International Journal of Engineering Sciences Paradigms and Researches (IJESPR)
(Vol. 33, Issue 01) and (Publishing Month: August 2016)
(An Indexed, Referred and Impact Factor Journal)
ISSN: 2319-6564
www.ijesonline.com

The Effect of Voice & Video Conferencing in QoS over WiMax

Abdelmuiz Abdelkhalig Ahmed Bakheet[1] and Dr. Hala Eldaw Idris[2]

[1,2]Department of Communication Engineering, Alneelain University, Khartoum, Sudan
[1]abdelmuiz3@hotmail.com and [2]haldaw@hotmail.com

Publishing Date: August 30, 2016

Abstract

No doubt the progress in wireless networks indicate that the real time application such as Voice over Internet Protocol is great technology which enables the transport voice and data over Internet Protocol based networks. VOIP has become the actual alternative solution of PSTN networks. VOIP applicable to transport voice data in the form of digital Internet Protocol packets over the TCP/IP based Internet. In the other hand, the increasing of users demand for IEEE 802.16 standard based WiMAX networks is growing up now days. This research paper studies the performance of voice and video conferencing over WiMAX networks. The effect of voice and video conferencing in (QoS) over WiMAX has been investigated in detail. Through two simulation experiments using Opnet Modeler.

Keywords: *Wimax, QoS, voice, Video conferencing, Opnet Modeler.*

I. Introduction

WiMAX [1] WiMAX supports fixed and mobile Internet access. It can be connected with an Internet Protocol (IP) based core network, which is chosen by operators that serve as Internet Service Providers (ISPs). 802.16e uses Scalable Orthogonal FrequencyDivision Multiple Access (SOFDMA) rather than Orthogonal Frequency-Division Multiplexing (OFDM). It employs two multiple duplexing schemes: Time Division Duplexing (TDD) and Frequency Division Duplexing (FDD). WiMAX base station uses T1 (1.544 Mbps), which may provide bandwidth to hundreds of Internet subscribers with frequency band frame 10 GHz to 66 GHz [2].

There is huge demand of Quality of Service (QoS) In Communication Networks as general especially in wireless networks and real-time application sush as voice and video.

Quality of Service (QoS) refers to the capability of a network to provide better service to selected network traffic over various technologies, including Frame Relay, Asynchronous Transfer Mode (ATM), Ethernet and 802.1 networks, SONET, and IP-routed networks that may use any or all of these technologies. The primary goal of (QoS) is to provide priority including dedicated bandwidth, controlled jitter and latency (required by some real-time and interactive traffic), and improved loss characteristics. Also important is making sure that providing priority for one or more flows does not make other flows fail. **(QoS)** technologies provide the elemental building blocks that will be used for future business applications in campus, WAN, and service provider networks.[3]

WiMAX, the Worldwide Interoperability for Microwave Access, is a telecommunications technology that provides for the wireless transmission of data in a variety of ways, ranging from point-to-point links to full mobile cellular-type access. The WiMAX forum describes WiMAX as a standards-based technology enabling the delivery of last mile wireless broadband access as an alternative to cable and Digital Subscriber Line (DSL). WiMAX network operators face a big challenge to enable interoperability between vendors which brings lower costs, greater flexibility and freedom. So it is important for network operators to understand the methods of establishing interoperability and how different products, solutions and applications from different vendors can coexist in the same WiMAX network.[6]

II. The Needs of WiMAX Networks

WiMAX can satisfy a variety of access needs. Potential applications include:

• Extending broadband capabilities to bring them closer to subscribers, filling gaps in cable, DSL and T1 services, WiFi and cellular backhaul, providing last-100 meter access from fibre to the curb and giving service providers another cost-effective option for supporting broadband services.

• Supporting very high bandwidth solutions where large spectrum deployments (i.e. >10 MHz) are desired using existing infrastructure, keeping costs down while delivering the bandwidth needed to support a full range of high-value, multimedia services.

• Helping service providers meet many of the challenges they face owing to increasing consumer demand. WiMAX can help them in this regard without discarding their existing infrastructure investments because it has the ability to interoperate seamlessly across various network types.

• Providing wide area coverage and quality of service capabilities for applications ranging from real-time delaysensitive Voice-over-Internet Protocol (VoIP) to real-time streaming video and non-real-time downloads, ensuring that subscribers obtain the performance they expect for all types of communications.

• Being an IP-based wireless broadband technology, WiMAX can be integrated into both wide-area third-generation (3G) mobile and wireless and wireline networks, allowing it to become part of a seamless anytime, anywhere broadband access solution.

• Ultimately, serving as the next step in the evolution of 3G mobile phones, via a potential combination of WiMAX and Code Division Multiple access (CDMA) standards called Fourth Generation (4G).[4]

III. The Wi Max QoS Techniques

WiMAX provides Quality of Service at both physical and MAC layer using the following techniques:

PHY – The Physical Layer 802.16e uses Scalable OFDMA to carry data, supporting channel bandwidths of between 1.25 MHz and 20 MHz, with up to 2048 subcarriers. It supports adaptive modulation and coding, so that in conditions of good signal, a highly efficient 64 QAM coding scheme is used, whereas where the signal is poorer, a more robust BPSK coding mechanism is used. In intermediate conditions, 16 QAM and QPSK can also be employed. Other PHY features include integration of the latest technological innovations, such as ‘beam forming’ and Multiple Input Multiple Output (MIMO) and Hybrid Automatic Repeat Request (HARQ) for good error correction performance. OFDM belongs to a family of transmission schemes called multicarrier modulation, which is based on the idea of dividing a given high-bit-rate data stream into several parallel lower bit-rate streams and modulating each stream on separate carriers, often called subcarriers, or tones. Multicarrier modulation schemes eliminate or minimize InterSymbol Interference (ISI) by making the symbol time large enough so that the channel-induced delays (delay spread being a good measure of this in wireless channels) are an insignificant (typically, The CPS is the vital part of the 802.16 MAC that defines the medium access method. It provides functions related to network entry and initialization, duplexing, framing, channel access and QoS. The SeS provides privacy to the subscribers across the wireless network. It also provides strong protection against theft of service to the operators. It describes how secure communications are delivered, by using secure key exchange during authentication, and encryption using AES or DES (as the encryption mechanism) during data transfer. Further features of the MAC layer include power saving mechanisms (using Sleep Mode and Idle Mode) and handover mechanisms. A key feature of 802.16 is that it is a connection oriented technology. The SS cannot transmit data until it has been allocated a channel by the BS. This allows 802.16e to provide strong support for QoS.[2]

Physical Layer Frequency division duplex (FDD) - The application of frequency-division multiple access to separate outward and return signals. The uplink and downlink sub-bands are said to be separated by the "frequency offset". Frequency division duplex is much more efficient in the case of symmetric traffic. Time division duplex (TDD) - The application of time-division multiple access to separate outward and return signals. Time division duplex has a strong advantage in the case where the asymmetry of the uplink and downlink data speed is variable. Forward Error Correction (FEC) - The technique to allow the receiver to correct some errors without having to request a retransmission of data. Orthogonal frequency-division multiplexing (OFDM) - The frequencies and modulation of FDM are arranged to be orthogonal with each other which eliminates most of the interference between channels. Orthogonal Frequency Division Multiple Access (OFDMA) - It works by assigning a subset of subcarriers to individual users. MAC Layer - Service Types Unsolicited Grant Service (UGS) - Supports Real Time data streams, having fixed size packets issued at regular intervals e.g.. VoIP Real Time Polling Service (rTPS) - Supports Real Time data streams, having variable size packets issued at regular intervals e.g.. MPEG Video Non Real Time Polling Service (nrTPS) - Supports delay tolerant data with variable packet sizes, for which a minimum data rate is specified e.g.. ftp Best Effort (BE) - Supports data streams where no minimum service is required and packets are handled on a space-available basis.[5].

The IEEE802.16d WiMAX standard offers four categories for the prioritization of traffic: (1) Unsolicited Grant Service (UGS), (2) Real-Time Polling Service (rtPS), (3) Non-Real Time Polling Service (nrtPS), and (4) Best Effort (BE). Each of this service class is intended for specific application(s).

With Best Effort service, voice packets may have to be buffered until a transmission burst slot becomes available and thus leading to degradation in service.

Unsolicited Grant Service (UGS):

UGS is primarily intended for Constant-Bit-Rate (CBR) services such as VoIP, which means that achieving low latency and low jitter is very important. At the same time, low percentage of packet drops is possible. UGS flows are configured to send fixed size packets at recurring intervals with as little latency and jitter as possible.

UGS has the following set of features:

UGS flows are buffered separately from each other and from flows in service classes such as nrtPS and BE.

UGS service flows are given strictly higher priority versus nrtPS and BE service flows, which implies that the system serves nrtPS and BE packets only after it has finished transmitting all outstanding UGS packets.

In the upstream, the system uses UGS to bypass the normal request-grant mechanism for upstream traffic by allowing the BS to give automatic grants to a UGS flow. Also, over-the-air latency in a WiMAX network is small (5-40 ms) relative to the latency on an IP backbone (100ms), which inherently ensures minimal latency.

Real-Time Polling Service (rtPS) The Real-Time Polling Service (rtPS) on the other hand is designed to support real-time service flows that generate variable size data packets on a periodic basis, such as MPEG video. The service offers real-time, periodic, unicast request opportunities, which meet the flow's real-time needs and allow the Subscriber Station (SS) to specify the size of the desired grant. A major drawback to using this QoS approach is the impact on the overall sector throughput. Polling overhead can reach up to 60% when using 3.5MHz channel.

This service requires more request overhead than UGS, but supports variable grant sizes for optimum data transport efficiency. Unlike UGS, the polling overhead exists even when the flows are idle, and for as long as they are active.

Non-Real-Time Polling Service (nrtPS) This service class is intended to support non-real-time service flows that require variable size data packets, and a minimum data rate, such as FTP. This is accomplished by offering unicast polls on a regular basis, which ensures that the service flow receives requests even during network congestion.

Best Effort (BE) The BE service is intended to support data streams that don't require minimum guaranteed rate, and could be handled on best available basis. Unicast polling requests are not guaranteed in this case, requiring contention requests to be used. BE packets may therefore take a long time to transmit during network congestions.

Scheduling A scheduler is present in both the Base Station (BS) and Subscriber Station (SS). The BS schedule controls all system parameters. It is the role of the BS to determine the burst profile and transmission periods for each connection. The choice of the coding and modulation parameters are decisions that are taken by the BS scheduler based on the quality of the link and the network load and demand.

The BS scheduler continuously monitors the received CINR values of each link, and determines the bandwidth requirements for each station taking into consideration the service class for each connection and the quantity of traffic required.

The role of the SS scheduler is to classify all of the incoming packets into the SS different connections. Table 1 lists the scheduling service types that can be used for some of the standard applications.[3]

IV. Simulation Setup

To investigate deeply in the Network performance in order to evaluate the effect of voice and video conferencing using WiMAX access network, we designed and simulated close-to-real-time network scenarios to examine the of voice and video conferencing in real-time applications deployed over IEEE 802.16 standard based WiMAX access technology using Opnet Modeler 14.5 network simulator.

Fig. 1 shows the WiMAX network topology designed and used for first scenario simulation. The WiMAX access network model is made up of two WiMAX Base Station (BS) and ten (4) mobile nodes and an Internet Protocol (IP) backbone.

Fig. 2 shows the WiMAX network topology designed and used for second scenario simulation. The WiMAX access network model is made up of three WiMAX Base Station (BS) and fifteen (15) mobile nodes and an Internet Protocol (IP) backbone.

The configuration of both scenarios of the BS and SS performance parameters are depicted in Figure 3 and Figure 4 below. We used a server backbone with one voice and video server. In this simulation setup, we performed the following experiments: Experiment 1: here we used scenario 1 simulation to study the effect of numbers of Mobile Nodes in QoS of Voice and video conferencing over WiMAX networks. Experiment 2: here we used scenario2 simulation to study the Voice and video conferencing performances over WiMAX by increasing numbers Mobile Nodes.

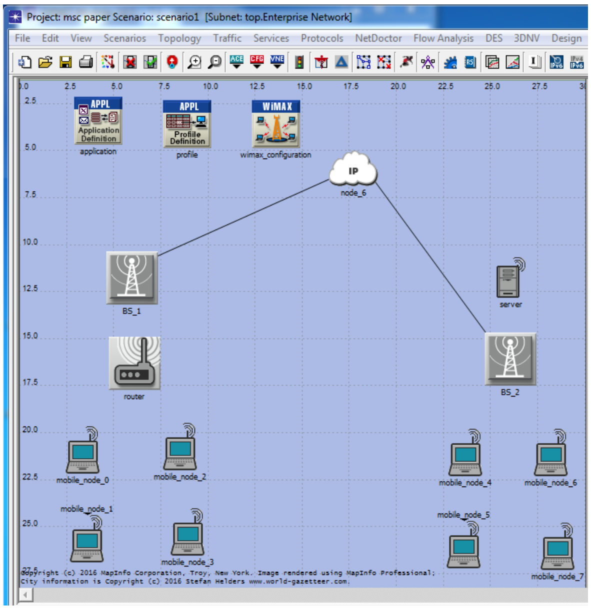

Figure 1: First Scenario Wimax Network Topology

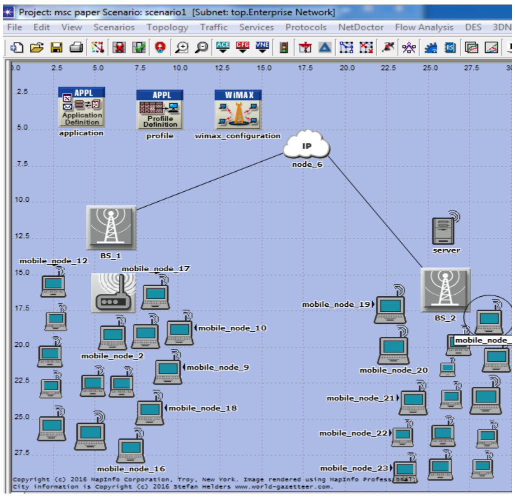

Figure 2: Second Scenario Wimax Network Topology

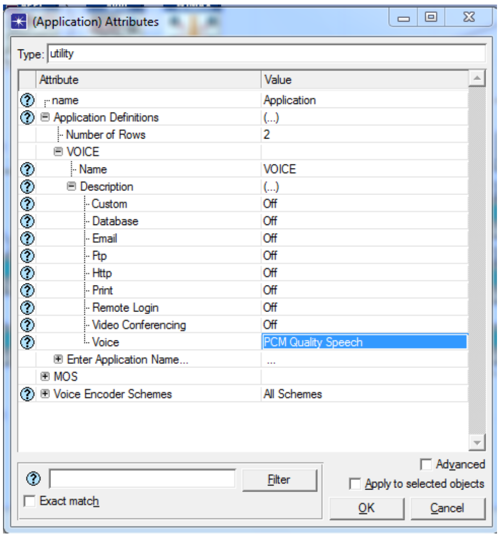

Figure 3: Voice Application Configuration

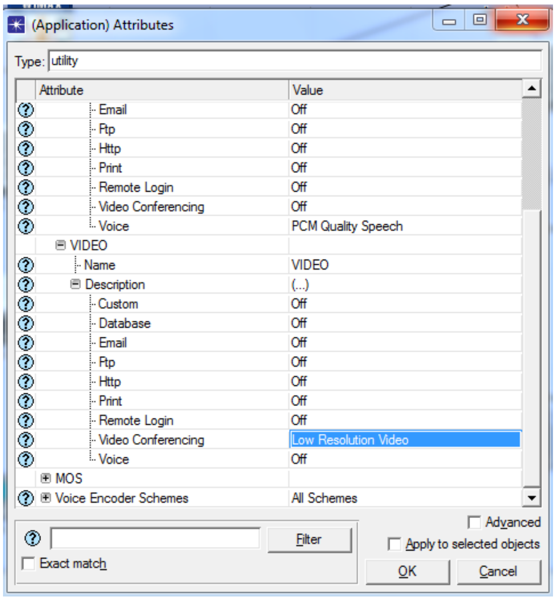

Figure 4: Video Conferencing Application Configuration

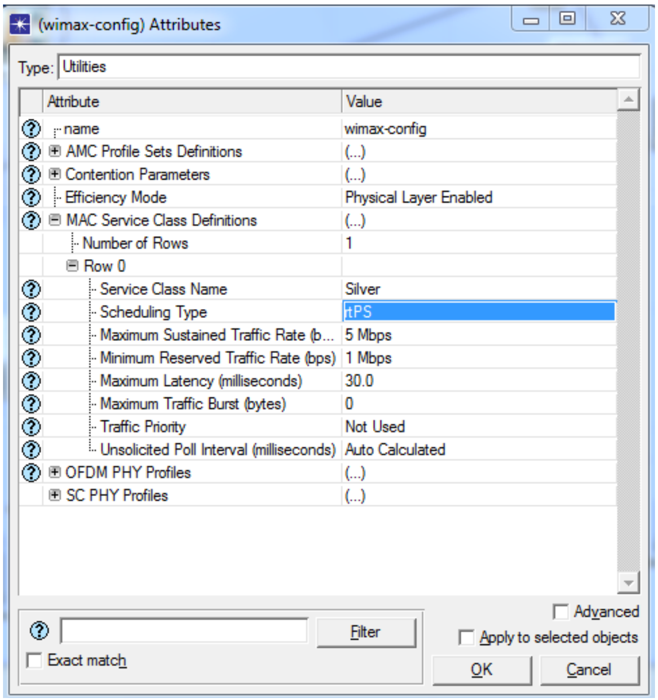

Figure 5: WiMax Configuration

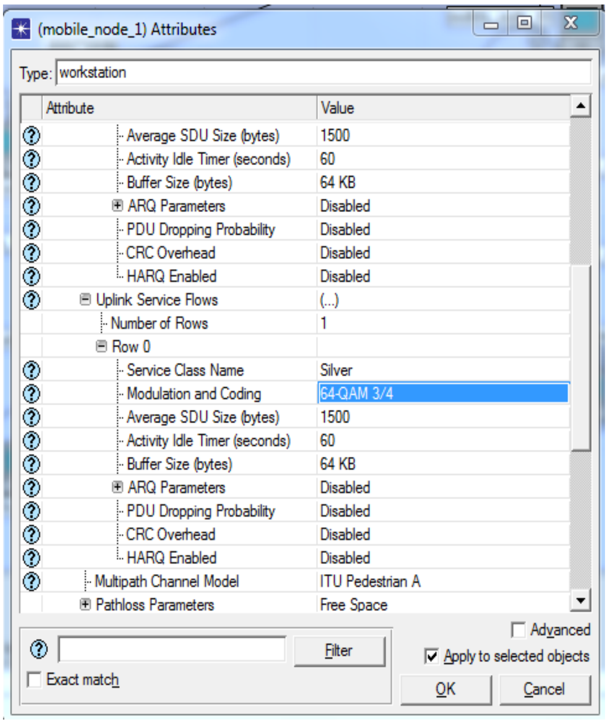

Figure 6: Mobile Nodes Configuration

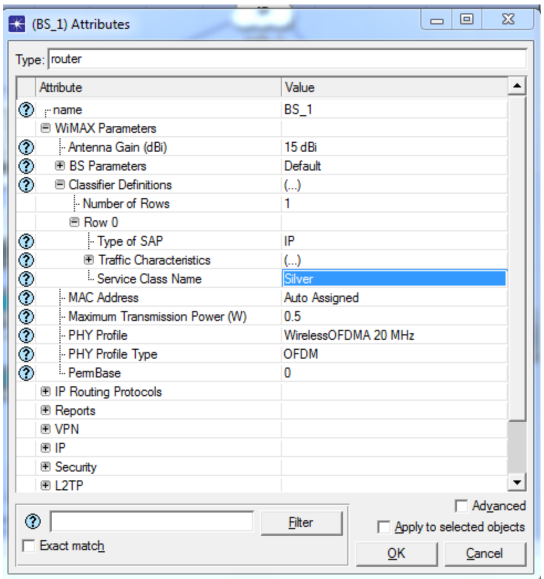

Figure 7: Base Stations Configuration

V. Simulation Result

The figures bellow shows the simulation results for three voice codec's.

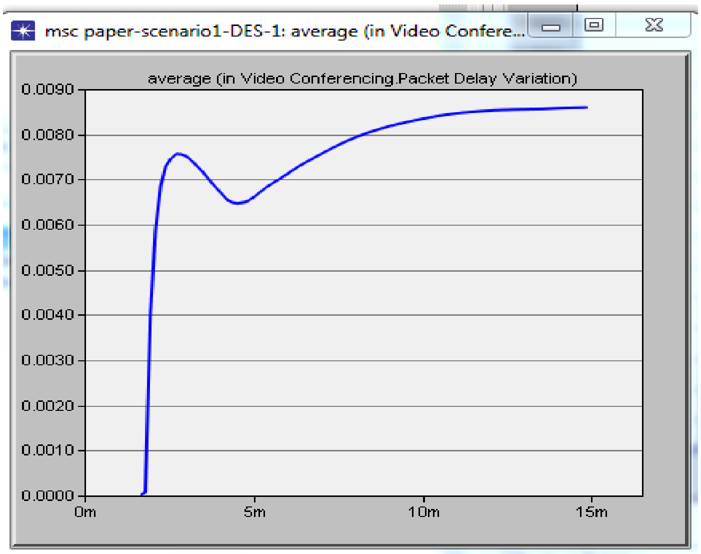

Figure 8 (a): Video Conferencing Packet Delay Variation (First scenario)

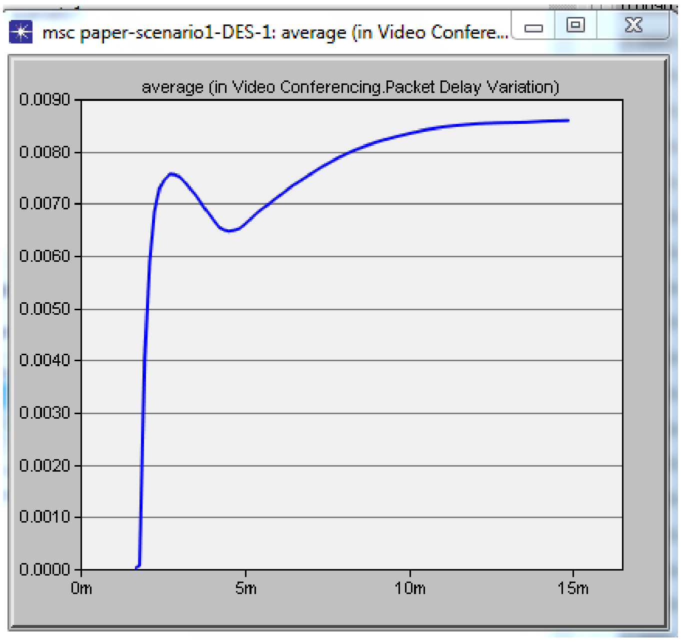

Figure 8 (b): Video Conferencing Packet Delay Variation (second scenario)

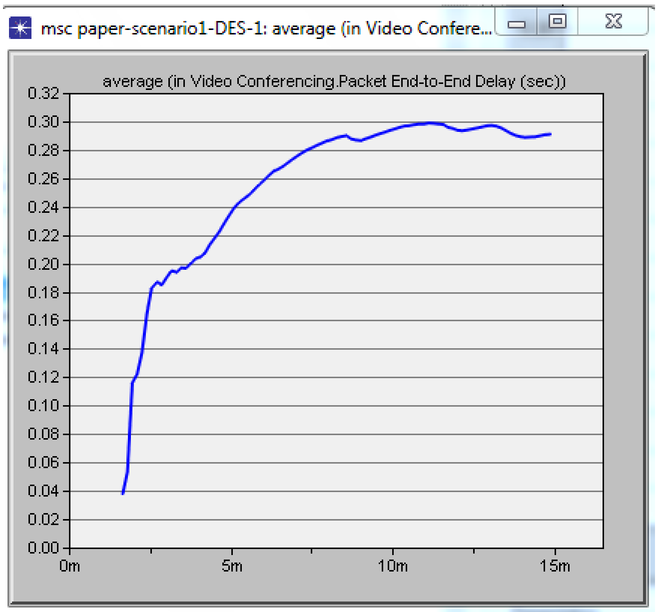

Figure 9 (a): Video Conferencing Packet End –To-End Delay (first scenario)

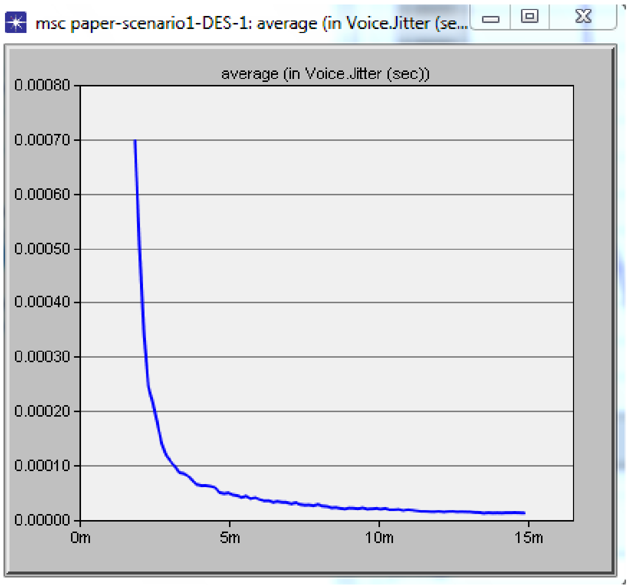

Figure 10(a): Voice Jitter

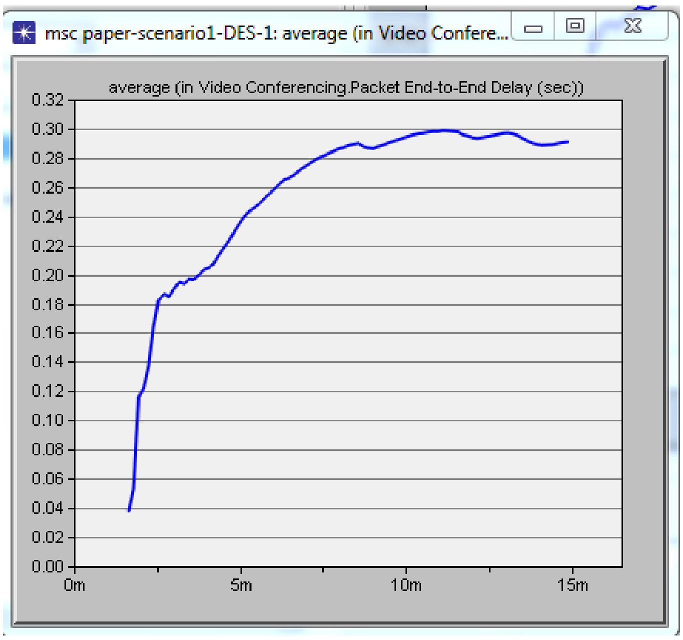

Figure 9(b): Video Conferencing Packet End–To-End Delay (second scenario)

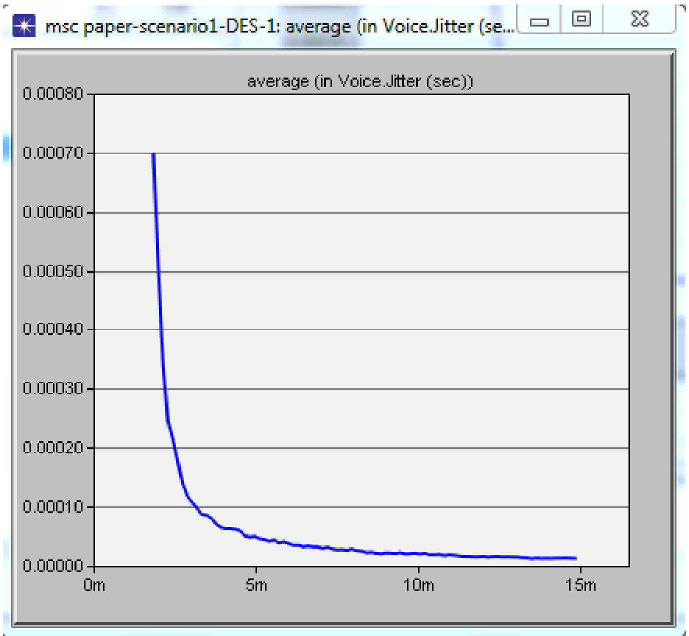

Figure 10 (b): Voice Jitter

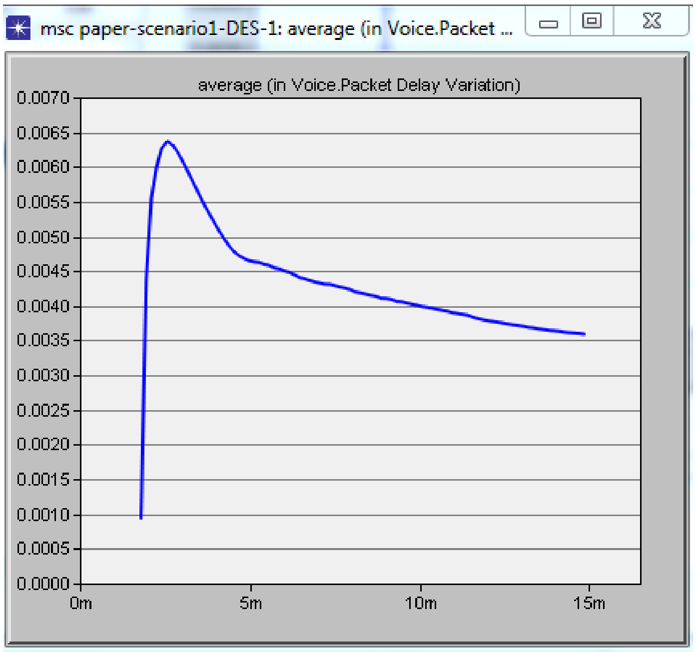

Figure 11: Voice Packet Delay Variation

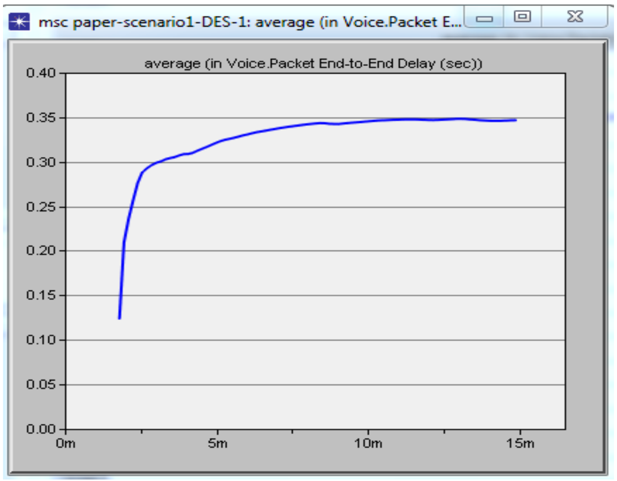

Figure 12: Voice Packet End –To-End Delay

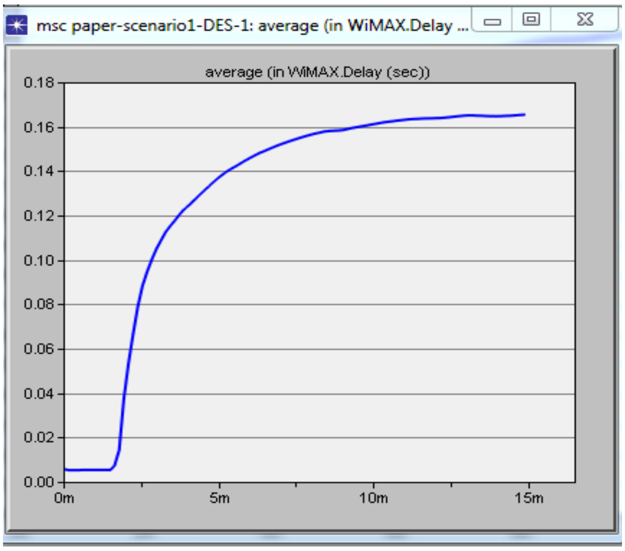

Figure 13: Wimax Delay

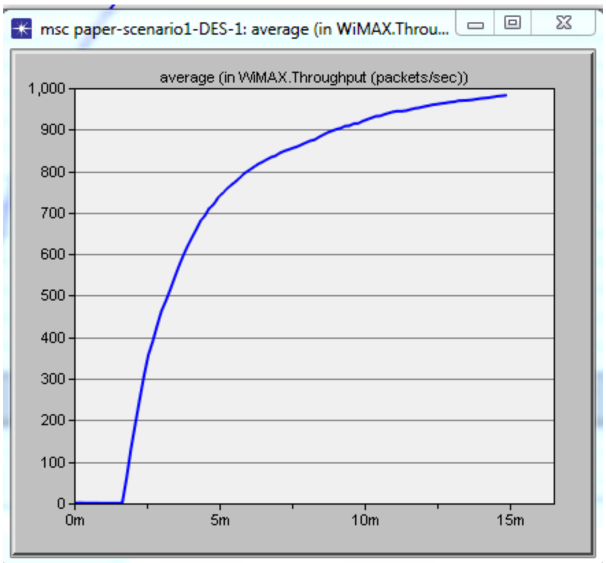

Figure 14: Wi Max Throughput

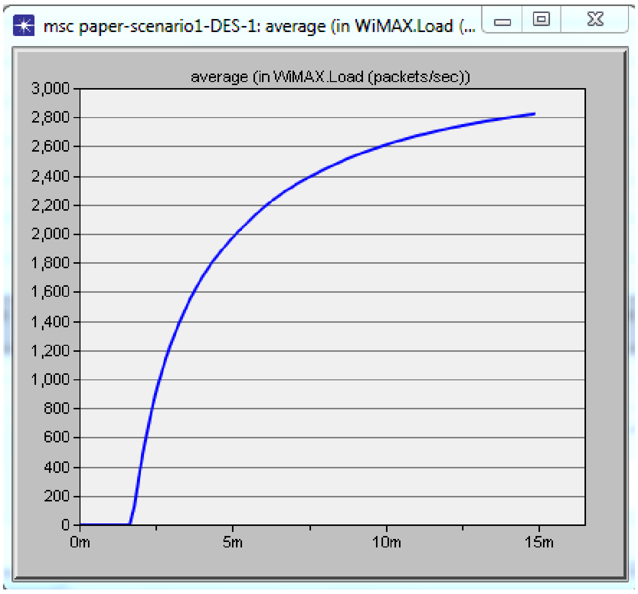

Figure 15: WiMax Load

VI. Conclusions

From above experimental study, evaluated the Voice and video conferencing over wimax using OPNET modeler 14.5 simulation tools in order to study the effect of Voice and video conferencing (QoS) over wimax networks.

We found that real time multimedia application such as voice and Video Conferencing is having higher load, maximum throughput and minimum delay as compared to non-real time.

Although of increasing the mobile node has been made in Second scenario, the video conferencing as well as voice (QoS) parameter such as Packet Delay Variation, Packet End –To-End Delay is stable as it's in the first scenario, and there is no change of the effect in performance metric.

References

[1] B. G. Lee and S. Choi, Broadband Wireless Access and Local Networks: Mobile WiMAX and WiFi. Boston, London: Artech House, 2007.

[2] W. Hrudey and Lj. Trajkovic, "Mobile WiMAX MAC and PHY layer optimization for IPTV," Mathematical and Computer Modelling, Elsevier, vol. 53, pp. 2119–2135, Mar. 2011.

[3] Internetworking Technologies Handbook 1-58705-001-3,Chapter 49 Quality of Service - Networking Link Efficiency Mechanisms, http://docwiki.cisco.com/wiki/Internetworking_Technology_Handbook.

[4] Wimax Security And Quality Of Service An End-To-End Perspective. Edited by Seok-Yee Tang Think Wireless Tech Pte. Ltd., Singapore Peter Muller ¨ IBM Zurich Research Laboratory, Switzerland Hamid R. Sharif University of Nebraska-Lincoln, USA.

[5] VoIP/Multimedia over WiMAX file:///F:/www/cse574-06/ftp/wimax_voip/index.htm

[6] IEEE Standard "802.16-2005. Part 16: Air interface for fixed and mobile broadband wireless access systems", December 2005.