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Analysis QoS of Voice over WiMAX Networks using Voice Codec's

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Abstract
Real-time services i.e. VoIP are becoming famous and are greater revenue earners for network service suppliers. These facilities are no longer limited to the wired domain and are being explored over wireless networks. Although some of the available wireless technique can support many low-bandwidth applications, the bandwidth requirements of some multimedia applications increase these techniques capacity. The IEEE 802.16-based WiMAX predicts to be one of the wireless access techniques capable of supporting higher bandwidth applications. In this paper, we exploit the rich set of reliable characteristics provided at the WiMAX medium access control (MAC) layer for building and transmission of MAC protocol data units (MPDUs) for supporting numerous VoIP streams. We explain the VoIP calls quality, often provided by R-score, with respect to the loss and delay of packets. We examine the quality of service (QoS) on long distance data transmission between two positions with VoIP over WiMAX will be performed. Performance of chosen parameters will be performed utilizing the network modeler, OPNET Simulator 14.5 [1,2].

Keywords: VOIP, WiMAX, ACK, QPSK, QAM, OPNET, LTE

I. INTRODUCTION

In spite the increasing popularity of data facilities, voice services still remain the greater revenue earner for network service supplier. The two most famous ways of offering voice facilities are packet switched telephone networks (PSTNs) and wireless cellular networks. The deployment of both of these networks forms needs infrastructures that are often very costly. Alternative solutions are being looked which can provide good-quality voice facilities at a comparatively lower cost. One way to obtain low cost is to utilize the already available IP infrastructure. Protocols that are utilized to carry voice signals throughout the IP network are generally known as voice-over-IP (VoIP) protocols. Supporting real time applications across the Internet has several challenges [4]. Facilities i.e. VoIP need least service assurance that goes beyond the best effort structure of current IP networks. Although some codec's are able of some levels of adaptation and error concealment, the VoIP quality remains sensible to performance reduction in the network. Maintaining good quality VoIP calls becomes even more challenging task when the network is explored to the wireless domain, either through 802.11-based wireless LANs or third-generation (3G) cellular networks [3,5,6]. This wireless extension of facilities is becoming more necessary as there is already a high need for real-time facilities throughout wireless networks. Although bare fundamental versions of facilities i.e. streaming audio, real-time news and video on demand are currently being supported, the high usage and bandwidth requirements of these multimedia applications far exceed the current 3G cellular and wireless LAN technologies capacity. Furthermore, most access techniques do not have the alternative to distinguish particular application needs or subscriber requirements. Wither fast development of wireless technologies, the task of offering broadband last mile connectivity is still a challenging task. The last mile is commonly referred to as a link from a service supplier's network to the subscriber, either a business or a residential home facility. Among the novel wireless broadband access techniques that are being taken, worldwide interoperability of microwave access (WiMAX) is perhaps the strongest competitor that is being supported and formulated by company consortium [2]. In this paper, we explain the probability of supporting VoIP streams over WiMAX and recommend means through which several VoIP streams quality can be enhanced. Particularly, the presentation of this paper is explained as follows: In Section 2, we offer a brief WiMAX overview. In Section 3, we explain the rich set of MAC layer characteristics of WiMAX, with specific emphasis on fragmentation and aggregation. In Section 4, we offer a brief QoS overview in IEEE802.16. In Section 5, we represent the impact of loss and delay on R-score, which is a metric utilized to show the VoIP quality. In Section 6, 7 we show the simulation model and results. Conclusions are provided in Section 8.

II. VOICE OVER IP (VOIP)

Voice over IP (VoIP) facilities have been importantly obtaining prominence across the last few years due to a no. of impressive benefits over their conventional circuit-switched counterparts involving but not restricted to high bandwidth efficiency, low cost, and reliability of utilizing several compression techniques. At the same time, the usage of wireless networks has also increased extremely over the past years. Wireless LAN (WLAN) solutions, integrated with better physical layer techniques, now promise high data rates of greater than 100 Mbps i.e. IEEE 802.11n. In this context, current attempts have concentrated on marrying the potential advantages of VoIP and WLANs to offer wireless telephone facilities. A natural question that then arises is *how well does VoIP perform over WLAN environments*? The answer of this question is counter-intuitive. Even though the WLANs boast very high data rates and a ganeral VoIP call carries only 128 Kbps of bidirectional data (utilizing G.711 voice codec), the no. of VoIP calls survived by these networks is terribly low. An IEEE 802.11b network for example can survive only 5 VoIP calls even at the greatest data rate of 11 Mbps, as we will present later. VoIP traffic in WLANs is featured by its small frame sizes, and IEEE 802.11 MAC is known widely for very worst performance for small frame sizes. For small frames the overheads at the various layers of the network stack themselves introduce an important burden. Addition to this IEEE 802.11 MAC protocol has other distributed impacts that further decrease the VoIP call capacity.

III. WiMAX OVERVIEW

WiMAX is a wireless metropolitan access network (MAN) technique that depends on the standards described in the IEEE 802.16 specification. This standard based technique is not only a simplifying but also a unifying deployment and development of WiMAX. The 802.16 standard can be utilized in a point-to-point or mesh configuration utilizing pairs of directional antennas. These antennas can be employed to increase the efficient range of the system relative to what can be obtained in the point-to-multipoint mode. WiMAX is seen as a solution to the outdoor broadband wireless access that is able of providing high-speed streaming data. It has the ability of providing high speed facilities up to 30 miles range, hence introducing strong competition to the available last mile broadband access technique, i.e. DSL and cable. WiMAX utilizes various channels for a single transmission and offers bandwidth of up to 100 Mbps [7]. The usage of orthogonal frequency-division multiplexing (OFDM) increases the data capacity and bandwidth by locating channels very near to one another and still avoids disturbance due to orthogonal channels. A typical WiMAX BS offers enough bandwidth to satisfy the needs of more than 50 businesses with T1-level (1.544Mbps) facilities and hundreds of homes with high speed Internet access. WiMAX's low cost of deployment integrated with available needs from underserved region generates major business opportunities.

IV. THE MAC LAYER OF WIMAX

WiMAX provides some reliable characteristics that can powerfully be exploited for providing real time facilities. Particularly, although the WiMAX MAC layer has been standardized, there are specific characteristics that can be tuned and build application and/or channel specific [8],[9]. For instance, the MAC layer does not confined itself to static size frames but permits variable-sized frames to be built and transferred. Let us first explain the WiMAX MAC layer.

The WiMAX MAC layer is comprised of three sub layers which communicate with one another through the service access points (SAPs), as illustrated in Figure.1. The service specific convergence sub layer offers the mapping or transformation of external network data with the SAP support. The MAC common part sublayer obtains this information in the MAC service data units (MSDUs) form, which are packed into the payload fields to made MPDUs. The privacy sublayer offers secure key exchange, authentication and encryption on the MPDUs and passes them across to the physical layer. Of the three sub layers, the common part sub layer is the core functional layer which offers bandwidth and sets up and manages link. Furthermore, as the WiMAX MAC offers a connection-oriented service to the user stations, the common part of sublayer also offers a connection identifier (CID) to determine which link the MPDU is supporting.

Let us explain the common part sublayer and its rich set of characteristics. This sublayer controls the on-air timing depending on consecutive frames that are partitioned into time slots. The frames size and the individual slots size within these frames can be changed on a frame-by-frame basis. This permits efficient assignment of on-air resources which can be used to the MPDUs to be transferred. Based on the feedback obtained from the recipient and on-air physical layer slots, the MPDU size can be optimized. In this research paper, we exploit the characteristics of the common part sublayer that alters the MPDUs size to adapt to the changing channel conditions.

4.1 Aggregation

The common part sublayer is able of packing more than one partial or complete MSDUs into one MPDU. In Figure 2, we represent how the MPDU payload can achieve more than two complete MSDUs, but not three. Thus, a part of the third MSDU is packed with the prior two MSDUs to fill up the rest payload field, preventing resources wastage. The size of payload is determined by on air timing slots and feedback obtained from the user station.

4.2 Fragmentation

The common part sublayer can also fragment an MSDU into several MPDUs. In Figure 3, we represent how a part of a single MSDU uses the complete payload field of an MPDU. Here, the payload field of the MAC packet data unit to be transferred is too small to achieve a complete MSDU. In that case, we divide a single MSDU and pack the fragmented part into the MPDU payload field.

V. MOTIVATION AND OVERVIEW

We can imagine 5 different techniques to enhance the call capacity by leveraging various components in the equation:

ACK Aggregation (AA)**:** ACK aggregation refers to forwarding a single ACK for a block of n frames. The results of ACK in the *decrement of* TACK.

Frame Aggregation (FA)**:** Frame aggregation means fusing numerous frames targeted to the same end subscriber into a single large frame.

Link Adaptation (LA)**:** Link adaptation means altering the transmission rate for the data frames. The 802.11b standard mentions 4 different data rates that can be utilized. The prevailing channel situations influence the data rate choice.

Time Saving (TS)**:** Time saving refers to *decreasing* TDIFS waiting time between two successive frames.

Header Compression (HC)**:** Header compression refers to decreasing the several headers size like the UDP/RTP/IP headers utilizing the mechanisms either introduced in literature or otherwise. This scheme also has restricted power in enhancing the call capacity in tune of only 0.1 calls.

A. ACK Aggregation (AA)

Adaptive ACK Aggregation Algorithm: Depending on the above results that there are cross over points we can imagine of an algorithm that alters the block size adaptively. We model this technique as a 2.5 layer solution in between the *interface queue and* MAC layer. We consider that there is no preset block size and we forward a block ACK request from the source node once all frames in the existed block are forwarded. Upon obtaining the block ACK request, the target node replies with a block ACK consisting the needed information. The source then initiates a new block and retransfers the needed frames of the prior block and proceeds with newer frames. Due to the existence of a block ACK request we can change the *block* size at will. We implement a simple adaptive technique where we increase the size of block upon obtaining a block ACK with all successes and decrease the size on obtaining a block ACK with even a single data loss. The explained algorithm is presented in Figure 3. The adaptive algorithm selects the right block size based on the no. of losses in the current block. Hence the adaptive algorithm should provide a performance that is the best of both worlds such as it should have a good delay feature as well as high saturated capability.

VI. QUALITY OF SERVICE IN IEEE 802.16

Originally, four different service kinds were provided in the 802.16 standard: UGS, rtPS, nrtPS and BE. The UGS (Unsolicited Grant Service) is same as the CBR (Constant Bit Rate) service in ATM, which creates a static size burst periodically. This facility can be utilized to substitute T1/E1 wired line or a constant rate service. It also can be utilized to support real time applications i.e. streaming or VoIP applications. Even though the UGS is simple, it may not be the best selection for the VoIP in that it can waste bandwidth during the voice calls off period. The rtPS (real-time polling service) is for a variable bit rate real-time facility i.e. VoIP. Each polling interval, BS polls a mobile and the polled mobile transfers bw_request (bandwidth request) if it has data to transfer. The BS grants the data burst utilizing UL-MAP-IE upon its reception. The nrtPS (non-real-time polling service) is very same as the rtPS except that it perits contention based polling. The BE (Best Effort) facility can be utilized for applications i.e. FTP or e-mail, in which there is no severe latency need. The allocation technique is contention based utilizing the ranging channel. Another service type known as ertPS (Extended rtPS) [10] was proposed to support variable rate real-time facilities i.e. video streaming or VoIP. It has a benefit over rtPS and UGS for VoIP applications because it carries lower overhead as compared to rtPS and UGS.

VII. SYSTEM MODEL AND IMPLEMENTATION

OPNET Simulator 14.5 was utilized to model the two-way VoIP calls build by subsribers on WiMAX network [11, 13]. Ten scenarios were implemented, and their results of simulation were compared to examine the impact on total performance because of environmental factors. Every scenario has a conversation pair, one workstation being the called and the other one being the caller. The caller begins forwarding data packets to the called through the WiMAX BS at 100 seconds after the simulation has began, and the called responses to the caller through the WiMAX BS to make a two-way communication. All nodes utilized in simulations can be detected in built-in OPNET library WiMAX.

Experimental Setup: We carried out a testbed to study the 802.11b network call capacity. The experimental establishment of the testbed is presented in Figure. 1. A total of two laptops, six desktops, two routers, and one 802.11b AP are utilized for the testbed. All the machines operate Linux operating system. The testbed contains two domains: a wired domain and a wireless domain. To compete several calls in the wireless domain we utilize three wireless interface cards in a single machine and operate three virtual machines on the physical machine, each related with a various wireless card. Two such machines have three wireless cards, respectively. The laptop linked to the AP is utilized as a real VoIP phone, and a VoIP call utilizing the KPhone [1] and SIP Express Router [2] is established between this laptop and another laptop linked to the wired domain. Other calls are competed utilizing bidirectional constant bit rate (CBR) traffic created by Iperf [3]. The frame size of every UDP packet created by Iperf is adjusted to be 92 bytes (involving 12 byte RTP header) and the data rate is adjusted to be 73.6 Kbps.

VIII. SIMULATION PARAMETERS

In our simulations, we utilize the following four metrics to measure the WiMAX performance in terms of end-to end QoS for VoIP.

(1) Mean Opinion Score (MOS): MOS offers a numerical measure of the quality of human speech in voice telecommunications, with value ranging from 1 to 5 where 5 is the best quality and 1 is the worst quality. In our simulation, we evaluate MOS through a non-linear mapping from R-factor as in:

$MOS = 1 + 0.035R + 7 * 10{-}6R$ (where R = 100 - Is - Ie - Id + A. Is is the impact of impairments that take place with the voice signal; Ie is the impairments caused by several types of losses takes place because of network and codecs, and Id shows the impairment caused by delay especially mouth to ear delay. Utilizing the default setting for Is and A, Eqn 1 can be decreased to R = 94.2 Ie Id.

(2) Packet end-to-end delay: The total voice packet delay is computed as:

De2e = Dn + De + Dd + Dc + Dde

where Dn, De, Dd, Dc and Dde show the network, encoding, decoding, compression and decompression delay, respectively.

(3) Jitter: In OPNET, jitter is calculated as the signed maximum difference in one way packets delay over a specific time interval. Let t(i) and t0(i) be the time transferred at the transmitter and the time obtained at the receiver, respectively. Jitter is computed as follows:

jitter = $maxi$=1 n ([t'(n)-t'(n-1)]-[t(n)-t(n-1)])

XI. NETWORK TOPOLOGY

The network topology contains a WiMAX Base Station and two WiMAX user Stations, one being the called and the other one being the caller. Ten scenarios will be described: (i) Scenarios from 1to 3: In the first three scenarios, both of the user stations (called and caller) were being kept static at 50 km, 100 km, and 200km respectively away from the BS. The nodes were kept static throughout the simulation; (ii) Scenarios from 4 to 7: In the four scenarios, both of the user stations (called and caller) were being kept static at 50 km and power 0.5 watt, but we change the modulation on every scenario as adaptive modulation, QPSK, 16-QAM and 64- QAM respectively, and (iii) Scenarios from 8 to 10: In the last three scenarios, both of the user stations (called and caller) were being kept static at 100 km , but we vary the maximum transmission power in both of the user stations (called and caller) and BS as 0.5 watt ,1.0 watt, 2,0 watt.

Table 1: Simulation Parameters for WiMAX

Cell Radius	15 km
No. of Base Stations	8
No. of Subscriber Stations per BS	10
Speed of the mobile nodes	50, 100, 150 m/s
Simulation time	1800 sec
Base Station Model	wimax_bs_ethernet4_slip 4_router
Subscriber Station Model	wimax_ss_wkstn
IP Backbone Model	ip32_cloud
Voice Server Model	ppp_server
Link Model (ASN - Backbone)	PPP_SONET_OC12
Physical Layer Model	OFDMA 20Mhz
MAC Protocol	IEEE 802.16e
Multipath Channel Model	ITU Vehicular A
Traffic Type of	Interactive Voice and

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Service	Data
Scheduling Type	ertPS, nrtPS
Application	IP telephony
Voice Codec (with and without silence suppression)	G 711, G.729, G.723, G.726,G.728
Inter repetation time	Constant 300

A Voice Codec utilized at the client side to change over the simple voice waves into advanced heartbeats and the other way around. There are different codecs sorts in view of the picked information rate, inspecting rate, and executed pressure calculation portrayed in table.2

Table2:Various Codecs

Codec	Coding rate
G.711	64
G.722	48
G.723	5.3 AND 6.4
G.726	16, 24
G.727	16 TO 40
G.728	16
G.729	8

X. SIMULATION RESULTS

The simulation duration for all four scenarios was 15 min to every scenario. The simulation parameters that are most interesting in this paper are end to end delay, throughput and MOS value.

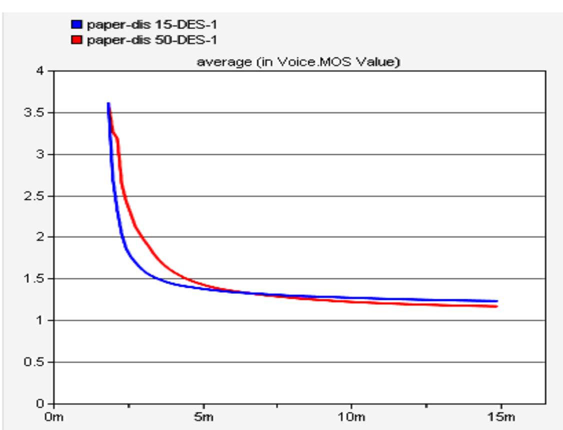

Figure.1MOS (Distance)

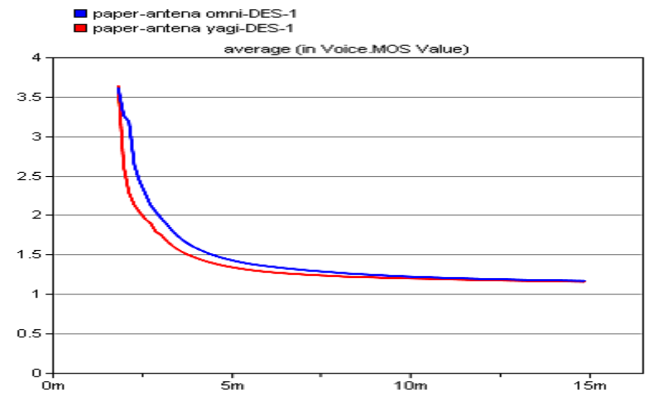

Figure.2MOS (Antenna)

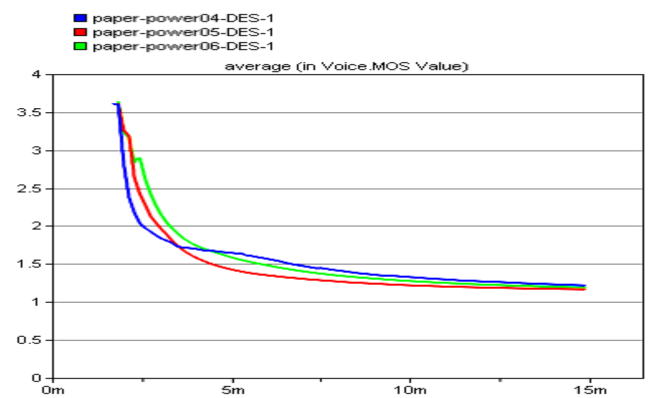

Figure.3MOS (Power comparison)

10.1 Packet end-to-end delay

Packet end-to-end delay is one of the most significant performances Metric in VoIP.

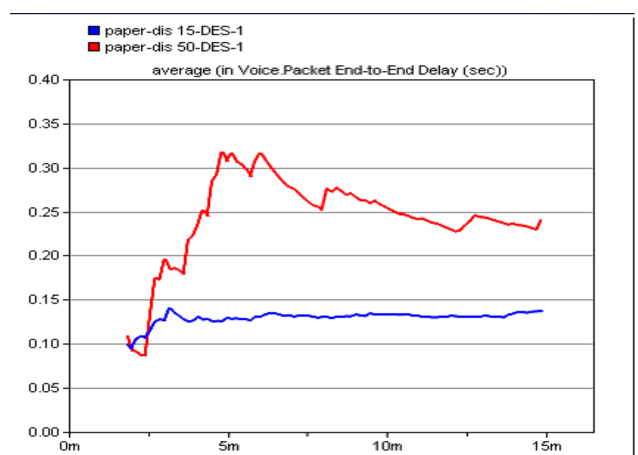

Figure.4 voice packet End-to-End delay(Distance comparison)

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Figures 4, 5, Figure 6 indicate the mean packet end-to-end delay, which increased 74% when increased the distance between SS and BS from 10 km to 15 km, and reduced by utilizing sector antenna and increased the transmission power of both SS and BS.

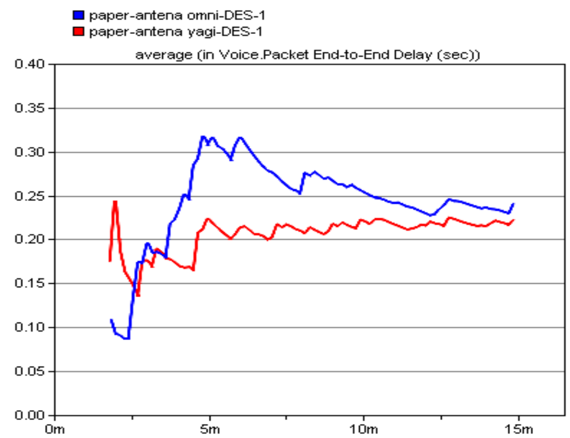

Figure.5 voice packet End-to-End delay(Antenna comparison)

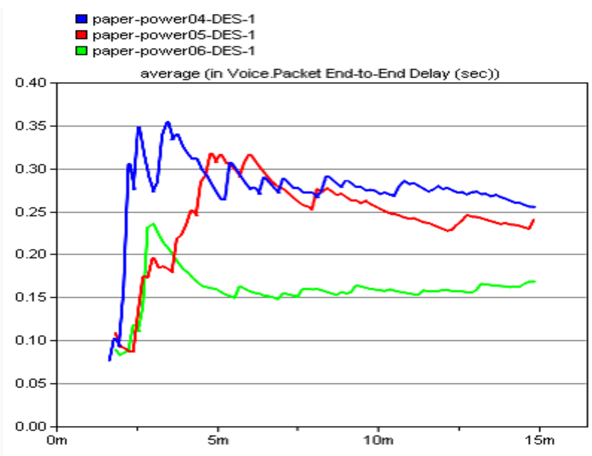

Figure.6 voice packet End-to-End delay (Power comparison)

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

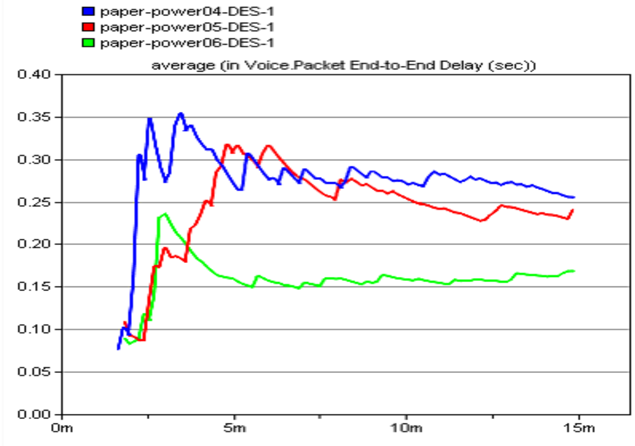

Figure.7 voice packet End-to-End delay (Power comparison)

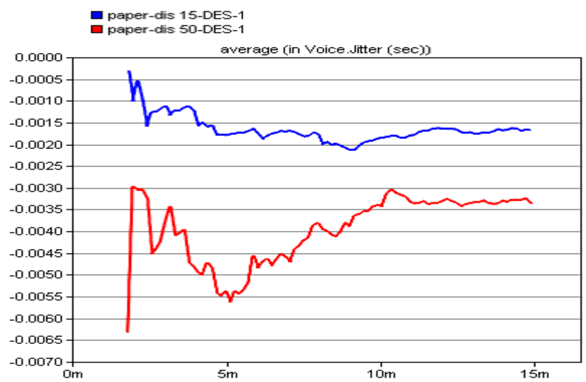

Figure.8 voice jitter (distance comparison)

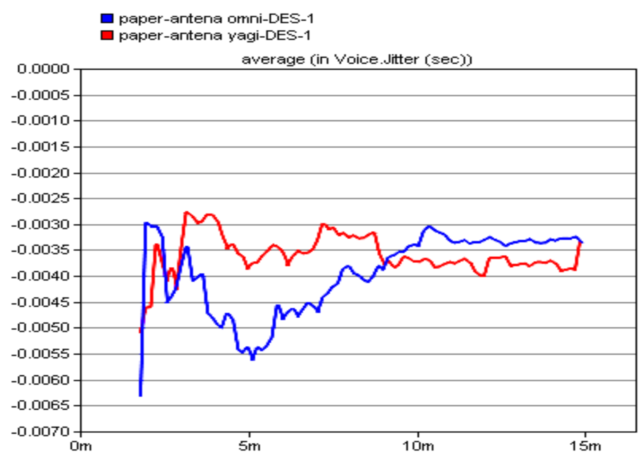

Figure.9 voice jitter (Antenna comparison)

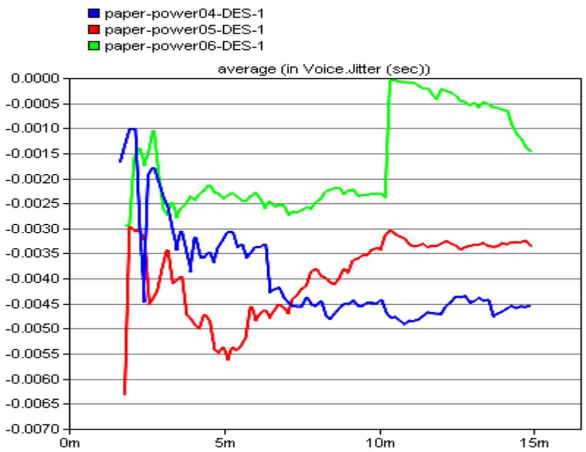

Figure.10 voice jitter (Power comparison)

CONCLUSIONS

A new wireless access techniques are being formulated, WiMAX is evolving as one of the predicting broadband techniques that can support several real-time facilities. However, the extension of VoIP calls over wireless networks is inevitable, we study the viability of supporting VoIP across WiMAX. This paper shows our development to VoIP in measuring the IEEE 802.16 performance. We have utilized the WiMAX Connection statistics (such as throughput, load) , and VoIP connection statistics (such as MOS value, jitter, End-to-End delay). The VoIP performance based on WiMAX, was measured and assessed at different: (i)distance ; (ii) Antenna modulation techniques; and (iii) power; As a result of the relative study, it was determined that : (i) In the three first scenario, When distance is 50 km (in WiMAX range) it has the best performance and the minimum delay value compared with 100km and 200 km (out of WiMAX range); (ii) In the four scenarios following We observe that the two scenario who employed 16-QAM and 64-QAM modulation technique have high performance ,but the two other scenario who utilize adaptive and QPSK technique have poor performance; and (iii)In the last three scenario, the performance is based on power value; when power is large, the performance is high and when power is low

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