Simulative Investigation of QoS parameters for VoIP over WiMAX networks

Priyanka [1], Jyoteesh Malhotra [2], Kuldeep Sharma [3]

[1,3] Department of Electronics, Ramgarhia Institue of Engineering and Technology, Phagwara, India

[2] Department of Electronics & Communication, Guru Nanak Dev University Regional Campus, Jalandhar, India

Abstract

WiMAX is a wireless broadband access technology which provides high data rates, long coverage and IP based connectivity to the stationary as well as mobile users. Voice over Internet Protocol (VoIP) through WiMAX is found to be very promising. In this paper, simulative investigations have been done to identify the suitable voice codec for the same. In doing so, silence suppressed voice signal are also used. Investigations have been done in terms of Quality of Service parameters like jitter, packet end to end delay and Mean Opinion Score. Extensive simulations have been done to shortlist the suitable audio codec for VoIP over WiMAX.

Keywords: *WiMAX, VoIP, OPNET, QoS, voice codec.*

1. Introduction

Over the past ten years mobile communications have transitioned from a luxury to a utility. WiMAX i.e. Worldwide Interoperability for Microwave Access is one of the hottest broadband wireless technologies which is based on IEEE 802.16 [1] specification and delivers high quality broadband services to residential and enterprise customers in an economical way [2].

The network can provide broadband wireless access (BWA) upto 30 miles for fixed station and 3 to 10 miles for mobile stations with transmission data rates between 1.5 and 75 Mbps per channel. It can be used not only as xDSL replacement for small business customers but also as a mobile internet access technology. The WiMAX standard air interface includes the definition of both the Medium Access Control (MAC) and the Physical (PHY) layers for the subscriber station and base station while the access network operability is defined by the WiMAX Forum, an organization consisting of operators and components and equipment manufacturers [3].

WiMAX provides wide range of applications such as voice over Internet Protocol (VoIP), Internet Protocol Television (IPTV), mobile data TV, mobile emergency response services, and wireless backhaul as substitute for fiber optic cable. VoIP has been widely accepted for its cost effectiveness and easy implementation. The bandwidth and delay sensitive properties of VoIP transmission can be utilized to study the influence of voice codecs over the transmitted voice quality. Another aim of the present work is to investigate the performance of VoIP using silence suppressed voice signals.

The rest of the paper is organized as follows: Section II describes about related work and Section III provides about the simulation methodology. Results are discussed in Section IV before we finally concluded in section V.

2. Related Work

There have been recent studies focusing on the performance evaluation of VoIP in WiMAX networks. Adhicandra et al [4] have investigated the data and voice support in the WiMAX network. Various QoS configurations were used to improve the performance of VoIP over Best Effort (BE) WiMAX. The extended real-time polling service (ertPS) scheduling class that was designed to support variable rate real-time services significantly improves the performance of VoIP over BE WiMAX.

Ali et al [5] provides a study of the bandwidth and delay requirements of VoIP services in WiMAX networks. They provide analysis of the effects of Robust Header Compression, Payload Header Suppression, Voice Activity Detection with Discontinuous Transmission and other bandwidth affecting factors. Farzin et al [6] focused in the coverage area for WiMAX and Wi-Fi networks, where constant values for WiMAX and Wi-Fi parameters are used. It is concluded that the best way to get wide coverage area is to use WiMAX network as backhole for Wi-Fi network.

Pentikousis et al [7] considered a fixed WiMAX network in order to evaluate the performance of VoIP. The performance of different transmission schemes in term of cumulative goodput, packet rate, sample loss rate and Mean Opinion Score (MOS) using R-score specified by ITU-T has been measured. Sun et al [8] have discussed different issues related to VoIP and voice quality measurement models.

They have outlined a new methodology for developing models for non intrusive prediction of voice quality.
Another simulation study compares VoIP over Wi-Fi and VoIP over WiMAX [9]. The simulation results show that the throughputs of Wi-Fi and WiMAX networks are affected by VoIP application. However, jitter and packet loss are experienced only in Wi-Fi networks.

3. Simulation Methodology

To evaluate the performance of VoIP over the WiMAX network, a scenario was designed in the network simulator OPNET with the assumption that the traffic generated in this network model is VoIP only. There are only peer-to-peer voice calls throughout the simulation, which means there is no voice conferencing and the subscriber stations (SS) are considered as fixed during the simulation runs. Fig. 1 illustrates the WiMAX network model considered in the simulations. The WiMAX network consists of four SSs and one BS which are connected to the server backbone having only one voice server through an IP backbone.

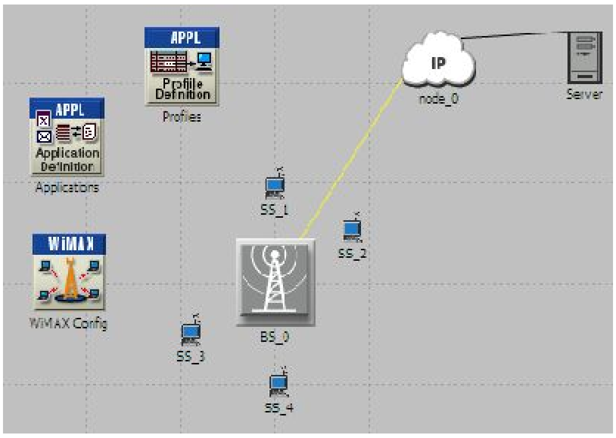

Fig.1 WiMAX Network Model

The WiMAX parameters used in the network are listed in the Table I & Table II. In this scenario, various voice codecs with different bit rates are used in the same WiMAX network to investigate the performance of VoIP.

3.1 Voice Codecs

VoIP uses Internet Protocol for transmission of voice as packets over IP networks. The process involves digitization of voice, the isolation of unwanted noise signals and then the compression of the voice signal using compression algorithms/codecs.
Voice compression using a codec is expected to provide good voice quality even after compression, with minimum delay. The codecs under investigation includes,

Table I: Network Design Parameters

WiMAX Physical Profile	Wireless OFDMA 20MHz
Bandwidth	20MHz
FFT	512
Duplexing Technique	TDD
Efficiency Mode	Physical Layer Enabled
WiMAX Service Class	ertPS
BS Transmission Power	10W
SS Transmission Power	0.5W
Modulation & coding	16 QAM 3/4
Multipath Channel Model	ITU Pedestrian B
Pathloss Model	Suburban Fixed (Erceg)
Terrain Type	Terrain B

Table II: Traffic Characteristics

Match Property	IP ToS
Match Condition	Equals
Match Value	Interactive Voice

G.711 is the international standard for encoding telephone audio. It is a public domain codec widely used in VoIP applications. It employs a logarithmic compression that compresses each 16-bit sample to 8-bits. As a result, its bit-rate is 64 kbps.
G.723.1 is one of the most efficient and licensed codecs with the highest compression ratio and is used in video conferencing applications. It has two versions with distinct bit-rates: 5.3 and 6.4 kbps.
G.729A is an industry standard offers toll-quality voice calls at reasonably low bit rate of 8Kbps. It is also a licensed codec and is based on the Conjugate-Structure Algebraic-Code-Excited Linear Prediction (CS-ACELP) algorithm.
G.726 is an ITU standard codec, which uses ADPCM (Adaptive Differential Pulse Code Modulation) algorithm, and transmits at rates of 16, 24, 32, and 40 kbps. It is primarily used for international trunks to save bandwidth. It is also the standard codec used in DECT (digital Enhanced Cordless Telephony) wireless phones.
G.728 is an ITU standard codec, which operates at bit rate of 16 Kbps using LDCELP (Low Delay Codebook Excited Linear Prediction) compression technique.

3.2 Performance Metrics

IP telephony/ VoIP are something that is growing up very fast. QoS parameters which are appropriate for the present investigation includes,
Mean Opinion Score (MOS): The MOS is the most widely used subjective measure of voice quality. MOS follows the measurement techniques specified in ITU-T P.800 [10], where different people are made to listen the voice signals and are made to rate the factors like distortion, delay, echo; noise etc on a scale of 1 to 5 where 1 is the minimum and 5 being the maximum. The mean opinion score is the arithmetic mean of all the individual scores.

Table III: Mean Opinion Score

Quality Scale	Score	Listening Effort Scale
Excellent	5	No effort required
Good	4	No appreciable effort required
Fair	3	Moderate effort required
Poor	2	Considerable effort required
Bad	1	No meaning understood with effort

Jitter is technically the measure of the variability over time of the latency across a network i.e. the variation in arrival time of consecutive packets. If two consecutive packets leave the source node with time stamps t_1 & t_2 and are played back at the destination node at time t_3 & t_4, then:

Jitter = $(t_4 - t_3) - (t_2 - t_1)$

Negative jitter indicates that the time difference between the packets at the destination node was less than that at the source node.

Packet End to End Delay: Packet End to End delay or latency is characterized as the amount of time it takes for speech to exit the speaker's mouth and reach the listener's ear. The total voice packet delay is calculated as:

$D_{e2e} = D_n + D_e + D_d + D_c + D_{de}$

Where D_{e2e} represents the end-to-end delays while D_n, D_e, D_d, D_c, D_{de} represent the network, encoding, decoding, compression and decompression delays, respectively.

The benchmark guidelines provided by the Telecommunication Standardization Sector of the International Telecommunications Union (ITU-T) in terms of packet end-to-end delay and jitter are tabulated below in order to judge the quality of voice in the present work. A good quality voice call should have a delay between 0 ms and 150 ms and a jitter between 0 ms and 20 ms. However, if a call experiences a delay greater than 300 ms or a jitter greater than 50 ms, it is considered to be of a poor quality. Otherwise, calls are considered to be of acceptable quality.

Table IV: Guidelines for Voice Quality

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

4. Results and Discussion

Voice Quality is important for VoIP system because of user's high demand for good quality voice services. Performance of VoIP in WiMAX is compared in terms of voice codecs G.711, G.723, G.726, G.728 & G.729 and produces following results.

Packet end-to- end delay: VoIP in WiMAX network would be transmitted to destination as a collection of packets where each one might follow different routes thus arrive at the destination with different delays. Figure 2 shows the average packet end-to-end delay for the different codecs.

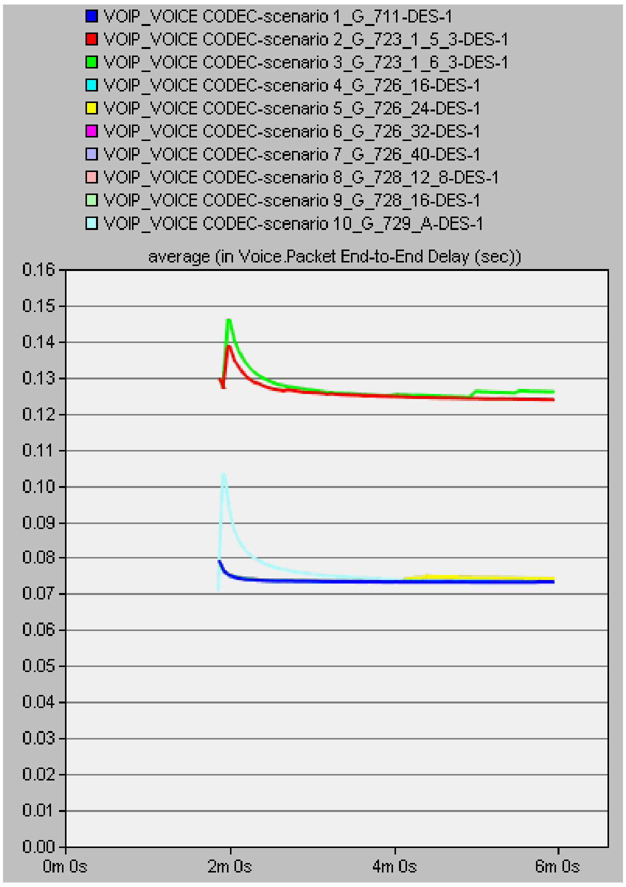

Fig.2 Relative Packet End to End Delay of voice codecs (sec)

Packet end to end delay for voice codec G.723.1 is highest. This is because, G.723.1 uses coding rate of 5.3Kbps or 6.3Kbps which results in the formation of packets of smaller size and larger count. As the number of packets increases in the network, the congestion in the network increases. Congestion directly affects the network packet delay and thus results in increased packet end to end delay.

The Packet end to end delay is lowest for codec G.711. This implies G.711 can provide better VoIP services in terms of end to end packet delay.

Mean Opinion Score (MOS): Mean Opinion Score is one of the most important performances metric in VoIP. Figure 3 shows the comparative results of average MOS values for the codecs used in the scenario.

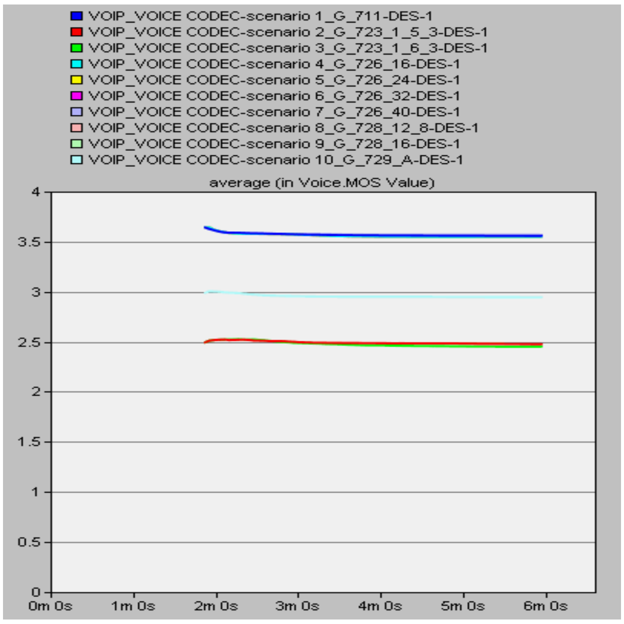

Fig.3 Relative Mean Opinion Score of different voice codecs

Mean Opinion Score value is highest for the voice codec G.711 and for the codec G.723.1, its value is lowest. G.723.1 generates packets of lower size and higher in number as compare to G.711. This results in network congestion and thereby packet drop. Hence the MOS value for G.723.1 is quite low.

Jitter: Fig. 4 & 5 shows the comparative results of voice jitter for the codecs that are used in this experiment.

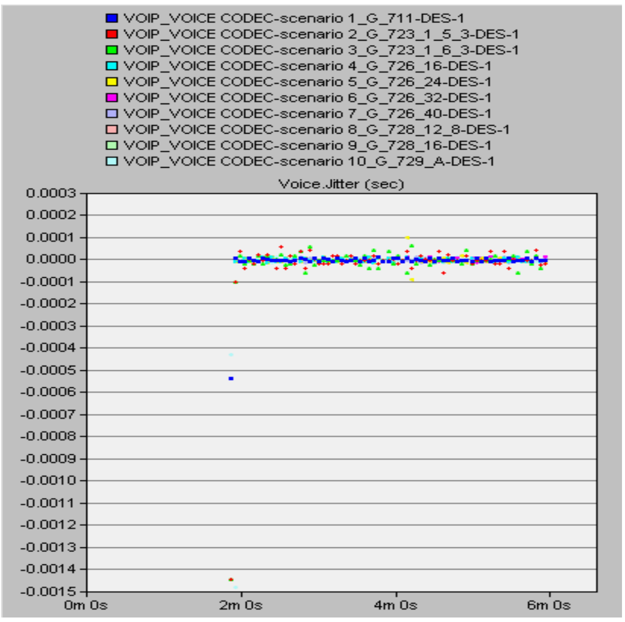

Fig.4 Relative Jitter of voice codecs (sec)

G.729A codec scheme has larger value of jitter and therefore yields highest variations in the curve among all other codec schemes. The negative value of jitter means that the time difference between the packets at the destination is less than that at the source. The voice jitter value for G.711 is lowest which shows that there is a minimum delay variation between the VoIP packets.

Considering throughput as one of the important parameter of WiMAX network, throughput represents the total data traffic in bits/sec forwarded from WiMAX layer to higher layers in all WiMAX nodes of the network. The following figure shows the average value of throughput in bits per sec for VoIP application in WiMAX network. The results show that the voice codec G.711 has the highest value and least value for G.723.1 codec.

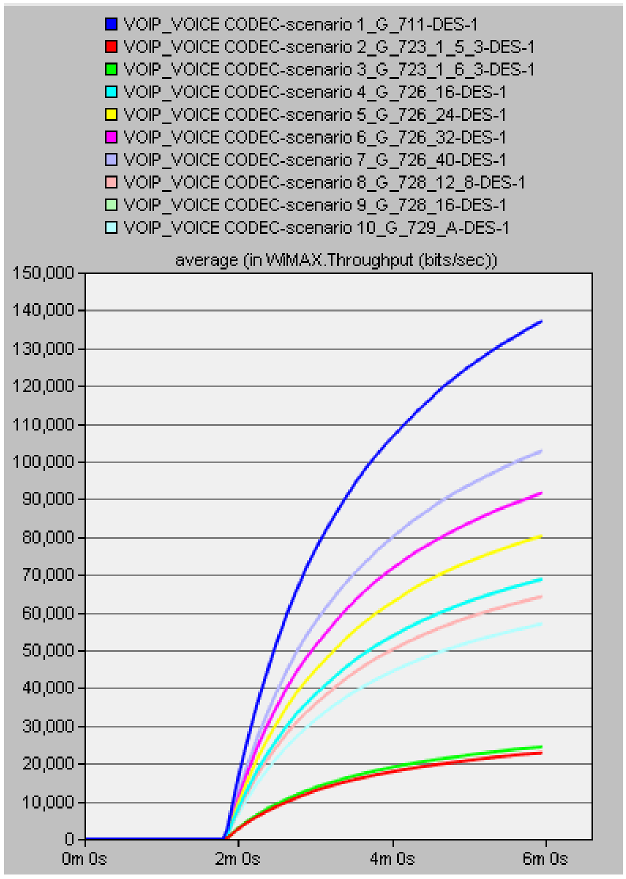

Fig.5 Relative Throughput of voice codecs (bits per sec)

When the codecs are compared for with and without silence suppressed voice signal, the following results are shown. As Mean Opinion Score of the silence suppressed signals is high but it shows extreme rise in jitter values which is not acceptable in VoIP application.

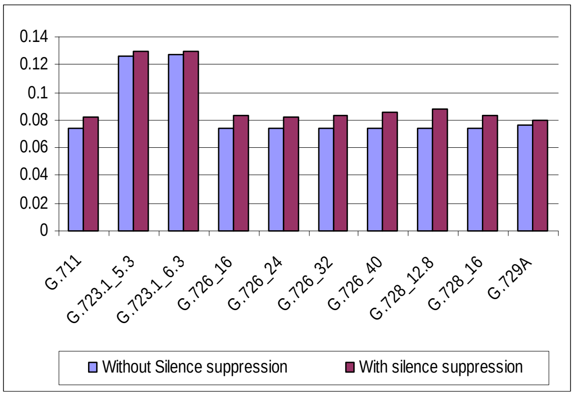

Fig.6 Packet End to End Delay with silence suppression (sec)

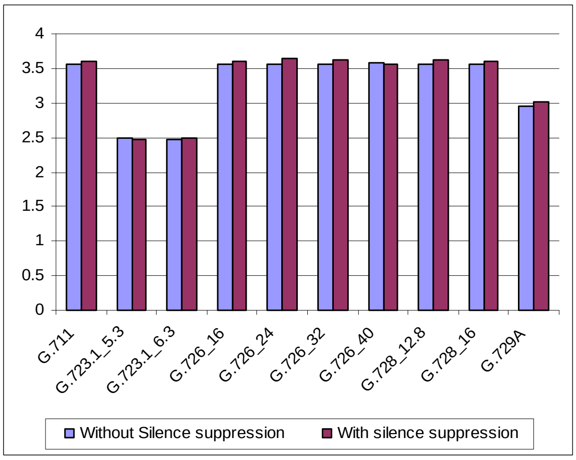

Fig.7 Mean Opinion Score with silence suppression

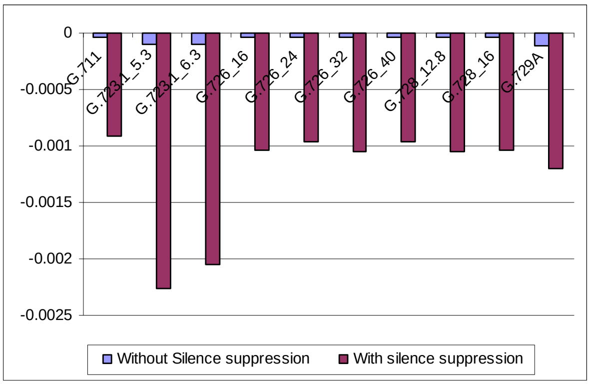

Fig.8 Jitter with silence suppression (sec)

5. Conclusions

With wireless mobile access becoming an integral part of modern life, the improvement of the performance of WiMAX network has become a topic of keen interest of the researchers. WiMAX is one of the technologies with high data rates and long coverage area. Voice over IP is considered as the application as it is expected to be a low cost and thereby, a popular communication system in the next generation communication networks. In this paper, the performance of VoIP over WiMAX networks is investigated in terms of important QoS parameters of VoIP such as jitter, MOS, packet end to end delay and throughput. The simulation results showed the strong dominance of G.711 codec in all the other categories and hence more desirable in VoIP applications.

References

[1] IEEE Std 802.16 TM- 2004, “Part 16: Air Interface for fixed broadband wireless access systems” Oct 2004.
[2] http://www.tutorialspoint.com/wimax/what_is_wimax.htm
[3] WiMAX Forum. http://www.wimaxforum.org.
[4] I. Adhicandra, “Measuring data and VoIP traffic in WiMAX networks,” Journal of Telecommunications, vol. 2, no. 1, pp. 1–6, Apr. 2010.
[5] A. A.Ali, S. Vassilaras, and K. Ntagkounakis, "A Comparative Study of Bandwidth Requirements of VoIP Codecs Over WiMAX Access Networks," IEEE computersociety, vol. 47, pp. 197-203, 2009.
[6] A. Farzin, “OPNET Simulation of IEEE 802.11(Wi-Fi) and IEEE 802.16(WiMAX) in a smallarea," COMMUNICATION NETWORKS, Project ENSC 895-g200, spring 2010.
[7] K. Pentikousis, E. Piri, J. Pinola, F. Fitzek, T. Nissilä and I.Harjula, "Empirical evaluation of VoIP aggregation over a fixed WiMAX testbed," in Proceedings of the 4th International Conference on Testbeds and Research Infrastructures for the Development of Networks & Communities, 2008, pp. 19.
[8] L. Sun and E. C. Ifeachor, "Voice quality prediction models and their application in VoIP networks," Multimedia, IEEE Transactions on, vol. 8, pp. 809-820, 2006.
[9] M. A. Qureshi, A. Younus, M. Saeed, F. A. Sidiqui, N. Touheed, and M. S. Qureshi, “Comparative study of VoIP over WiMAX and Wi-Fi,” International Journal of Computer Science, vol. 8, no. 1, pp. 433–437, May 2011.
[10]http://www.itu.int/rec/T-REC-P.800-199608-I/en.