one BS and five SSs.

5.5. Comparison with Previous Studies

Studies 2 and 3 used Network Simulator 2 (NS-2) to design their network, whereas Studies 1, 4, 5, and the present study used OPNET as shown in table 8. These studies indicate that the number of cells in a network significantly influences the selection of the best scenario to achieve improved performance.

6. Conclusions and Recommendations

In this study, the performance of VoIP over WiMAX networks was evaluated and analyzed based on several parameters, including jitter, MOS, end-to-end delay, and sent and received packets. Subsequently, G.711, G.723 and G.729 voice codecs were simulated to identify the most suitable voice codec for VoIP over WiMAX networks. The results indicated that VoIP performed best under G.723 compared with under G.711 and G.729. VoIP applications could perform better under UGS service class, as evident in the research outcomes. The application of a UGS connection improved traffic resiliency to a high level, thereby reducing delay rate and maximizing throughput. Finally, we present several recommendations to improve the results of this study, such as the implementation of these scenarios with other traffic types, including HTTP, video on demand, and FTP. Furthermore, we can apply the five service classes for VoIP with silence suppression, in addition to implementing the proposed scenarios with mobile WiMAX.

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