Analysis of VOIP Traffic over WiMAX Networks

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***Abstract*— VoIP applications are being broadly employed in present networks challenging their abilities to offer a experience level of good quality to the users. Specially, new wireless broadband techniques i.e. WiMAX, deployed and their performance requires to assess for checking the performance levels of VoIP services. The Quality of Service is the primary challenge of implementation VOIP over WiMAX networks. This paper builds an attempt to study the performance measurements of VoIP for static users, as well as, the variation of the Quality of Service parameters. The experimental results had demonstrated that utilizing sector antenna for both SS and BS and decrease the distance between them, enhance the mean MOS and reduce the mean packet end-to-end delay.**

Keywords: Worldwide interoperable Microwave Access (WiMAX); Physical (PHY) layer; Voice over Internet Protocol (VOIP); Quality of Service (QoS) and Optimized Network Engineering Tools (OPNET).

I. INTRODUCTION

VoIP (Voice over Internet Protocol) is broadly deployed technique and telecommunication operators look for revenue from it. The important benefit of this technique is usage of available infrastructure in the form of internet connection. The use of this kind of communication is very cost efficient. Unluckily, this benefit brings some weaknesses that are expected because of the low quality of wireless connection. The Quality of Service (QoS) is mostly supervised issue by telecommunication vendors and operators. The quality of speech influenced by several factors i.e. packet delay, packet loss, echo, jitter, noise [1], inharmonic and harmonic distortion [2] etc. The QoS parameters are closely associated; with user's satisfaction with a obtained speech quality. The objective mechanisms assess the speech depending on a signal processing without requirement of real listeners. PESQ mechanism (Perceptual Evaluation of Speech Quality) is mostly employed objective mechanism for measurement of the speech quality [3]. The IEEE 802.16 standard for BWA, WiMAX forum predict to provide high data rate over broad regions to a large number of users where broadband is not available. Employ this first industry wide standard can for static wireless access with considerably higher bandwidth as compared to most cellular networks [4]. WiMAX (IEEE802.16) technique assures broadband access for the last mile up to 3 - 10 miles (5 – 15 km) for mobile stations and 30 miles (50 km) for static stations. It offers a wireless backhaul network that makes enable high speed Internet access to small and medium business customers, residential customers as well as Internet access for cellular base stations [5] and Wi-Fi hot spots. The first version of the IEEE 802.16 standard works in the 10–66GHz frequency band and needs line of sight (LoS) towers. After that the standard increased its operation by different PHY specification to 2-11GHz frequency band enabling non line of sight (NLoS) connections, which need mechanisms that effectively remove the impairment of multipath and fading. It provides support to both point-to-multipoint (P2MP) and multipoint-to multipoint (mesh) modes. WiMAX will replace other broadband mechanisms seeking in the same segment and will become a best solution for the deployment of the renowned last mile infrastructures in locations where it is very complex to get with other techniques, i.e. DSL or cable, and where the costs of maintenance and deployment of these techniques would not be beneficial. In this manner, WiMAX will connect rural regions in developing countries as well as underserved metropolitan regions. It is utilized to provide backhaul for enterprise campus, carrier structures and Wi-Fi hotspots. WiMAX provides a best solution for these issues because it offers a cost-efficient, quickly deployable solution [7]. The aim of this study was to analyze a case of QoS deployment over a WiMAX network and to analyze the ability of a WiMAX network to provide sufficient QoS to voice application. The mechanisms considered involve generating the WiMAX network employing OPNET, conducted wide simulations to examine the packet end-to-end delay, MOS, packet delay variation and jitter deploying the VOIP application, setting the WiMAX configurations within various distances between SS and BS, various antenna type and transmission power for both SS and BS.

II. VOIP OVERVIEW

VoIP, also called IP Telephony, is the real-time transmission of voice signals utilizing the Internet Protocol (IP) over a private data network [8] or the public Internet. In simpler way, VoIP changes the voice signal from your telephone into a digital signal that transmits across the Internet. One of the most important benefits of VoIP over a conventional public switched telephone network (PSTN) is that one can build a long distance phone call and bypass the toll charge. This combined data/voice solution permits large organizations

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(with the funding to build the transfer from a legacy network to a VoIP network) to convey voice applications over their available data networks. Not only will this technological advancement have an effect on the large conventional telecommunications industry, it will change the cost and pricing structures of conventional telephony [9]. Moreover, when this is compared with circuit-switched services, IP networks can contain 5 to 10 times the number of voice calls over the same bandwidth.

III. VOIP CODECS

Before to the transmission of a voice call over an IP-based network a person's voice, (which is an analogue sound wave) must be changed into a digital form and encoded. A limited amount of data compression can also take place for saving bandwidth during the later transmission. On receiving the voice data at the other side, this process must be reversed. Various voice-encoding algorithms are utilized known as G-Series. The general ones are G.711, which is widely used in the telecommunications industry within PSTN networks, and G.72912. Codecs are different in the algorithms they utilize for sampling the analogue voice wave and the sophistication of the compression utilized. This in turn finds the amount of digital bandwidth needed for the encoded sample. G.711, for instance needs a comparatively higher bandwidth while G.729 works at 8Kbps. However, finally there is a exchange between the quality of the voice signal obtained, the sophistication of the algorithms and the amount of bandwidth need.

IV. PERFORMANCE METRICS

In our simulations, we employ the following metrics to measure the performance of WiMAX network with respect to end-to-end QoS for VoIP.

4.1 Mean Opinion Score (MOS)

MOS offers a numerical evaluation of the quality of human speech in voice telecommunications, with value range from 1 to 5 where 5 is the best quality and 1 is the worst quality.

4.2 Packet end-to-end delay (De2e)

The total voice packet delay; De2e computed utilizing the formula:

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

where De, Dn, Dc, Dd and Dde represent the encoding, network, decoding, compression and decompression delay respectively[15].

4.3 Jitter

Jitter defined as the highest difference in one-way delay of the packets in a specific time interval.

4.4 Packet delay variation (PDV)

PDV defined as the variance of the packet delay, which can be, computed from the following Eq.

$$PDV = \frac{\sum_{i=1}^{n}([t'(n)-t(n)]-\mu)^2}{n}$$

V. SIMULATION AND RESULT ANALYSIS

This implementation was made employing OPNET 14.5 using Windows-7, 64-bit operating system.

5.1 MOS

Figures 1, 2 and 3 shows the mean MOS for various scenarios. A major realization is that the mean MOS reduces with the increase of the distance between SS and BS and utilizing sector antenna type and the mean MOS for transmission power = 0.4 watt better than transmission power = 0.5 or 0.6 watt.

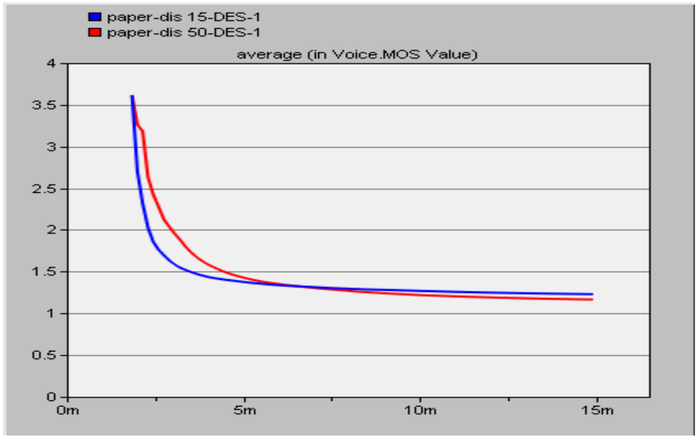

Figure.1MOS (Distance)

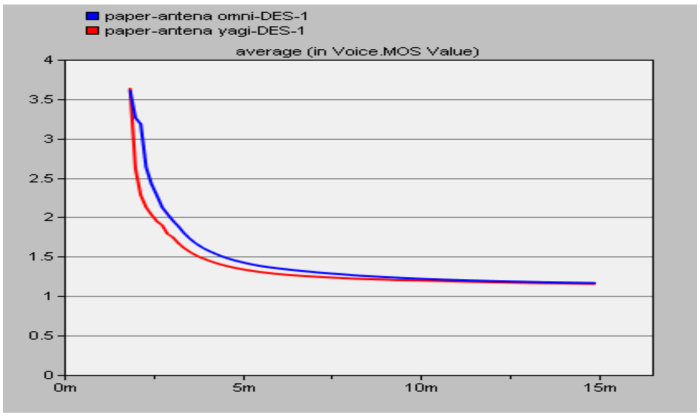

Figure.2MOS (Antenna)

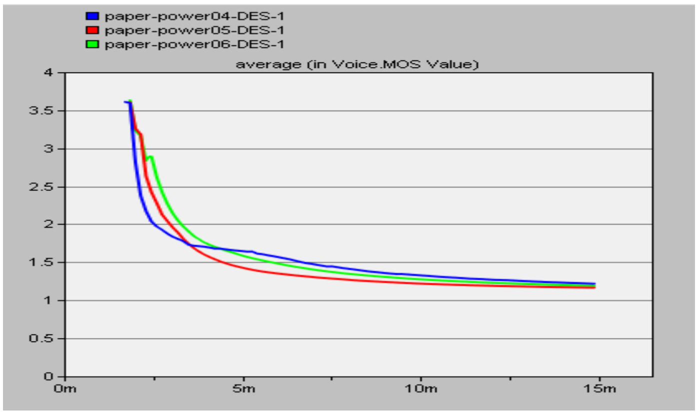

Figure.3MOS (Power comparison)

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5.2 Packet end-to-end delay

Packet end-to-end delay is one of the most significant performances Metric in VoIP. Figures 4, 5, Figure 6 indicate the mean packet end-to-end delay, which increased 80% when increased the distance between SS and BS from 15km to 50 km, and reduced by utilizing sector antenna and increased the transmission power of both SS and BS.

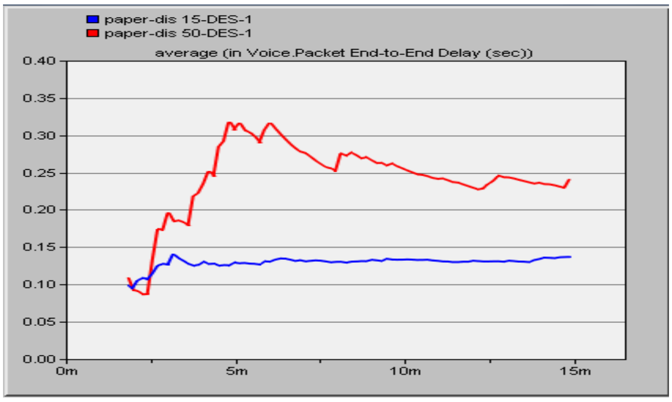

Figure.4 voice packet End-to-End delay(Distance comparison)

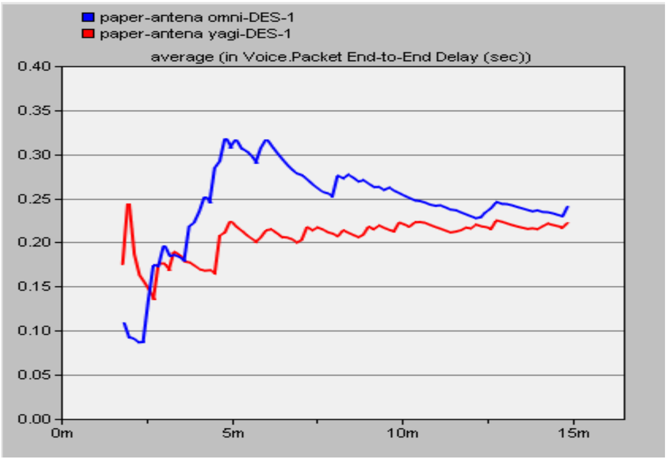

Figure.5 voice packet End-to-End delay(Antenna comparison)

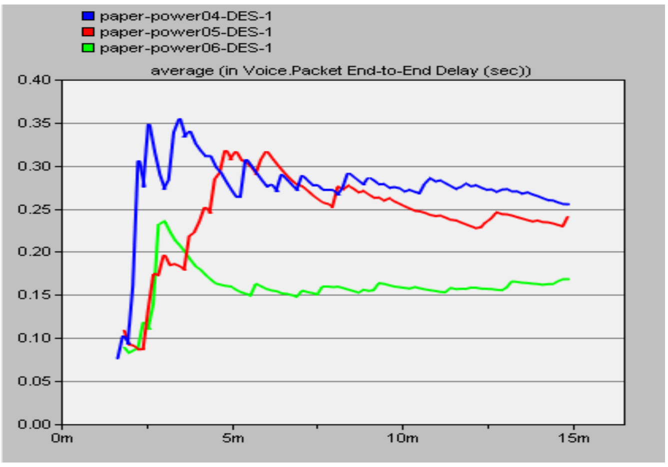

Figure.6 voice packet End-to-End delay (Power comparison)

5.3 Jitter

The jitter value can be negative which means that the time difference between the packets at the destination node is less than that at the source node. Figures 7, 8 and Figure 9 show the jitter

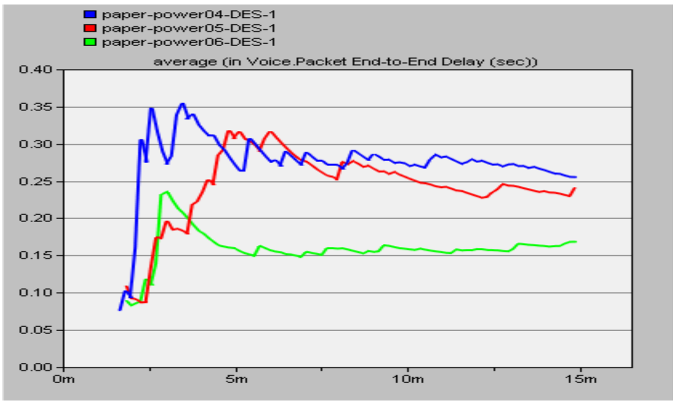

Figure.7 voice packet End-to-End delay (Power comparison)

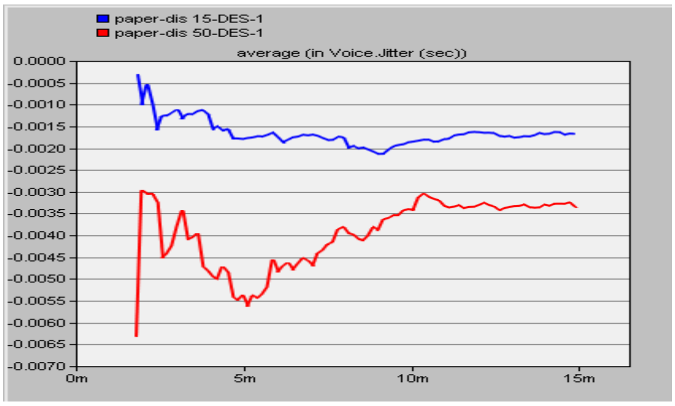

Figure.8 voice jitter (distance comparison)

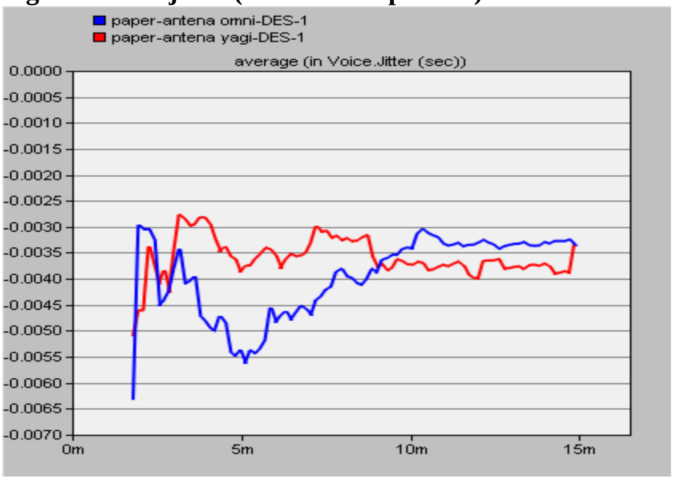

Figure.9 voice jitter (Antenna comparison)

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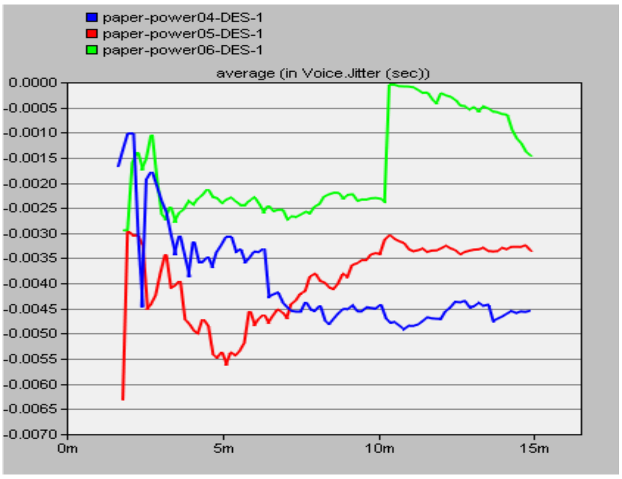

Figure.10 voice jitter (Power comparison)

5.4 Packet delay variation (PDV)

Packet delay variation plays an important role in the network performance reduction and influences the user-perceptual quality. Higher packet delay variation affects in congestion of the packets results shows in the network overhead. Figures 10, 11 and Figure 12 show packet delay variation for various scenarios.

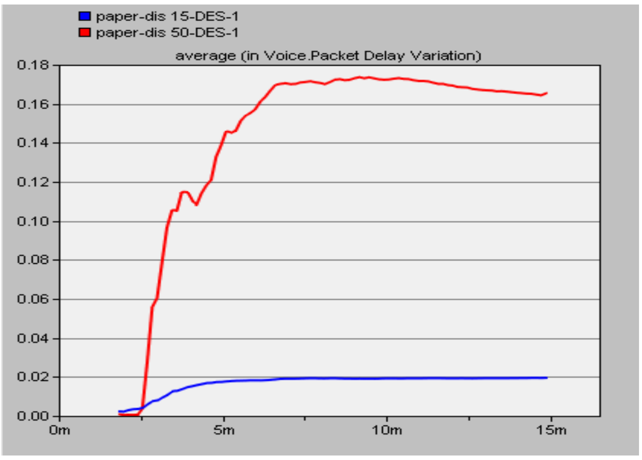

Figure.11 packet delay variation (Distance comparison)

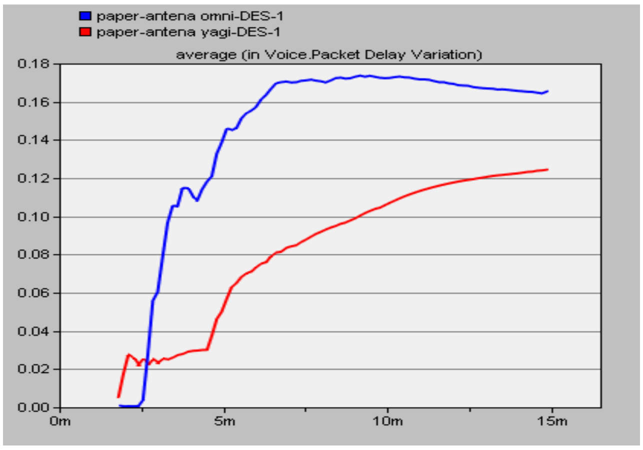

Figure.12 packet delay variation (Antenna comparison)

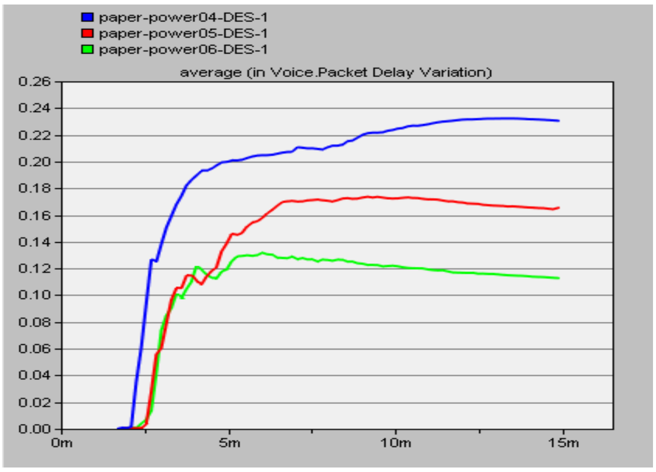

Figure.13 packet delay variation (power comparison)

CONCLUSION

4G generation networks with various technologies provide several multimedia services to the user. It also offers the luxury of using the best existing technology for the needed service to a user, business organizations and companies. In this study, we have carried out wide simulation study to measure the WiMAX performance for serving VoIP traffic. We have examined various significant vital parameters i.e. end-to-end delay, MOS, jitter and packet delay variation. Simulation results describe that with increasing the distance between SS and BS, the mean MOS reduced and the average delay raised. If we utilize sector antenna the mean MOS nearly as it was in Omni antenna but the average delay reduced, at last when we increased the transmission power, the average MOS for transmission power = 0.4 watt better as compared to transmission power = 0.5 or 0.6 watt.

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