Improving Quality of VoIP over WiMAX

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Abstract

Real-time services such as VoIP are becoming popular and are major revenue earners for network service providers. These services are no longer confined to the wired domain and are being extended over wireless networks. Although some of the existing wireless technologies can support some low-bandwidth applications, the bandwidth demands of many multimedia applications exceed the capacity of these technologies. The IEEE 802.16-based WiMAX promises to be one of the wireless access technologies capable of supporting very high bandwidth applications. In this paper, we exploit the rich set of flexible features offered at the medium access control (MAC) layer of WiMAX for the construction and transmission of MAC protocol data units (MPDUs) for supporting multiple VoIP streams. We discuss the quality of VoIP calls, usually given by R-score, with respect to the delay and loss of packets. We analysis the quality of service (QoS) on long distance data transfer between two locations with VoIP over WiMAX will be performed. Performance of selected parameters will be done using the network simulator, OPNET Modeler 14.5 [1,2].

Keywords: *Voice over Internet Protocol (VoIP); R-score; worldwide interoperability of microwave access (WiMAX); Forward error correction (FEC); Automatic repeat request (ARQ)*

1. Introduction

DESPITE the growing popularity of data services, voice services still remain the major revenue earner for network service providers. The two most popular ways of providing voice services are packet switched telephone networks (PSTNs) and wireless cellular networks. The deployment of both of these forms of networks requires infrastructures that are usually very expensive. Alternative solutions are being sought which can deliver good-quality voice services at a relatively lower cost. One way to achieve low cost is to use the already existing IP infrastructure. Protocols that are used to carry voice signals over the IP network are commonly referred to as voice-over-IP (VoIP) protocols. Supporting real time applications over the Internet has many challenges [4]. Services such as VoIP require minimum service guarantees that go beyond the best effort structure of today's IP networks. Although some codec's are capable of some levels of adaptation and error concealment, the VoIP quality remains sensitive to performance degradation in the network. Sustaining good quality VoIP calls becomes even more challenging when the IP network is extended to the wireless domain, either through 802.11-based wireless LANs or third-generation (3G) cellular networks [3,5,6].Such wireless extension of services is becoming more essential as there is already a huge demand for real-time services over wireless networks. Although bare basic versions of services such as real-time news, streaming audio, and video on demand are currently being supported, the widespread use and bandwidth demands of these multimedia applications far exceed the capacity of current 3G cellular and wireless LAN technologies. Moreover, most access technologies do not have the option to differentiate specific application demands or user needs. Wither rapid growth of wireless technologies, the task of providing broadband last mile connectivity is still a challenge. The last mile is generally referred to as a connection from a service provider's network to the user, either a residential home or a business facility. Among the new wireless broadband access technologies that are being considered, worldwide interoperability of microwave access (WiMAX) is perhaps the strongest contender that is being supported and developed by a consortium of companies [2]. In this paper, we explore the possibility of supporting VoIP streams over WiMAX and suggest means through which the quality of multiple VoIP streams can be improved. Specifically, the organization of this paper is listed as follows: In Section 2, we provide a brief overview of WiMAX. In Section 3, we discuss the rich set of MAC layer features of WiMAX, with particular emphasis on aggregation and fragmentation. In Section 4, we provide a brief overview of QoS in

IEEE802.16. In Section 5, we show the effect of delay and loss on R-score, which is a metric used to represent the quality of VoIP. In Section 6, 7 we present the simulation model and results. Conclusions are drawn in Section 8.

2. WiMAX overview

WiMAX is a wireless metropolitan access network (MAN) technology that is based on the standards defined in the IEEE 802.16 specification. This standard based approach is not only a simplifying but also a unifying development and deployment of WiMAX. The 802.16 standard can be used in a point-to-point or mesh topology using pairs of directional antennas. These antennas can be used to increase the effective range of the system relative to what can be achieved in the point-to-multipoint mode. WiMAX is envisioned as a solution to the outdoor broadband wireless access that is capable of delivering high-speed streaming data. It has the capability of delivering high-speed services up to a range of 30 miles, thus posing strong competition to the existing last mile broadband access technologies, such as cable and DSL. WiMAX uses multiple channels for a single transmission and provides bandwidth of up to 100 Mbps [7]. The use of orthogonal frequency-division multiplexing (OFDM) increases the bandwidth and data capacity by spacing channels very close to each other and still avoids interference because of orthogonal channels. A typical WiMAX base station provides enough bandwidth to cater to the demands of more than 50 businesses with T1-level (1.544Mbps) services and hundreds of homes with high speed Internet access. WiMAX's low cost of deployment coupled with existing demands from underserved areas creates major business opportunities.

3. The MAC layer of WiMAX

WiMAX offers some flexible features that can potentially be exploited for delivering real time services. In particular, although the MAC layer of WiMAX has been standardized, there are certain features that can be tuned and made application and/or channel specific [8],[9]. For example, the MAC layer does not restrict itself to fixed-size frames but allows variable-sized frames to be constructed and transmitted. Let us first discuss the MAC layer of WiMAX.

The MAC layer of WiMAX is comprised of three sub layers which interact with each other through the service access points (SAPs), as shown in Fig.1. The service-specific convergence sub layer provides the transformation or mapping of external network data with the help of the SAP. The MAC common part sublayer receives this information in the form of MAC service data units (MSDUs), which are packed into the payload fields to form MPDUs. The privacy sublayer provides authentication, secure key exchange, and encryption on the MPDUs and passes them over to the physical layer. Of the three sub layers, the common part sub layer is the core functional layer which provides bandwidth and establishes and maintains connection. Moreover, as the WiMAX MAC provides a connection-oriented service to the subscriber stations, the common part sublayer also provides a connection identifier (CID) to identify which connection the MPDU is servicing.

Let us discuss the common part sublayer and its rich set of features. This sublayer controls the on-air timing based on consecutive frames that are divided into time slots. The size of these frames and the size of the individual slots within these frames can be varied on a frame-by-frame basis. This allows effective allocation of on-air resources which can be applied to the MPDUs to be transmitted. Depending on the feedback received from the receiver and on-air physical layer slots, the size of the MPDU can be optimized. In this paper, we exploit this feature of the common part sublayer that modifies the size of the MPDUs to adapt to the varying channel conditions.

3.1 Aggregation

The common part sublayer is capable of packing more than one complete or partial MSDUs into one MPDU. In Fig. 2, we show how the payload of the MPDU can accommodate more than two complete MSDUs, but not three. Therefore, a part of the third MSDU is packed with the previous two MSDUs to fill up the remaining payload field, preventing wastage of resources. The payload size is determined by on air timing slots and feedback received from the subscriber station.

3.2 Fragmentation

The common part sublayer can also fragment an MSDU into multiple MPDUs. In Fig. 3, we show how a portion of a single MSDU occupies the entire payload field of an MPDU. Here, the payload field of the MAC packet data unit to be transmitted is too small to accommodate a complete MSDU. In that case, we fragment a single MSDU and pack the fragmented part into the payload field of the MPDU.

4. Quality of service in IEEE 802.16

Originally, four different service types were supported in the 802.16 standard: UGS, rtPS, nrtPS and BE. The UGS (Unsolicited Grant Service) is similar to the CBR

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(Constant Bit Rate) service in ATM, which generates a fixed size burst periodically. This service can be used to replace T1/E1 wired line or a constant rate service. It also can be used to support real time applications such as VoIP or streaming applications. Even though the UGS is simple, it may not be the best choice for the VoIP in that it can waste bandwidth during the off period of voice calls. The rtPS (real-time polling service) is for a variable bit rate real-time service such as VoIP. Every polling interval, BS polls a mobile and the polled mobile transmits $bw_{request}$ (bandwidth request) if it has data to transmit.

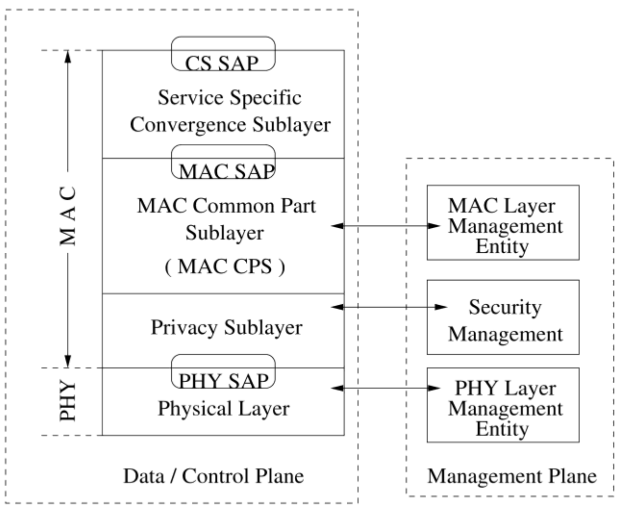

Fig.1 WiMAX MAC layer with SAPs

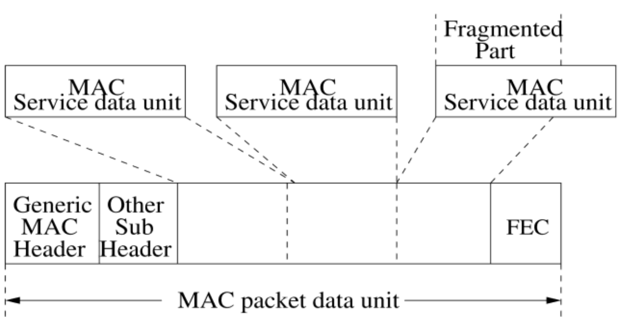

Fig.2 MPDU accommodating multiple MSDUs.

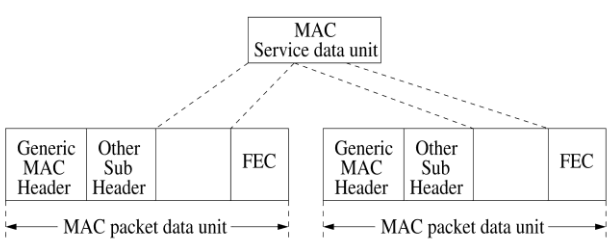

Fig.3 Single MSDU fragmented to multiple MPDUs.

The BS grants the data burst using UL-MAP-IE upon its reception. The nrtPS (non-real-time polling service) is very similar to the rtPS except that it allows contention based polling. The BE (Best Effort) service can be used for applications such as e-mail or FTP, in which there is no strict latency requirement. The allocation mechanism is contention based using the ranging channel. Another service type called ertPS (Extended rtPS) [10] was introduced to support variable rate real-time services such as VoIP and video streaming. It has an advantage over UGS and rtPS for VoIP applications because it carries lower overhead than UGS and rtPS.

5. Delay & loss sensitivity of VoIP

As VoIP packets travel through a network, there evidently are some congestion and channel-related losses. In addition, the packets suffer delay, depending on the congestion at the intermediate routers. Both the loss and delay of packets adversely affect the quality of VoIP calls, which is generally expressed in terms of R-score.

5.1 Quality of VoIP and R-Score

A VoIP application works as follows: First, a voice signal is sampled, digitized, and encoded using a given algorithm/coder. The encoded data (called frames) is packetized and transmitted using RTP/UDP/IP. At the receiver's side, data is depacketized and forwarded to a payout buffer, which smooths out the delay incurred in the network. Finally, the data is decoded and the voice signal is reconstructed.
The quality of the reconstructed voice signal is subjective and is therefore measured by the MOS. The MOS is a subjective quality score that ranges from 1 (worst) to 5 (best) and is obtained by conducting subjective surveys. Although these methods provide a good assessment technique, they fail to provide an online assessment, which might be used for adaptation purposes. The ITU-T E-Model [3] provides a parametric estimation and defines an R-factor that combines different aspects of voice quality impairment. It is given by;

$$R = 100 - I_s - I_e - I_d + A \quad (1)$$

where Is is the signal-to-noise impairments associated with typical switched circuit networks paths, Ie is an equipment impairment factor associated with the losses due to the codec's and network, Id represents the impairment caused by the mouth-to-ear delay, and A compensates for the above impairments under various user conditions and is known as the expectation factor. We note that the contributions to the R-score due to delay and loss impairments are separable. This does not mean that the delay and loss impairments are totally uncorrelated, but their influence can only be measured in an isolated manner. The expectation factor covers intangible and almost impossible to measure quantities like expectation of qualities. However, no such agreement on measurement of expectation on qualities has yet been made and, for this reason, the expectation factor is usually dropped from the R-factor in most studies. The R-factor ranges from 0 to 100 and a score of more than 70 usually means a VoIP stream of decent quality. The R-score is related to the MOS through the following nonlinear mapping [3]:

$$MOS = 1 + 0.035R + 7 * 10^{-6} R(R - 60)(100 - R) \quad (2)$$

for 0 <R<100. If R < 0, the MOS takes the value of 1 and, similarly, if R > 100, the MOS takes the value of 4.5. Among all of the factors in (1), only Id and Ie are typically considered variables in VoIP [8]. Using default values for all other factors, the expression for the R-factor given by (1) can be reduced to [3]

$$\mathrm{R} = 94.2 - Ie - Id \quad (3)$$

Mean Opinion Score (MOS) indicates the quality of received audio after transmission in a numerical measurement. The MOS value is expressed in a scale from the worst (1) to the best (5) which is clarified in Table 1.[10]

Table 1: MOS indication

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Boor	Annoying
1	Bad	Very annoying

6. System model

6.1 Implementation

OPNET Modeler 14.5 was used to simulate the two-way VoIP calls made by users on WiMAX network [11, 13]. Ten scenarios were implemented, and their simulated results were compared to analyze the effect on overall performance due to environmental factors. Each scenario has a conversation pair, one workstation being the caller and the other one being the called. The caller starts sending data packets to the called through the WiMAX base station at 100 seconds after the simulation has started, and the called replies to the caller through the WiMAX base station to form a two-way communication. All nodes used in simulations can be found in built-in OPNET library WiMAX.

6.2 Simulation Parameters

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

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

$$MOS = 1 + 0.035R + 7 * 10^{-6} R(R - 60)(100 - R)$$

where R = 100 - I_s - I_e - I_d + A. Is is the effect of impairments that occur with the voice signal; I_e is the impairments caused by different typesof losses occured due to codecs and network, and I_d represents the impairment caused by delay particularly mouth to ear delay. Using the default setting for Is and A, Eqn 1 can be reduced to R = 94.2 I_e I_d.

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

De2e = Dn + De + Dd + Dc + Dde

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

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

jitter = $max_{i=1}^{n}$([t'(n)-t'(n-1)]-[t(n)-t(n-1)])

(4) Packet delay variation(PDV): PDV in OPNET is defined as the variance of the packet delay, which is computed as follows:

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

where u is the average delay of the n selected packets.

All these parameters of VOIP parameters will be discussed in terms of: (i) distance comparison; (ii) modulation comparison; (iii) antenna comparison, and (iv) power comparison.

6.3 Network Topology

The network topology consists of a WiMAX Base Station and two WiMAX Subscriber Stations, one being the caller and the other one being the called. Ten scenarios will be discussed: (i) Scenarios from 1to 3: In the first three scenarios, both of the subscriber stations (caller and called) were being held fixed at 50 km, 100 km, and 200km respectively away from the base station. The nodes were held fixed throughout the simulation; (ii) Scenarios from 4 to 7: In the four scenarios, both of the subscriber stations (caller and called) were being held fixed at 50 km and power 0.5 watt, but we change the modulation on each 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 subscriber stations (caller and called) were being held fixed at 100 km , but we change the maximum transmission power in both of the subscriber stations (caller and called) and base station as 0.5 watt ,1.0 watt, 2,0 watt. The nodes were held fixed throughout the simulation.

Table 2: states of scenario

No.of scenario	Power	Modulation	Distance	State of scenario
Scenario 1	0.5W	Adaptive	50km	Distance variation
Scenario 2	0.5W	Adaptive	100km	
Scenario 3	0.5W	Adaptive	200km	
Scenario 4	0.5W	Adaptive	100km	Modulation variation
Scenari0 5	0.5W	QPSK	100km	
Scenario 6	0.5W	16-QAM	100km	
Scenario 7	0.5W	64-QAM	100km	
Scenario 8	0.5W	Adaptive	100km	Power variation
Scenario 9	1.0W	Adaptive	100km	
Scenario10	2.0W	Adaptive	100km	

7. Simulation Results

The duration of the simulation for all four scenarios was 15 minutes to each scenario. The parameters that are most interesting in this paper are throughput, end-to-end delay, and MOS value.

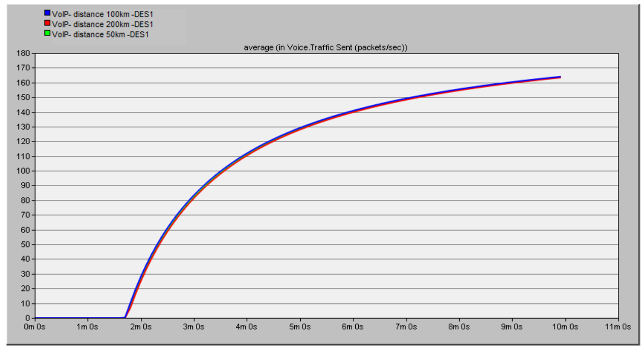

Fig.4 voice traffic sent (packet/sec)[distance variation]

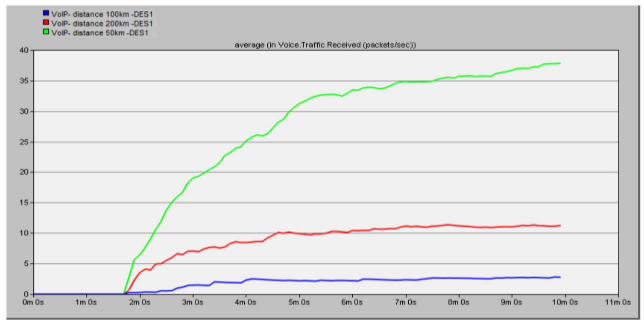

Fig.5 voice traffic received (packet/sec)[distance variation]

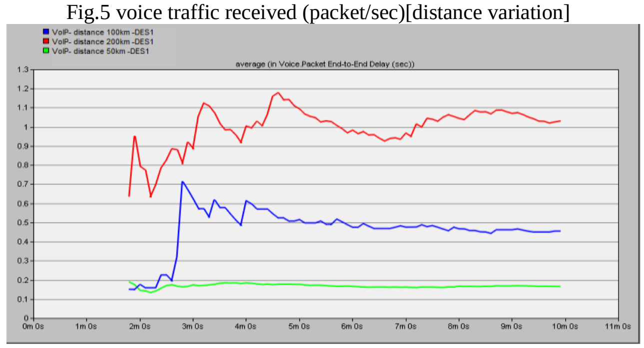

Fig.6 voice packet End-to-End delay (sec)[distance variation]

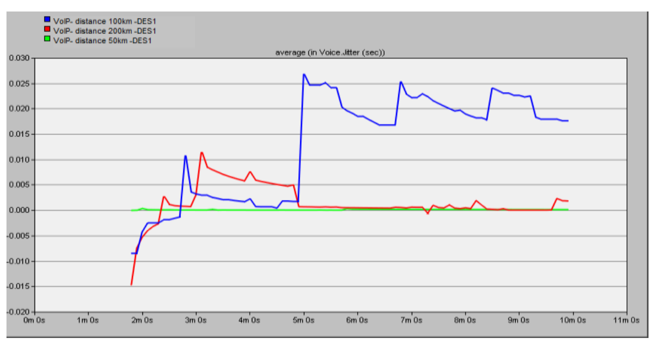

Figure 7. voice jitter (sec) [distance variation]

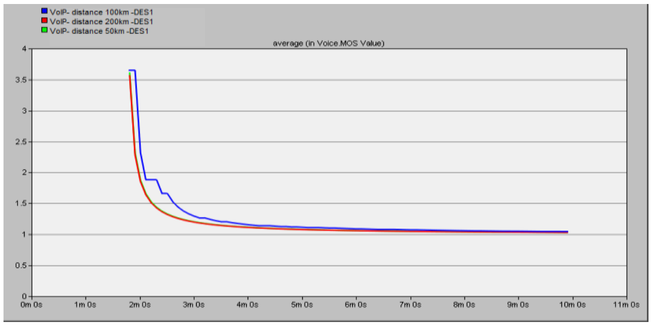

Fig.8 voice MOS value [distance variation]

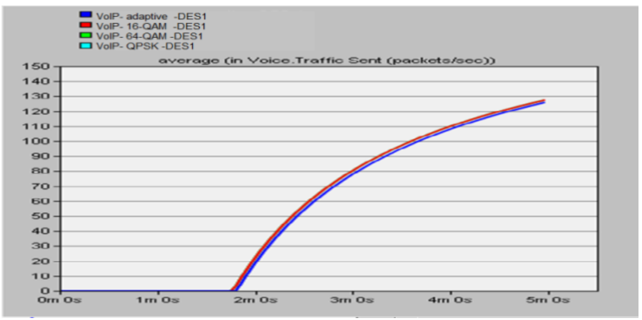

Fig.9 voice traffic sent (packet/sec) [modulation variation]

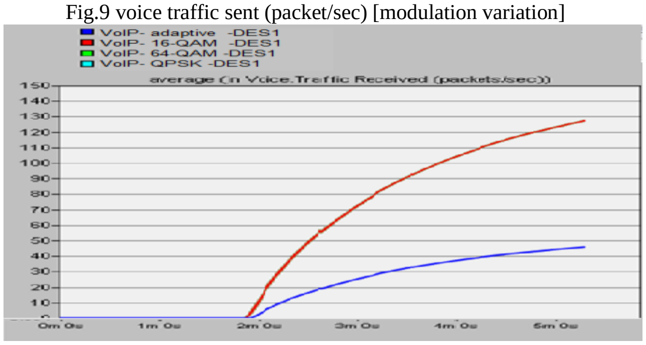

Fig.10 voice traffic received (packet/sec) [modulation variation]

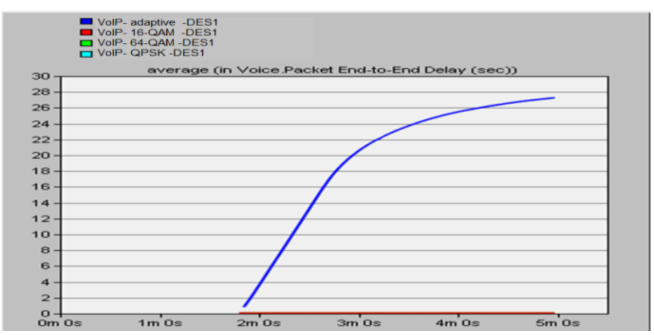

Fig.11 voice packet End-to-End delay (sec)[modulation variation]

Fig.12 voice jitter (sec) [modulation variation]

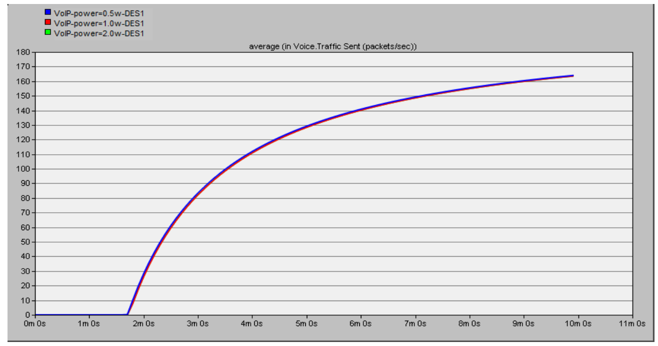

Figure 13. voice traffic sent (packet/sec) [power variation]

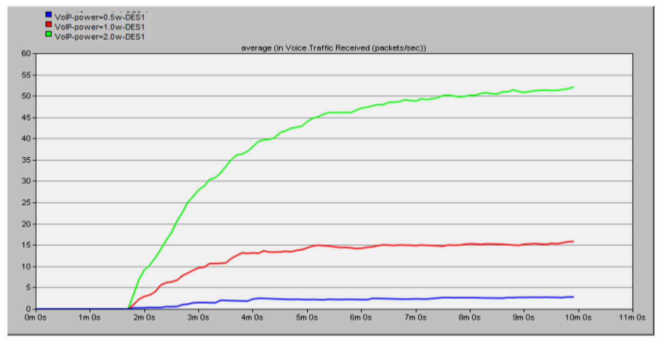

Figure 14. voice traffic received (packet/sec) [power variation]

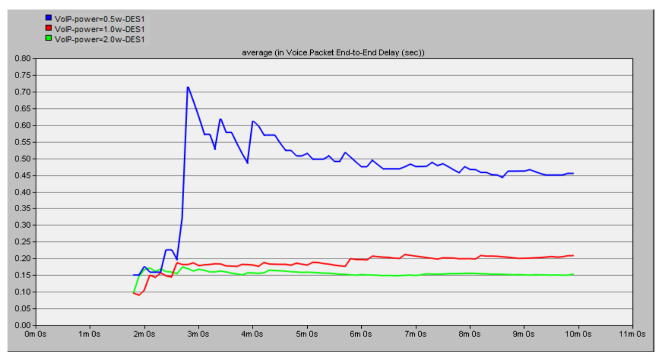

Figure 15. voice packet End-to-End delay (sec)[power variation]

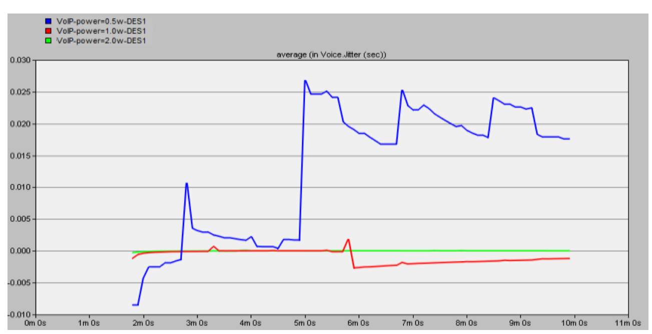

Figure 16. voice jitter (sec) [power variation]

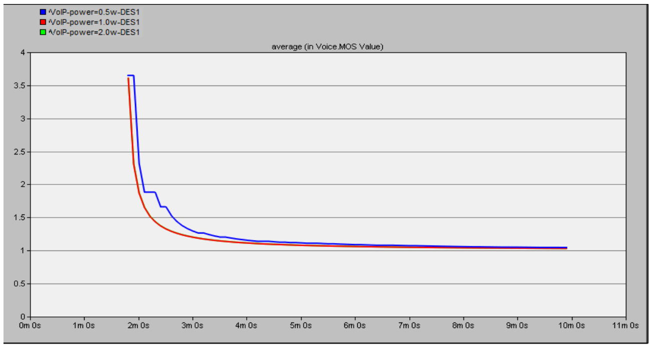

Figure 17. voice MOS value [power variation]

4. Conclusions

A new wireless access technologies are being developed, WiMAX is emerging as one of the promising broadband technologies that can support a variety of real-time services. Since the extension of VoIP calls over wireless networks is inevitable, we study the feasibility of supporting VoIP over WiMAX . This paper presents our progress to VoIP in evaluating the performance of IEEE 802.16. We have used the WiMAX Connection statistics (e.g. load, throughput) , and VoIP connection statistics (e.g. jitter , MOS value, End-to-End delay). The performance of the VoIP based on WiMAX, was evaluated and assessed at different: (i)distance ; (ii) modulation schemes; and (iii) power; As a result of the comparative study, it was found that : (i) In the three first scenario, When distance is 50 km (in range of WiMAX) it has the best performance and the least delay value compared with 100km and 200 km (out of range of WiMAX); (ii) In the four scenarios following We note that the two scenario who used 16-QAM and 64-QAM modulation have high performance ,but the two other scenario who use adaptive and QPSK have poor performance; and (iii)In the last three scenario, the performance depend on value of power; when power is lage , the performance is high and when power is low.

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