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Comparisons of QoS in VoIP over WIMAX by Varying the Voice codes and Buffer size

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Abstract: Voice over Internet Protocol (VoIP) is developed for voice communications system based on voice packets transmitted over IP network with real-time communications of voice across networks using the Internet protocols. Quality of Service (QoS) mechanism is applied to guarantee successful voice packets transmitted over IP network with reduced delay or drop according to assigned priority of voice packets. In this paper, the goal of simulation models is present to investigate the performance of VoIP codecs and buffer size for improving quality of service (QoS) with the simulation results by using OPNET modeler version 14.5. The performance of the proposed algorithm is analyzed and compared the quality of service for VoIP. The final simulated result shows that the VoIP service performance best under G.729 voice encoder scheme and buffer size 256 Kb over WiMAX network.

Keywords: VoIP, Codecs, QoS, WiMAX, Buffer size

1. INTRODUCTION

Worldwide Interoperability for Microware Access (WiMAX) is a standard based on IEEE 802.16 broadband wireless access metropolitan area technology. It is an air-interface standard for microwave and millimeter-wave band. This server can act as a wireless extension cable and DSL technology, enabling wireless broadband access. The signal cover of WiMAX technology ups to 50km, WiMAX data rates between 1.5 to 75 Mbps. Also, it supported multimedia applications such as voice over IP (VoIP)

VoIP is developed for voice communications system based on voice packets transmitted over IP network, in possibility to reduce a communication costs. It provides real-time communications of voice across networks using the Internet protocols with Quality of Service. QoS transmitted over IP network which can reduced delay or drop according to assigned mechanism is applied to guarantee successful voice packets priority of voice packets.

In the present study, OPNET simulator is use to implement the proposed VoIP Network. We examine all the various buffer size that can drop the quality of service over wireless network (WiMAX). Accordingly to this study, it is relied on using codecs and buffer size to explore its impact on packet delay, jitter and throughput, are calculated and analyzed. The comparisons were carried out between different codecs (G.711, G.723, G.729 and GSM) and different buffer size (32kb, 64kb, 128kb and 256 kb) with are the most appropriate to improve QoS for VoIP.

This article is organized in five sections. Section 1 introduces the work. Section 2 gives the Materials and Methods for improving quality of service (QoS). Section 3 shows the results and analysis based on the modeling and simulation study. Section 4 describes the experimentation carried out in this work. Section 5 concludes the paper.

2. Materials and Methods

2.1 Quality of Service (QoS)

One can broadly divide QoS in two types: QoS for network and QoS for user. QoS for network guarantees that the packet for the voice communication shall not be delayed or dropped. A QoS for user corresponds to the degree of user satisfaction in service. These parameters are explained below:

Delay Takes place when the packets of data that contain the voice in digital form take more than estimated time in order to reach the destination. Delay can be caused by a number of factors, in factors, including, type of network, queuing discipline and type of voice packet traffic [1].

Jitter take place while transporting the voice and packet over switched network, the data may have a time variation in order to reach the destination. When some of data packets take more time in order to reach the destination the effect of this variation shall result into a jitter, for the listener at the destination [1].

Throughput Shall take place when the total received packets is given to each traffic class and measured as the mean of the number of packets produced per unit time. Throughput is inversely proportional; robust network has a lower degree of packet drop [1].

2.2 Codecs for VoIP

Codecs are the algorithm that is used to convert voice data format from analog to digital in VoIP process. In VoIP process, when a user talks using telephone or microphone, the voice format is first converted into digital format, compressed and then encoded into a predated format using codecs. Codecs is vary in the sound quality. There are many types of codecs developed and standardized by ITU-T such as G.711, G723 and G.729 for this purpose [2]. Consequently, packets are transmitted through IP network to the destination. At the destination, the digital form is converted back into the voice form.

G.711 is defined in ITU-T standard for speech codec. It delivers precise speech transmission and takes very low processor requirements. It employs pulse code modulation (PCM) or Analog-to-Digital Converter (ADC). The PCM sampling rate for the voice is 8000 frequencies per second; with a tolerance rate voice bandwidth of 4000 Hz. PCM processed samples are represented in 8 bit format, and with a high bit rate of 64 Kbps [3]. In addition, there are two versions of G.711, namely, A-law and U-law. A-law is designated for computer processing and its sample rate encode is 13 bit samples, this E1 standard is used in most of the rest of the world (other than North America and Japan). U-law is the T1 standard used in North America and Japan. Its rate encode is 14 bit samples [3]. Codec G.711 provides a higher signal range and it is the codec used by the PSTN network and is believed to be good for VoIP.

G.723 is based on the ITU-T standard that was designed for voice and multimedia communication over stand phone system. It gives high compression with high quality audio. It uses lots of processor power; these particulars are specified by the H.323 and H.324 series standards. It provides two compressed stream bit rate 5.6 Kbps and 6.3 Kbps. The higher bit rate is indicate greater quality. The code operates on speech frames of 30 ms corresponding to 240 samples at a sampling rate of 8000 voice frequencies per second [4]. It has been optimized to represent high quality speech with low bandwidth requirements using a limited amount of complexity and suitable for applications such as VoIP.

G.729 is another ITU-T standard and it has the ability to compress the payload for low bit rate by using an algorithm know as Conjugate Structure – Algebraic Code Excited Liner Predication (CS-ACELP). It gives excellent bandwidth utilization and is error tolerant. This coder offers good quality speech at a reasonably low bit rate of 8 Kbps and works on a frame of 80 speech samples [5]. It allows moderate transmission delay and is very useful in applications such as teleconferencing or visual telephony where quality, delay and bandwidth are important. Nowadays Skype is taking benefits from this standard.

GSM stands for Global System for Mobile communication this based on the codec operating with a bit rate of 13 kbps. The GSM codec provides good-quality speech. The speech input is a 16 bit word sampled at 8 KHz is analyzed by LP.

2.3 WiMAX Network

Worldwide Interoperability for Microware Access (WiMAX), is a standard based on IEEE 802.16 broadband wireless access metropolitan area technology. It is an air-interface standard for microwave and millimeter-wave band. This server can act as a wireless extension cable and DSL technology, enabling wireless broadband access. The signal cover of WiMAX technology ups to 50 km, WiMAX data rates between 1.5 to 75 Mbps. Also, it supported multimedia applications such as voice over IP (VoIP) [6]. WiMAX support its application through four distance traffic classes:

Best Effort (BE) is designed for application such as web browsing [7] that do not require QoS.

Non Real-Time polling service (nrtPS) support non real-time application such as File Transport Protocol (FTP) [7] that requires variable size of data.

Unsolicited Grant service (UGS) supports Constant Bit rate (CBR) application such as VoIP without silence suppression [8] where Base Station (BS) assigns a fixed bandwidth to users.

Real-time Polling service (rtPS) supports real-time applications with variable size data such as MPEG [8] where BS allocates bandwidth based on Subscriber Station (SS) request.

2.4 Buffer size

Buffer is memory location within router where packets are placed in queues before they get processed upon their turn [9]. The intermediate devices like router and switch in a network have buffer where the packets wait in a queue before and after processing. Depending on the packet arrival rate and the packet departure rate, which may be higher or lesser then packet departure rate, the packet size may have an impact in the percentage of discarded packets. As multiplexing increases the packet size, big packets are expected to be discarded in the bigger percentage than small ones.

3. MODELLING AND SIMULATION

3.1 OPNET Modeler

OPNET (Optimized Network Engineering Tool) is a tool to simulate the behavior and performance of VoIP network, Quality of Service (QoS) analysis of and performance of VoIP network, Quality of Service (QoS) analysis of simulator of network communication and network device and protocols. OPNET provides performance analysis of computer network and applications [10] through this we can design;

3.1.1 Simulation Model

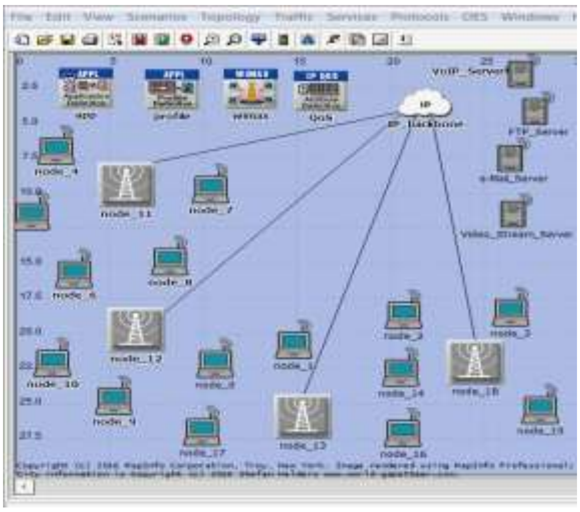

Figure. 1 The Simulation Network Model

The following figure 1 present the network model. This simulation model was run in different scenarios to determine the best audio encoding schemes and buffer size of utilizing VoIP over integrating wireless (WiMAX). All the scenarios follow the similar structure and the similar topology. Each scenario is implementing with the codec G.711, G723, G729 and GSM furthermore buffer size such as 32kb, 64kb, 128kb plus 256 kb. Various comparisons are conduced to fine the value of various parameters.

3.1.2 Simulation Parameter Setup

VoIP in Fixed WiMAX network Base Station (BS) were simulated with fifteen (15) mobile devices, where mobile devices subscriber's stations are place around each BS. All BSs were connected to the IP back bone (internet) using point-to-point protocol (PPP) without any server BS. Basic parameters associated with VoIP in WiMAX Configuration attributes, application's configuration, application profiles, task's definition, BSs and SSs for the model were configured as show in figure 1.

Table 1. Subscriber Station Parameters

Parameter	***Value***
Antenna Gain (dBi)	-1 dBi
Type of SAP	IP
Match Value	Interactive Voice (6)
Server ice das s Name	Gold
Max Transmission Power	Adaptive
PHY Pro file	0.5 W
PHY Profile Type	Wireless OFDMA 20 MHZ
Multipatch Channel Mode	OFDM
Patholoss Model	Vechicular Emifonrnaits
Terrain Type	Terrain Type A

Table 2. Base Station Parameters

Parameter	***Value***
Antenna Gain (dBi)	15 dBi
Match Value	Interactive Voice (6)
Server ice das s Name	Gold
Max Transmission Power	Adaptive
Max Transmission Power	Adaptive
PHY Profile Type	Wireless OFDMA 20 MHZ
Multipatch Channel Mode	OFDM
Multipatch Channel Mode	OFDM
Perm Has e	3

4. Simulation results and Discussion

This paper investigates the performance of WiMAX network using different quality of service (QoS) with are explained below.

4.1.1 Quality of Service (QoS)

Quality of Service (QoS) represents the whole performance of a WiMAX network, witness by the users of the network. To evaluation the quality of service, various related aspects of network service are often considered, for example error rates, bandwidth, throughput, load, transmission delay, availability, jitter etc.

4.1.2 Performance Parameters

The performance parameters are used to analyze simulation with based on the simulation results; a comparison between the effects of different codec G.711, G723, G729 and GSM as well buffer size such as 32kb, 64kb, 128kb the last 256 kb on QoS of VoIP. As stated earlier, three QoS measurements, such as voice packet end to end delay (sec), voice packet jitter (sec) and throughput (packet/sec).

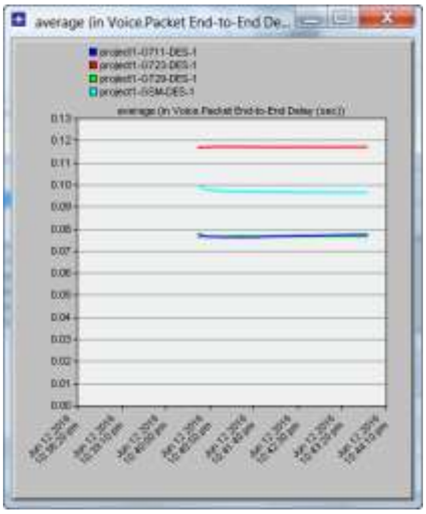

Figure. 2 Delay (sec) under various audio codes

Average end to end delay metric is show in figure 5: G711 present the best performance with respect to the other codes.

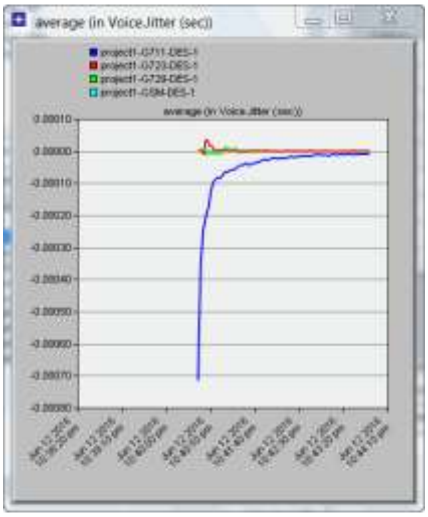

Figure. 3 Jitter (sec) under various audio codes

Figure 3 describe the average voice jitter comparison using different codecs. From graph, the jitter of G.711, G.723, G.729 and GSM increased and become very close to zero.

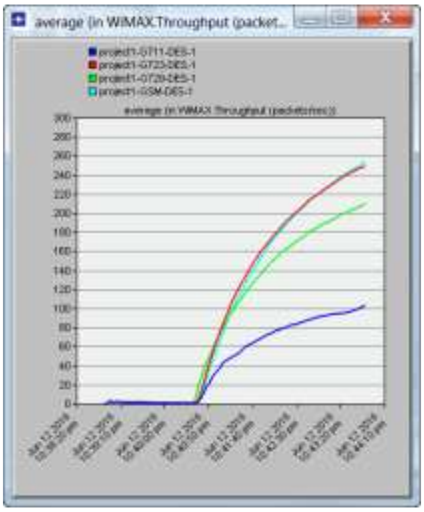

Figure. 4 Throughput (sec) under various audio codes

In figure 4, network scenarios indicated that G.729 scenario is the best in traffic send and receive in comparison with other scenarios.

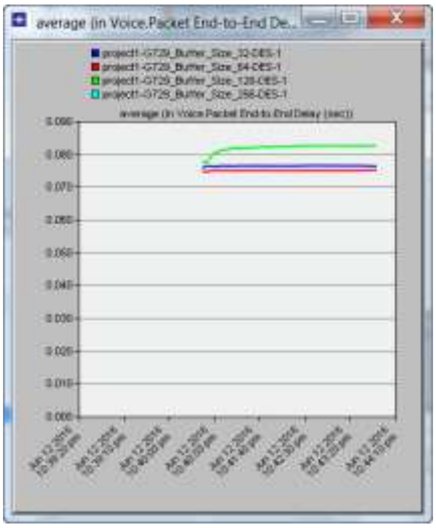

Figure. 5 Delay (sec) under various buffer size

In figure 5, it is presented in average end to end delay metric with various buffer size: 64 Kbits present the best performance with respect to the other buffer size. While high buffer size (128 kbits and 256 kbits) increased delay value.

Figure. 6 Jitter (sec) under various buffer size

The performance analysis is illustrated in figure 6. The investigations present that buffer size 256 Kbits increased and become much closed to zero.

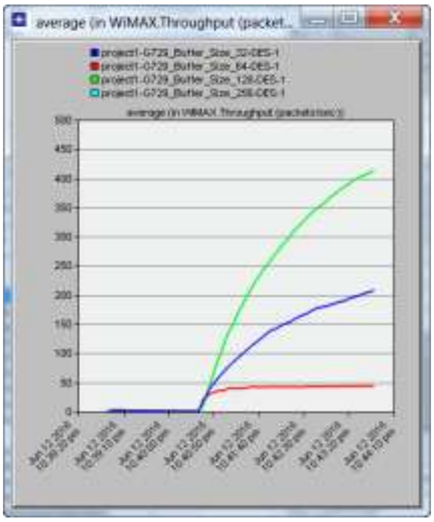

Figure. 7 (sec) under various buffer size

The simulated voice throughput in figure 7, it is observed that both 128 Kbits and 256 Kbits have same throughput at 400 (packet/sec) and this value is more than buffer size other (32 Kbits and 64Kbits).

5. CONCLUSIONS

In this paper various performance of QoS such as Jitter, Delay and Throughput, are analysed on VoIP codes and different buffer size with the help of the observation obtained from different codes and buffer size, it was found that as the no. Of buffer size increases, the value of QoS parameters (Delay and Throughput) also increases; an optimized value of QoS parameter is obtained.

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