ISSN 2047-3338

Effect of Mobility Patterns on VoIP QoS in Mobile WiMAX

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***Abstract*—VoIP over Mobile WiMAX provides high data rate voice connectivity to high speed mobile as well as to fixed users. Voice over Internet Protocol (VoIP) enables the transport of voice data over internet protocol base networks. The WiMAX mobile network provides data services to both mobile as well as pedestrian users and the quality of voice is greatly affected with the mobility patterns of mobile users. The mobility behavior of the user plays an important role in determining the quality of Voice. For providing better quality of service to end user, the Mac layer of WiMAX defines various QoS classes for different types of applications. In this research paper, we analyze the performance of VoIP codecs over mobile WiMAX when resources are allocated to VoIP users using best effort Scheduling scheme. We analyze the performance of voice codecs under two conditions, first, the jitter buffer is enabled and disables, second the mobile user are moving at different speeds in random way point and group mobility patterns. Our comprehensive simulation results show that the performance of voice codecs remain same for random way point and group mobility patterns. Our results reveal that the G729 provides better voice quality as well as support more number of connections as compared to other voice codecs.**

***Index Terms*–VoIP Codecs, Mobile WiMAX, Mobility Models and BE Scheduling Class**

I. INTRODUCTION

THE IEEE 802.16 standard [1] [2] has been designed as an access network and emerging technology to provide triple plays multimedia services to the end user [3] with promising QoS. The WMAN provides cost effective infrastructure to service providers with addition of complexity and provide committed QoS to end-users.

VoIP over mobile WiMAX has been emerging as cost effective services with guaranteed QoS. The voice services are sensitive to delay and require improved mechanism to provide better QoS. The quality of VoIP users might be degraded due to mobility impact which make the user connectivity irregular as compared to its counterpart wired DSL network. Fading effect has significant influence on the mobile network performance; therefore it decreases the throughput and increases end-to-end delay [4]. VoIP over best effort mobile WiMAX requires attention to sort out different issues including the network architecture, system design, network capacity, configurations and QoS for differ-ent mobility models.

The purpose of this study is to evaluate the performance of VoIP codecs under different mobility patterns. We first create mobile WiMAX network, deployed the required applications for streaming video, FTP and VoIP telephony. We evaluate the performance of number of VoIP codecs under different network setup with different mobility patterns. The evaluated results reveal that MoS for different codecs is much below the threshold as recommend by ISO. The simulation is per-formed using QualNet simulator with different parameter setting for various codecs.

The rest of the paper is organized as follows. In section 2, we provide back ground study of mobile WiMAX and mobility patterns. In section 3 the VoIP codecs and QoS metrics are presented. In section 4 the simulation model is outlined. In section 5 we analyze the performance of VoIP codecs under different network scenarios. Section 6 provides related work and section 7 concludes the article and provides future directions.

II. MOBILE WiMAX

The WiMAX technology, based on the IEEE 802.16-2004 Air Interface Standard is rapidly proving itself as a technology that will play a key role in fixed broadband wireless metropolitan area networks. Fixed WiMAX, based on the IEEE 802.16-2004 [1] Air Interface Standard has proven to be a cost effective fixed wireless alternative to cable and DSL services.

Table I. Fixed WIMAX vs. Mobile WIMAX

Parameter	Fixed WiMAX	Mobile WiMAX
Frequency GHz	3.5,5.8	2.3,2.5,3.5,etc
Channel MHz	3.5,7,10,4	3.5,7,8.75,10,14 etc
Duplexing	TDD/FDD	TDD/FDD
Multiple Access	TDMA	OFDMA
Mobility	No	Yes

A. Mobile WiMAX

Mobile WiMAX is a broadband wireless solution that enables convergence of mobile and fixed broadband net-works through a common wide area broadband radio access technology and flexible network architecture. The Mo-bile WiMAX Air Interface adopts Orthogonal Frequency Division Multiple Access (OFDMA) for improved multi-path performance in non-line-of-sight environments. Scalable OFDMA (SOFDMA) [5] is introduced in the IEEE 802.16e Amendment to support scalable channel bandwidths from 1.25 to 20 MHz.Release-1 Mobile WiMAX profiles will cover 5, 7, 8.75, and 10 MHz channel bandwidths for licensed worldwide spectrum allocations in the 2.3 GHz, 2.5 GHz, 3.3 GHz and 3.5 GHz frequency bands. Mobile WiMAX systems offer scalability in both radio access technology and network architecture, thus providing a great deal of flexibility in network deployment options and service offerings. Some of the salient features supported by mobile WiMAX are: high data rate, Quality of Service, scalability, security and mobility. The comparison of fixed and mobile WiMAX is presented in Table I.

B. Mobility Models

In a mobile wireless network, mobile station (MS) can move in many different ways. The mobility models are designed to describe the movement pattern of mobile users. The mobility patterns play a significant role in determining the protocol performance. For example, the nodes in Ran-dom Way Point model behave quite differently as compared to nodes moving in groups. Therefore, there is a real need for developing a deeper understanding about mobility models and their impact on protocol performance. In this experimental evaluation, we choose the random way point [4] and the group mobility model [6] to evaluating their effects on quality of voice for users under different speeds.

i. *Random Way Point*: In random mobility models, the mobile station moves randomly and freely without restrictions. The Random way Point model was first proposed by Johnson and Maltz [4] and become a bench mark for performance evaluation of different applications. In the simulation study, the implementation and configuration of this mobility model is as follows: as the simulation starts, each VoIP mobile station selects destination randomly and moves toward with constant speed distributing over (0, MaxSpeed) where Max speed is the maximum allowable speed. The speed and direction of mobile node is chosen randomly and indecently of other mobile stations in a sys-tem. One the mobile station reaches its selected destination the mobile station paused for duration of some time and then again chooses another random destination and starts moving towards it at the speed of (0, MaxSpeed). The whole process is repeated again and again until the simulation ends.

ii. *Group Mobility Model*: In Reference Point Group Mobility (RPGM) model, the nodes are uniformly or randomly distributed in geographic scope of a group. Each group has a logical center. Thus the group motion behavior is determined by providing a path for the center. A reference point is assigned to each node then node follows the group movement. Each node is randomly placed in the neighborhood of its reference point at each step and at each instant each node has a speed and direction that is derived by randomly deviating from that of the logical center. Group and each node movement is based on the Random Way Mobility (RWP) model. Some applications such as ubiquitous computing, battle field etc. are based on group motion.

III. VOIP CODECS

Voice codecs are used to convert an analog voice signal into digitally encoded version. The codecs may vary in sound quality, required bandwidth and computational requirements etc. Table II shows the commonly used voice codecs with their algorithms, bit-rates and mean opinion score (MOS).

A. Voice Quality

In VoIP, quality simply means being able to listen and speak in a clear and continuous voice without unwanted noise. Speech quality should be approached from an end-to-end perspective; that is, regardless of the systems, devices, and transmission methods used, any voice-quality metric should be expressed in the context of the user’s experience. The techniques used to measure the voice quality of a VoIP call are the Mean Opinion Score (MOS) [7] and Perceptual Speech Quality Measurement (PSQM) [8]. In the realization of VoIP applications, the availability of Quality of Services (QoS) is of great importance for the end users. The recommended One-Way delay and jitter for voice applications is 150 ms and 50 ms respectively. The specification of different codecs is shown in Table II.

Table II. Voice Codec Specification

Voice Codec	Bit Rates (Kbps)	Algorithm	MoS
G.711	64	PCM	4.5
G.729	8	CS-ACELP	4.2
G.723.V6	6.3	Multi-rate Code	3.98
G.723.V5	5.3	Multi-rate Code	3.6
G.726	32	ADPCM	4.2

B. VoIP codec Performance Metrics

The performance metrics used in our experiments include: I) Mean Opinion Score (MoS), II) Packet loss, III) End to End Delay, IV) Jitter.

i. *Mean Opinion Score:* The most common measurement metric of voice quality is the MOS [7]. The relationship between audio performance characteristics and a quality score makes to the MOS a valuable standard for network assessments, benchmarking, tuning, and monitoring. The MOS value can be ranged from 1 bad to 5 excellent. The MOS for different codecs are represented in II. The conventional codec for fixed telephone lines G.711 has a MOS of 4.0 at 64kbps. The VoIP systems usually use G729, G726 and G723 due to their low bandwidth requirements. The G729 has a MOS value of 4.0 and operates at the rate of 8kbps. The G.723 which is mainly used for Video Telephony has a MOS 3.8 and operates at 5.3 /6.8 kbps.

ii. *Packet Loss:* Packet loss does excessive damage to the voice signal, because retransmission while transmitting voice signal cannot be considered as an option. Loss of voiced frames causes significant degradation of the signal.

iii. *Delay*: VoIP delay is the amount of time which is taken for speech to exit the speaker's mouth and reach the listener's ear. There are three types of delay; those are used in today's telephony networks: propagation delay, serialization delay, and handling delay. Propagation delay is caused by the length of signal must travel through a communication channel. Handling delay; also called processing delay de-fines many different causes of delay (actual packetization, compression, and packet switching) and is caused by devices that forward the frame through the network. Serialization delay is the amount of time it takes to actually place a bit or byte onto an interface.

iv. *Jitter*: The arriving time of packets varies as a result of different queuing times or different routes and is referred to as Jitter. Jitter can be taken care of by using an adaptive jitter buffer which adapts itself according to the delay encountered over the networks, to provide a smooth voice stream at the output. In this research paper we are analyzing the jitter for different voice codecs under mobility model.

IV. SYSTEM MODEL

Our simulation setup consists of WiMAX base station, VoIP users, dedicated video streaming server and users, FTP server and FTP users. The base station provides communication channels for mobile user for having any sort of communication service. The simulation setup is shown in figure 1. The values for major WiMAX network parameters are listed in table III. In our experiments there are four groups of users those are randomly placed around the BS in a circular fashion.

The first group consists of 10 stationary users having higher priority CBR (Constant Bit Rate) traffic for whole simulation period. Each user in this group has transmission rate of 264 kbps. The second group also consists of 10 active users, accessing the video from video on demand server. The users in shit group are randomly distributed in both classes and have packet inter arrival time 20 msec. The third group consists of 10 active mobile users downloading FTP files from FTP server. The fourth final group consists of 20 active mobile VoIP users, having VoIP flows with different data rates depending on the codec selection. The VoIP users are running on Best Effort scheduling scheme. The simulation is performed for different voice codecs including: G711, G728, G729 and G723 having data rates 64kbps, 16kbps, 8kbps and 5.3kbps respectively. In our experiments, we are not using comfort noise generation and voice activity detection mechanism to generating VoIP traffic among the VoIP users.

Table III. Mobile WiMAX Simulation Parameters

Description	Value
Terrain size	1500 x 1500 m
Number of cells	1
Number of base station	1
Number of SSs	50
Operating frequency	2.4 GHz
System bandwidth	10 MHz
Frame size (msec)	5
Frame size ratio of DL to UL	2:1
Phy Scheme	OFDMA
Duplex scheme	TDD
Modulation Technique	Adaptive
Mobility Models	RWP & Group Mobility
Fading Model	Rayleigh fading Model
SS transmit power	23 dBm
Simulation time	1000s

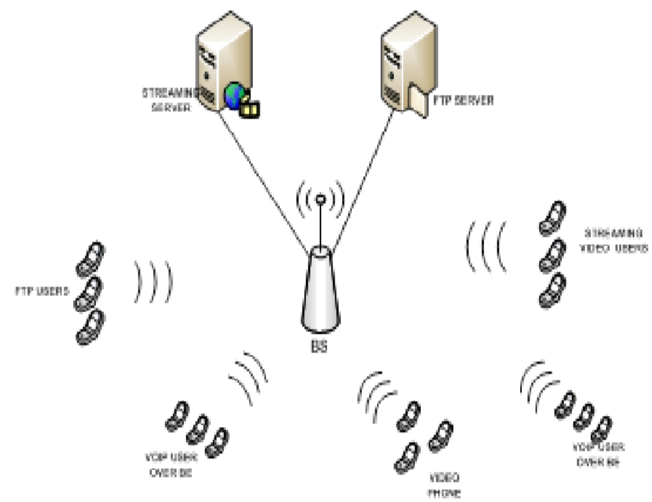

Figure 1. Simulation System Model

V. RESULT ANALYSIS

We have performed experiments using Random Way Point and Reference Point Group Mobility model under two simulation scenarios. In the first scenario, the jitter buffer is adjusted at the receiver side and in second scenario no jitter buffer is adjusted at the receiver side. We have studied the performance of QoS parameters for different set of codecs under these two scenarios and mobility models.

The results in Figure 2 and Figure 3 represent the MoS, packet loss, average end-to-end delay and average jitter for RPGM model with and without any jitter buffer adjustment at the receiver side. The results of the simulation show that the packet loss is about 1% for all codecs with no jitter buffer implementation which is less than the packet loss when jitter buffer is adjusted at the receiver side. However comparing the performance of different codecs under jitter buffer the results reveal that G711 experience higher packet loss as compared to other codecs that may be because of higher bit rate of G711 or low bandwidth in the jitter buffer. When the number of flows

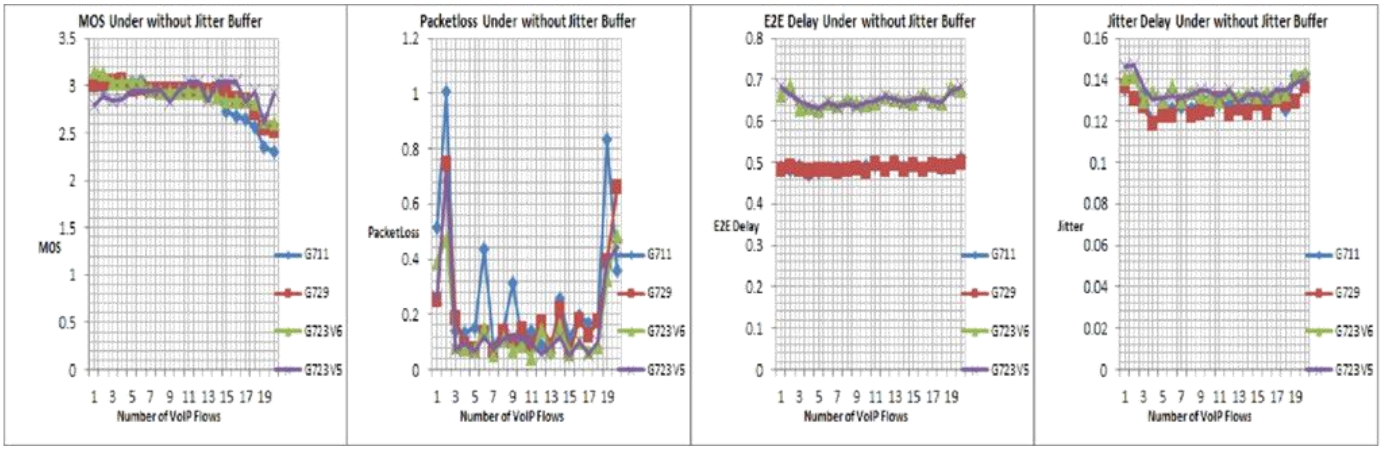

Figure 2. Group Mobility Model without Jitter Buffer. A) MOS B) Packet Loss C) E2E Delay (Sec) D) Jitter (Sec)

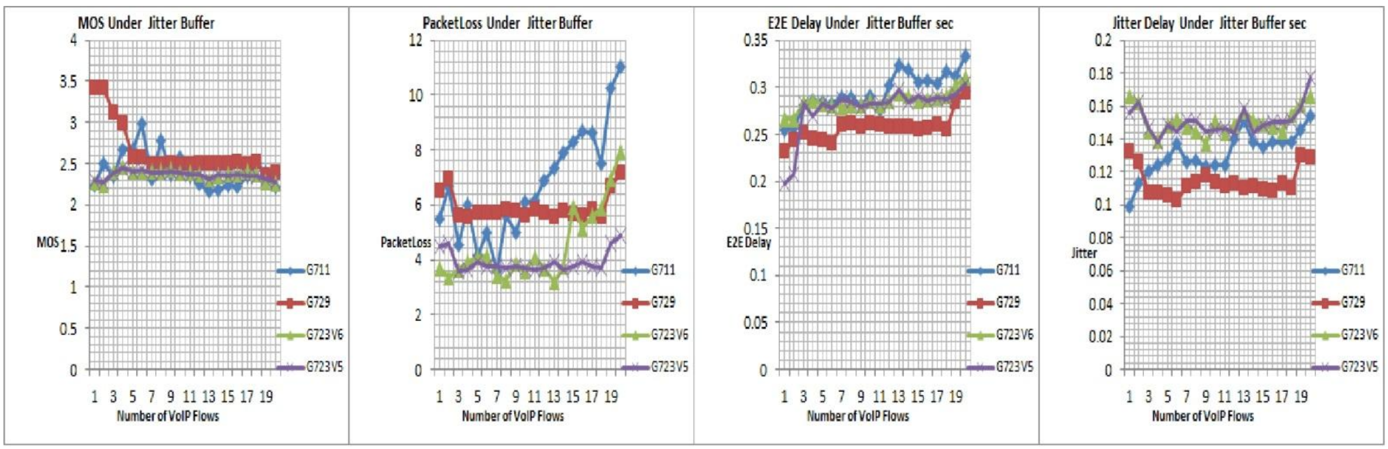

Figure 3. Group Mobility Model with Jitter Buffer. A) MOS B) Packet Loss C) E2E Delay (Sec) D) Jitter (Sec)

increased, both versions of G723 experience less packet loss as compared to packet loss of G729, but G729 shows consistence behavior as the number of flows increased.

The MoS value decreased as the number of VoIP sessions are increased and become stable at around 2.3-2.5. The G729 out performs when its MoS value is compared with other codecs. In wired network, the G711 provides premium voice quality but in mobile WiMAX the MoS value of G711 decreases as the number of VoIP flow increased. This behavior is due to higher bandwidth requirement of G711 especially when it is run over the BE WiMAX network, as BE only allocates the available resources to the users without getting the resources from the other real-time traffic. Analyzing the MoS value under jitter and no jitter buffer, the MOS value is greater than without jitter buffer which shows that applying fixed length jitter buffer may effects the quality with the increased in number of VoIP sessions.

The jitter represents the variation and transit delay of VoIP packets which is caused by queuing, contention and serialization effects on the path through the network. In general, higher level of jitter is more likely to occur on either slow or heavily congested links or unavailability of desired bandwidth for voice conversation. The higher the jitter the lower the quality of received voice. The end-to-end delay is delivering the speech contents from the speaker's mouth to the listener's ear and also depends on the availability of network resources and codecs processing delay from analog to digital signals. In context of implementing VoIP setup under jitter buffer and without jitter buffer, the E2E delay is more when applying jitter buffer in a network as compared to without jitter buffer setup. However, the jitter is minimized when we implied jitter buffer in a network. Comparing the delay and jitter for voice codecs the delay and jitter of G729 and G711 is less than G723 in both versions. The results of simulation also show that the placement of jitter buffer may decrease the jitter in conversation but it increases the end to end delay.

The results for RWP mobility model with and without jitter buffer implementation are shown in Figure 5 and Figure 4. Under jitter buffer implementation, the MoS value is between 2.3 and 3.5 for all codecs and with no jitter buffer it falls between 1.5 and 3. Similarly in RPGM mobility model the MoS value for G729 is much better than other voice codecs.

The user experiences less packet loss under jitter buffer as compared to no jitter buffer. As the numbers of VoIP flows are increased the packet loss is also increased. In RPGM mobility, the G711 experiencing higher packet loss compared to other VoIP codecs. We believe it is a result of its higher data rate requirement because in these experimental setup

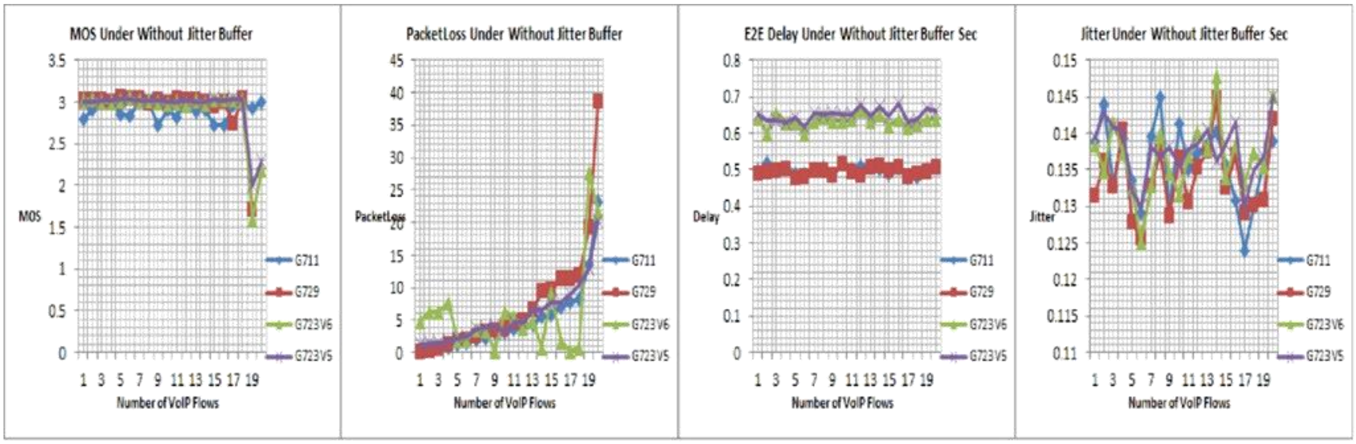

Figure 4. Random Way Point Mobility Model without Jitter Buffer. A) MOS B) Packet Loss C) E2E Delay (Sec) D) Jitter (Sec)

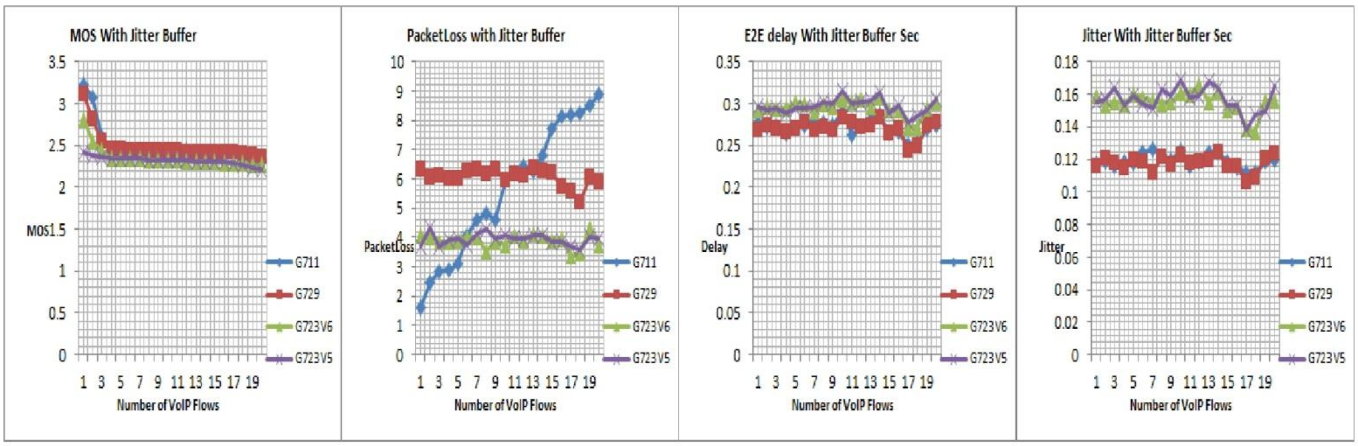

Figure.5. Random Way Point Mobility Model with Jitter Buffer. A) MOS B) Packet Loss C) E2E Delay (Sec) D) Jitter (Sec)

bandwidth resources to VoIP packets are allocated through BE QoS scheduling. In terms of packet loss, both versions of G723 suffer less packet loss compared to G729 and G711.
The jitter buffer holds packets for short duration before handling over to upper layer, which results in more E2E delay in case of jitter buffer as compared to without jitter buffer implementation. The delay under jitter buffer does not vary much as compared to variation delay in without jitter buffer implementation. Comparing codecs performance, the G729 suffers less E2E delay for both scenarios as compared to G711 and G723. The G729 and G711 suffer low jitter as compared to G723 because of codec complexity and also do not varies much with the number of VoIP flows. Similarly for RWP mobility model, in RPGM mobility the jitter is less in case of jitter buffer however the jitter is still at acceptable level for human conversation.

From the simulation study for both group and random mobility model we conclude that the mobility models do not have much impact on the performance of received quality for different voice codecs. However, the implementation of jitter buffer at receiver side may decreases the end-to-end delay and jitter thus results in higher packet loss. The packet loss can be minimized by proper dynamic adjustment of jitter buffer at the receiver side. Comparing the codecs performance we conclude that the bandwidth requirement of G723 is less than compared to G711 and G720. But it suffers from higher jitter and higher E2E delay, thus having low MoS value. Considering all QoS parameters and bit rate requirement of codecs we are of the view that G729 is best suited to transmitting voice using BE scheduling scheme. The adaptation of G729 also increases the number of VoIP users that can be accommodated as compared to G711. We also conclude that the proper dynamic adjustment of jitter buffer may also improve the MoS value and received quality of voice in mobile WiMAX.

VI. RELATED WORK

Supporting real-time applications over any wireless network (for example, 3G cellular networks, IEEE 802.11-based wireless LANs, and IEEE 802.16-based WiMAX) poses many challenges, including limited bandwidth, coping with bandwidth fluctuations, and lost or corrupted data. The performance evaluation of streaming and VoIP services over such network is widely researched and many proposals have been defined to improve the quality of multimedia service over high speed wireless networks.

In [9], the authors measured the capacity of WiMAX link using BE and the performance of mixed traffic. The authors did not evaluate the VoIP performance regarding RTP jitter and delay. The authors of [10], identified the mobility influence on throughput, the packet loss and delay with the main focus on signal strength. The authors have not reported the results with respect to the VoIP flows and QoS parameters. In [11], the authors focused on different queue management mechanism for RTP traffic and their effect on jitter, packet loss and delay. Results shown DiffServ perform better under burst traffic condition up to a congestion level. The RWP mobility model is the best model due to its performance as compare to random walk and direction models [12]. In [13], the authors study the Mobile WiMAX capacity to support the multimedia applications with different user speed. The results showed that handover has high influence on the performance of multimedia applications; however the use of cross-layer information through MIH performance can be improved. The authors in [14] studied the effect of various delay factors for VoIP traffic over WiMAX network and concluded that the number of base stations and range of base station play role in increased delay for voice traffic.

In [15] the authors evaluated voice over IP (VoIP) performance over fixed WiMAX using synthetic traffic generation. They evaluated the performance of G723.1 VoIP traffic and measured the capacity of a WiMAX test bed in terms of VoIP calls for both up-link and down-link.

The authors in [16] present simulation results for mixed best effort and multimedia traffic. Their results indicate that the average delay for the best effort traffic grows more sharply on the up-link than on the down-link, because of the bandwidth-request mechanism and signaling overhead. For multimedia traffic in the rtPS traffic class, delay and delay variation are stable until the SS population saturates In this simulation study, we evaluate the performance of different voice codecs over BE effort WiMAX network. We studied the performance of VoIP codecs in a mobile WiMAX under the conditions that the VoIP users are running through BE scheduling class with real time back ground stream and FTP traffic running over other scheduling classes. We also studied the effect of mobility models on received quality of VoIP users with the conditions that RTP jitter buffer is on and off at the receiver side. To the best of our knowledge, this study is the first one which considers the mobile VoIP users running over the BE scheduling class with ertPS, nrtPS back ground traffics. We also provide the recommendation about which voice codec performs well when VoIP users have BE scheduling class.

VII. CONCLUSIONS & FUTURE WORK

In this paper, we use experimental measurements to study the performance of multimedia applications over a mobile WiMAX network. Different voice codecs used to test Voice-over-IP (VoIP) applications in conjunction with mobility models and adjustment of jitter buffer for different number of VoIP flows. The mobile WiMAX-based network solidly supports different QoS scheduling techniques for different type of traffics based on priority. In simulation, we have allocated BE scheduling to the VoIP users and other multimedia users are running over defined QoS scheduling techniques. We conclude, in mobile WiMAX under BE effort scheduling scheme the G729 VoIP codec provides better voice quality to end users. It also supports more number of users in a network as compared to G711 which provides premium voice quality in wired network. The placement of jitter buffer improves the QoS parameters but it requires adaptive adjustment according to underlying network conditions. Moreover, we conclude that the mobility models do not have much impact on the performance of received voice quality. The future work includes: Analyzing the voice over BE mobile WiMAX with background TCP traffic running over BE scheduling scheme and mechanism for adaptive adjustment of jitter buffer. Further, we are also intending to develop a scheduler that can maintain listen able MOS of VoIP users in BE network.

REFERENCES

[1] IEEE 802.16 Working Group, IEEE Standard for Local and Metropolitan Area Networks, Part 16: Air Interface for Fixed Broadband Wireless Access Systems, IEEE Std. 802.16-2004, October 2004 Std.

[2] IEEE 802.16 Working Group, Amendment to IEEE Standard for Local and Metropolitan Area Networks, Part 16: Air Interface for Fixed Broadband Wireless Access Systems Physical and Medium Access Control Layer for Combined Fixed and Mobile Operation in Licensed Bands, IEEE Std. 802.16e-2005, December 2005. Std.

[3] N. Scalabrino, F. De Pellegrini, I. Chlamtac, A. Ghittino, and S. Pera, "Performance evaluation of a wimax testbed under voip traffic," in Proceedings of the 1st international workshop on Wireless network testbeds, experimental evaluation & characterization, ser. WiNTECH '06. New York, NY, USA: ACM, 2006, pp. 97–98. [Online]. Available: http://doi.acm.org/10.1145/1160987.1161009

[4] H. Arora and H. Sethu, "A simulation study of the impact of mobility on performance in mobile ad hoc networks," in Proceedings of the Applied Telecommunication Symposium San Diego, California, April 14-18, 2002.

[5] H. Y. l, "scalable ofdma physical layer in ieee 802.16 wirelessman," Intel Technology Journal, vol. Vol 08, 2004.

[6] X. Hong, M. Gerla, G. Pei, and C.-C. Chiang, "A group mobility model for ad hoc wireless networks," in Proceedings of the 2nd ACM international workshop on Modeling, analysis and simulation of wireless and mobile systems, ser. MSWiM '99. New York, NY, USA: ACM, 1999, pp. 53–60. [Online]. Available: http://doi.acm.org/10.1145/313237.313248

[7] P.800.1series p: telephone transmission quality, telephone installations, local line networks, itu std.

[8] ITU-T P.862, Perceptual evaluation of speech quality (PESQ), an objective method for end-to-end speech quality assessment of narrowband telephone networks and speech coders, February 2004. ITU, Geneva Std.

[9] Z. N. Aleksandar Bozinovski and T. Janevski, "Performance evaluation of real-time services in mobile wimax," Telfor Journal,, vol. Vol. 2, No. 1,, 2010.

[10] K. A. Shuaib, "A performance evaluation study of wimax using qualnet."

[11] G. Epiphaniou, C. Maple, P. Sant, and M. Reeve, "Affects of queuing mechanisms on rtp traffic: Comparative analysis of jitter, end-to-end delay and packet loss," in Availability, Reliability, and Security, 2010. ARES '10 International Conference on, feb. 2010, pp. 33–40.

[12] M. I. M. Saad, "Performance analysis of random-based mobility models in manet routing protocol," European Journal of Scientific Research, vol. Vol.32 No.4, pp. pp.444–454, 2009.

[13] Sousa, K. Pentikousis, and M. Curado, "Evaluation of multimedia services in mobile wimax," in Proceedings of the 7th International Conference on Mobile and Ubiquitous Multimedia, ser. MUM '08. New York, NY, USA: ACM, 2008, pp. 64–70. [Online]. Available: http://doi.acm.org/10.1145/1543137.1543149.

[14] Safak and B. Preveze, "Analysis of delay factors for voice over wimax," in Computer and Information Sciences, 2008. ISCIS '08. 23rd International Symposium on, oct. 2008, pp. 1–6.

[15] K. Pentikousis, E. Piri, J. Pinola, F. Fitzek, T. Nissila,¨ 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, ser. TridentCom '08, 2008, pp. 19:1–19:10.

[16] Cicconetti, L. Lenzini, E. Mingozzi, and C. Eklund, "Qual-ity of service support in ieee 802.16 networks," Network, IEEE, vol. 20, no. 2, pp. 50 –55, march-april 2006.

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