Performance Evaluation of Next Generation WIMAX Networks using different Service Protocols

Ritesh Sadiwala

Professor, Department of Electronics and Communication Engineering, RKDF University, Bhopal, INDIA
riteshsadiwala@gmail.com

* Corresponding Author: Ritesh Sadiwala

Abstract

In telecommunication, traditionally telephone networks uses basic concept called circuit switching for doing communication between sender and receiver such as Public Switched Telephone Network (PSTN). Nowadays this type of switching is being replaced by packet switching where the transmission of data is done in packets and networks based on this are referred to as connectionless networks. WIMAX Technology is an upcoming wireless technology that provides us high speed mobile data and telecommunication services. This technology is used to replace the traditional telephone networks i.e., Public Switched Telephone Network (PSTN). In this paper performance of such networks which uses VOIP technology in WIMAX Networks is investigated using different service protocols and for which OPNET 14.5 Simulator is chosen as the platform. OPNET Simulator provides simulated real-life environment, in order to study the performance of the next generation networks. The parameters used to evaluate the performance of the networks are jitter and end to end delay. Finally based on the simulations best VOIP codec is also found out.

Keywords: VOIP Codecs, OPNET, QOS, VOIP, PSTN.

I. Introduction

A Telecommunication protocol such as Worldwide Interoperability for Microwave Access (WiMAX) provides wireless Internet access for fixed and mobile nodes. It is based on IEEE 802.16 wireless Metropolitan Area Network standard and has been deployed extensively in recent years to solve the problems associated with point-to-multipoint broadband outdoor wireless networks [1]. Many authors have worked on various QoS parameters using different service classes in WiMAX. A study was conducted on various quality parameters impacting the WiMAX service performance of a WiMAX network. The study suggests that these critical parameters of QoS are required to increase the performance of a WiMAX network. . The main objective of the work presented in this paper is to evaluate the performance of NGWIMAX Networks over wireless local area network. The greatest technical problem in supporting multimedia services over IP is that real-time traffic must reach its destination within a preset time interval (delay) and with some tolerance of the delay variation (jitter). This is difficult because the original UDP/IP operates on a best-effort basis and permits dropping of packets on the way to a destination [2]. The simulation model was done using OPNET Modeler [3] [4]. OPNET has gained considerable popularity in academia as it is being offered free of charge to academic institutions. That has given OPNET an edge over DES NS2 in both market place and academia [5]. Thus, Step by step, increasing the number of calls, investigations have been done in terms of important Quality of Service parameters like jitter, packet end to end delay and throughput and results are found out.

The rest of this paper is organized as follows. Section II gives a brief description of WiMAX, VoIP codecs and its related technology. Section III provides the model design and configuration networks. Results are given in Section IV. Finally, section V concludes the paper.

II. Description of WIMAX, VOIP CODECS and its Technology

WiMAX (Worldwide Interoperability for Microwave Access) is a wireless communications standard intended to provide 30 to 40 Mps data rates, providing up to 1 Gbit/s for fixed stations. It is based on IEEE 802.16e-2005 standard [6], which added some improvement to 802.16-2004 standard [5], taking into account mobility. Several new techniques (OFDMA turbo code, FFT, EAP, MIMO ...) are used for a better support for Quality of Service. It can be used in both point to point (P2P) and the typical WAN type configurations. WiMAX supports different multimedia applications as VoIP, voice conference and online gaming. The IEEE 802.16 technology (WiMAX) is an improved alternative to 3G or wireless LAN networks for providing ease of access, low cost and large coverage area.

A. Quality of Service in WiMAX Networks

Quality of Service (QoS) [12] is the ability to communicate in good conditions a type of traffic, in terms of availability, throughput, transmission delay, jitter, packet loss, etc. It has become an important factor to support variety of applications that use network resources. These applications include multimedia services, voice over IP…etc. The term Quality of service refers to the probability of the telecommunication network meeting a given traffic contract, or the probability of a succeeding packet in the transition between two points in the network. The main aim of a good network is to deliver priority including better throughput, controlled jitter and enhanced loss characteristics. Quality of Service is measured by various parameters like Jitter, delay, packet loss, load and throughput.

1) Jitter: Jitter is the variation of delay between the two consecutive packets from the T-stream traffic in the output queue [14]. For non-real-time data communications, delayed packets can be stored for an indefinite amount of time at local buffers. For real-time applications like VoIP service, delayed packets may become useless after a pre specified amount of time. The delay jitter buffers hold these ''precocious'' and delayed packets in an attempt to neutralize the effects of the packet inter arrival jitter. This helps to maintain real-time communication over packet-switched networks.

2) End-to-End Delay: End-to-end delay is the time interval in which a packet can travel from one node to another node. VoIP is very sensitive to delay; thus, it must be controlled and managed. Further, it is inefficient to wait for all the packets arriving in an organized order. So, some packets may be dropped if they don't arrive in time and this can cause short periods of silence in the audio stream and can cause bad VoIP quality. Ideally, the delay constraint for VoIP packets is not above 80ms.

3) Delay: Transmission time includes delay due to codec processing as well as propagation delay. ITU-T Recommendation G.114 recommends the following one-way transmission time limits for connections with adequately controlled echo:

a) 0 to 150 ms: acceptable for most user applications.

b) 150 to 400 ms: acceptable for international connections.

c) 400ms: unacceptable for general network planning purposes.

4) Packet Loss: Packet loss is inevitable in IP networks and occurs for various reasons. For example, it occurs when routers or switch work beyond capacity or queue buffers over flow. VoIP network packet loss, above some threshold rate, introduces audio distortions that cause voice quality to be decreased as the rate of packet loss increases [16].

5) VoIP Codecs: RTP and UDP protocols are the logical choice to carry voice when TCP protocol favors reliability over timeliness. Voice signals are digitally encoded. This means that each voice signal is converted from digital to analog and back. The analog signal is firstly sampled based on a sampling rate of 8 KHz, 8 bits per sample is the most frequently cases. Next, the output is encoded according to many factors: the compression rate and the framing time or the frames length. Finally, one or more of these frames are encapsulated into an RTP/UDP/IP packet for transmission over the network. All these practices are accomplished by one of various audio codecs, each of which vary in the sound quality, the bandwidth required, the computational requirements, encoding algorithm and coding delay [8, 9, 14] .The codecs along with their description used in the network simulation is given below:

a) G.711 is the default standard for all vendors, very low processor requirements. This standard digitizes voice into 64 Kbps and does not compress the voice, It performs best in local networks where we have lots of available bandwidth.

b) G.729 is supported by many vendors for compressed voice operating at 8 Kbps. Excellent bandwidth utilization and Error tolerant with quality just below that of G.711.

c) G.723.1 was once the recommended compression standard. It operates at 6.3 and 5.3 Kbps. High compression with high quality audio. Although this standard reduces bandwidth consumption, voice is much poorer as compared with G.729 and is not very popular for VoIP.

III. Network Model Design and its Configuration

The network model presents a campus wireless LAN. This wireless LAN consists of six outdoor Access Points (APs) connected to one Gbps switch as shown in Fig. 1. The general network topology showed in Fig. 2. A number of wireless workstations (WSs) are connected to each AP. WSs are suggested to be at the fixed place, then step by step, increasing the number of workstations (WSs), four scenarios were implemented, for 4,6,8 and 9 WSs at each AP.

For each type the values of QOS parameters were taken. To investigate the performance of VoIP with TCP on IEEE 802.11g simulations were performed using OPNET Modeler 14.5. OPNET Modeler is a powerful communication system discrete event simulator (DES) developed by OPNET Technologies. OPNET Modeler 14.5 assists with the design and testing of communication protocols and networks, by simulating network performance for wired and/or wireless environments [4]. The simulation lasts for variable time from 20- 300 seconds. The attributes that shown in Table I are defined for the network. Best effort type of service was used and G.729A (silence) codec as recommended by [12].

TABLE I. VOICE ATTRIBUTES AND THEIR VALUES

Attribute	**Value**
Silence length (seconds)	default
Talk spurt length(seconds)	default
Symbolic destination Name	Voice destination
Encoder scheme	G.729A (Silence)
Voice frames per packet	1
Type of service	Best effort (0)
RSVP parameters	None
Traffic mix	All discrete
Signaling	None
Compression delay(seconds)	0.02
Decompression delay(seconds)	0.02

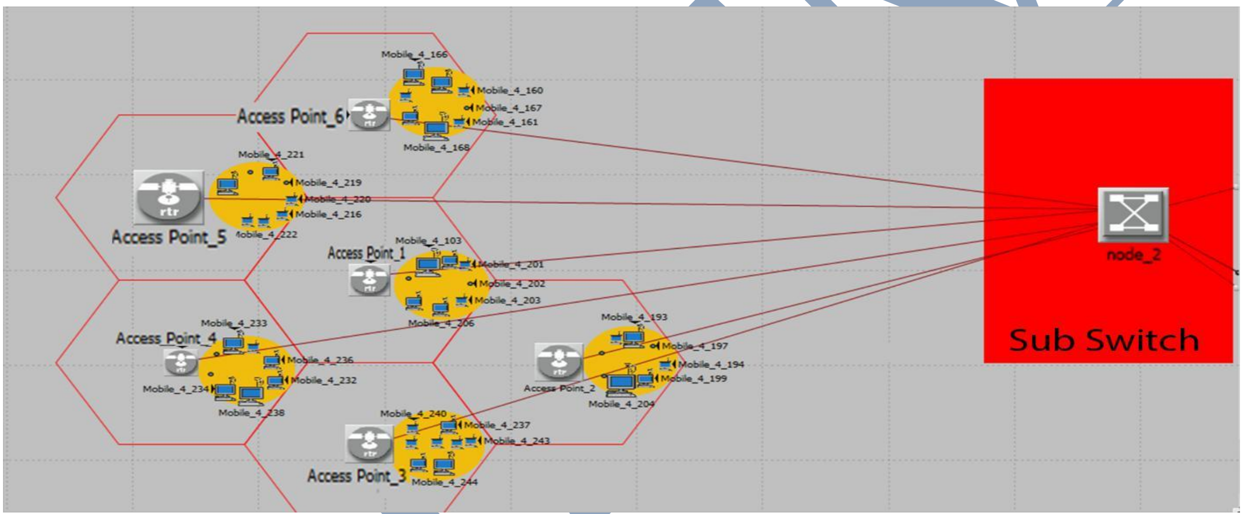

Figure 1. The network model

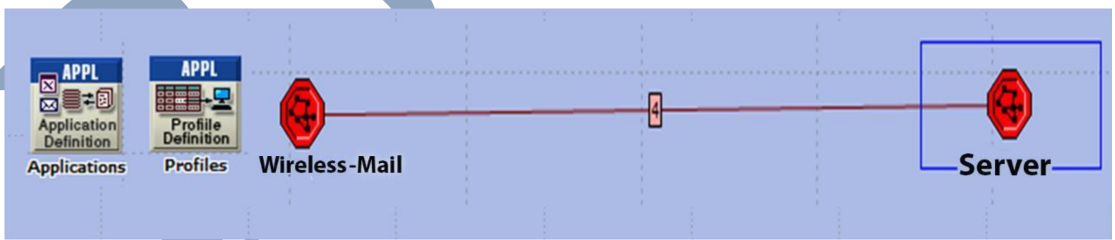

Figure 2. The general network topology

IV. RESULTS AND DISCUSSIONS

An important parameter in determining the QoS for VoIP is its timing operations. So, the jitter and the delay will be studied initially. Fig. 3 shows the average packet end-to-end delay for the four scenarios i.e. the scenarios with 4 WSs, 6 WSs, 8 WSs and 9 WSs. The values of end-to-end delay, for all scenarios, are constant because all the scenarios have the same bit rate and the network is not congested. Wireless LAN delay represents the end-to-end delay of all the packets received by the wireless LAN MACs of all WLAN nodes in the network and forwarded to the higher layer. This delay includes medium access delay at the source MAC, reception of all the fragments individually, and transfer of the frames through AP if access point functionality is enabled [17] [18].Fig. 4 represents the wireless LAN delay. The wireless LAN delay values for the scenarios with 4 WSs, 6 WSs and 8 WSs are very small. They are accepted and even good. But the wireless LAN delay for 9 WSs and 10 WSs reach 270 ms and 1.2 sec respectively i.e. not accepted. Table II presents the values of the wireless LAN delay for the scenarios with 4, 6, 8 and 9 WSs.

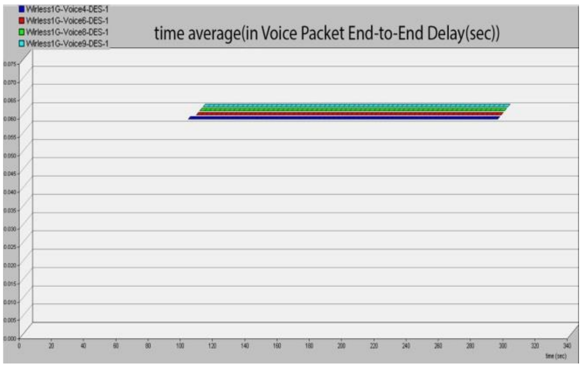

Figure 3. End-to-end delay

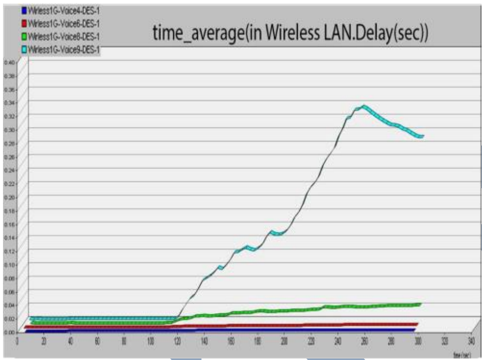

Figure 4. Wireless LAN delay

TABLE II. WLAN DELAY FOR THE FOUR SCENARIOS

Scenario	Time
4 WSs	2 ms
6 WSs	4 ms
8 WSs	26 ms
9 WSs	270 ms

The load & throughput test is concerned with the receipt of the payload data without considering the overhead of network against the load. Load represents the total load (in bits/sec) submitted to wireless LAN layers by all other higher layers in all WLAN nodes of the network. This statistic does not include the bits of the higher-layer packets that are dropped by WLAN MACs upon arrival and not considered for transmission because of, insufficient space left in the higher layer packet buffer of the MAC [17] [19]. The load can be observed in Fig. 5.

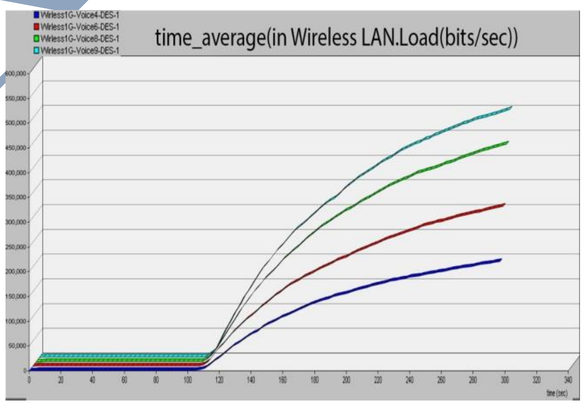

Figure 5. Wireless LAN load

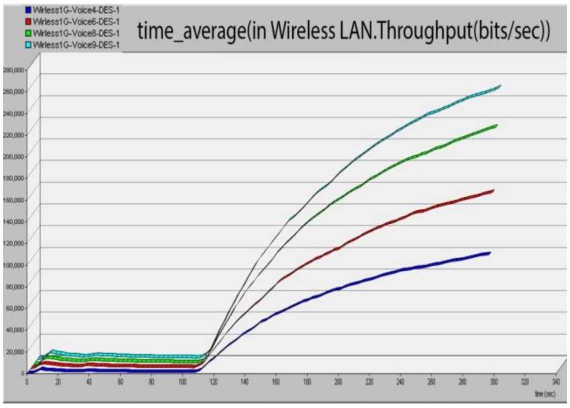

Figure 6. Wireless LAN throughput

Throughput represents the total number of bits (in bits/sec) forwarded from wireless LAN layers to higher layers in all WLAN nodes of the network [17] [19]. Throughput is represented in Fig. 6. Increasing the number of the WSs connected to each AP, the Load and the throughput are also increased. But there is no benefit from the increase in throughput for more than 8 WSs, because of the increase in the wireless LAN delay, which exceeds the accepted value for VoIP.

From the above figures, it can be concluded that the capacity of the campus wireless LAN in the studied case is 8 WSs using VoIP service for each Access Point. In other words, the capacity of the whole wireless campus LAN is 48 VoIP clients at the same time.

V. CONCLUSIONS

In this paper, simulative investigations have been done for VoIP in WiFi campus network using OPNET Modeler. Workstations (WSs) are increased one by one and investigations have been done in terms of important Quality of Service parameters like jitter, end-to-end delay and wireless LAN load and wireless LAN throughput. Based on the simulation results, the jitter values in all scenarios are neglected because the network is not congested. Wireless LAN delay is acceptable for the scenarios with 4, 6, and 8 WSs i.e. the capacity of the campus wireless LAN in the studied case is 8 WSs using VoIP service for each Access Point. The throughput of the wireless networks depends on network load. But there is no benefit from the increase in throughput for more than 8 WSs, because of the increase in the delay, which exceeds the accepted value for VoIP. As a future work, more types of traffic like Email, FTP and HTTP will be added to make the case more realistic.

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