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Comparison between IPv4 and IPv6 VoIP over WiMAX Delay

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ABSTRACT

This paper is to plan and design OPNET simulation model to study and analyze delay in VoIP over WiMAX using IPv4 and IPv6. The model was designed and implemented using OPNET simulation. It was found that IPv4 has better performance than IPv6 concerning mean delay time and jitter, while IPv6 has better performance than IPv4 concerning end-to-end delay time.

Keywords:- VoIP; WiMAX; IPv4; IPv4; H.323; SIP.

I. INTRODUCTION

Today voice over IP is one of the most dominant communication technology because of it's the easiest way to make a phone call through internet by sending a packet through packet switching instead of public switching telephone network (PSTN), it has a benefit over (PSTN) that voice data packet may take the best path while compared to (PSTN) which is a circuit switching require a dedicated path [1].

There are two major categories for transmitting voice over IP network called Internet technology and voice over IP (VOIP), when the voice data packets were transmitted over public network it's called internet technology, while if the voice data packet were transmitted over a managed network it's called VoIP, the primary difference between managed IP network and unmanaged IP network is the quality, however the differences are getting narrowed with the technology advancement [2].

Sometimes voice over internet protocol (VoIP) referred as internet telephone, IP telephone, voice over the internet [3].

VoIP components**:** End-user equipment: it is used to access the VoIP system to communicate to others endpoints. Network component: the network devices such as switches, routers Gatekeeper: the gatekeeper has three functions, they are signalling gateway, media gateway and media control [4].

Correspondingly, making a call over the internet, signalling protocol plays a major role since it improves communication between network components. The key role of signalling are session establishment: when decides if to accept, reject or redirect the call, user location (the receiver (Rx)) location must be found, call participant management: allowing to join or leave an existing session and session registration. [1]

II. VOIP PROTOCOL

The most VoIP common protocols are H.323 and SIP (Session Initiation Protocol), H.323 Specified by International Telecommunication Union (ITU) consists of protocols such as H225 for registration, admission state and call signalling, H245 control signalling [5], H235 for security. [4], H248"control protocol for media gateway across a converged internetwork" [3]. Real-Time transport protocol (RTP) used for applications that have real-time characteristics[6], enabling data streams transported to multiple destinations through IP multicast, it uses two RTP session establishments, one for audio and the second for video, each one has an identifier (SSRC)[3], real-time transport control protocol (RTCP) which works with RTP static information such as data delivery over large multicast network, one way delay, packet is lost and jitter occurs [3] in ongoing session [7].

H.323 Components are Terminal: which represents the end user equipment. Gateway: which is used to communicate with other different networks. Gatekeeper: which is used to provide services example (authentication, call routing). [4]

Table 1: H.323 codec.

Codec H.323	G.711	G.723	G.728	G.722	G.729
Channel (Kbps)	64	5.3, 6.3	16	48,56, 64	8

Session Initiation Protocol: A signalling protocol specified by Internet Engineering Task Force (IETF), operate on the application layer, uses TCP allowing security using SSL/TLS and UDP to provide a fast connection. [4]

SIP main element: User Agent (UA): representing the end user, ether its Tx or Rx. Gateway: which allow SIP users to communicate with different VOIP network. Proxy server (PS): which route the SIP request in the SIP network. Register: which declare if the Rx is ready to receive a call or not and bind the users IP into SIP URL. Location Server (LS): which store all the binds and the User Agent location.[8]

Table 2 SIP codec

Codec	G.711	G.722	G.726	G.728	G.729
Channel Kbps	64	48,56, 64	26,16, 48,32	16	8

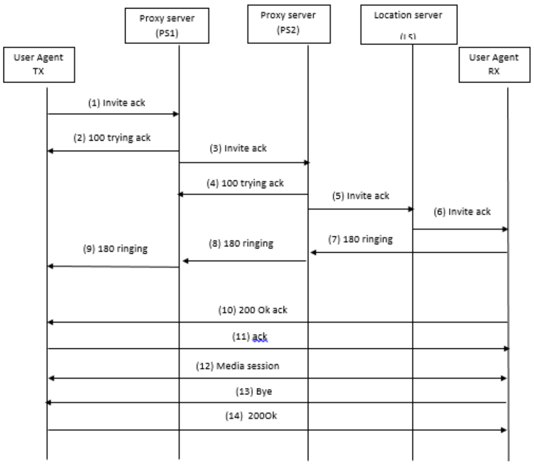

Figure (1) the SIP signalling

SIP signaling steps: The user agent (Tx) send an invite ack to proxy server (PS1) which PS1 replay by sending 100 trying to (Tx), PS1 send invite ack to PS2, PS2 replay by sending 100 trying and send to the location server to determine the user Agent (Rx) location when Rx is found the (PS2) send to Rx an invite ack, this is the point where Rx phone starts to ring, when Rx start ringing the User Agent (Rx) send 180 ringing ack to Proxy server (PS2) which send the 180 ringing to Proxy Server(PS1) which get to User Agent (Tx), when the User Agent (Rx) Pick-up and answer the phone call the User Agent send (Rx) send 200 Ok ack the same way the 180 ringing ack was sent, the User Agent (Tx) send ack replaying for receiving the 200 ok ack from User Agent (Rx), this is the point where RTP (pear o-pear) start working, the invite ack where sent between SIP element contain Session Description Protocol (SDP) which negotiate RPT parameter, when the User Agent hang-up the phone, the User Agent (UA) send Bye ack to the second User Agent

In addition, Ok ack sent replaying for the Bye ack.

While sending to PSTN, the session description protocol (SDP) is converted into ISUP to relay signals between switches and the RTP media stream are converted into TDM channel to transmit the voice signals.[2]

III. QUALITY OF SERVICE

QoS achieved by managing the delay. [4], VoIP is generally measured in term of delay, jitter and packet end-to-end delay.[9], Packet end-to-end (E2E) delay "is measured by calculating the delay from the speaker to the receiver including the compression and decompression"[10], Jitter "is variation in arrival time of consecutive packets"[11].

Table (3): ITU-T Precept for Voice Quality. .

Network parameter	Good	Accepted	Poor
Delay (ms)	0-150	150-300	More than 300
Jitter (ms)	0-20	20-50	More than 50

IV. WIMAX

Wireless communication networks become an essential part of modern lifestyle requirements, the demand for high-speed data rate and quality of service is being the leading factor in the evolution of technology like WiMAX.[12], WiMAX is 4^{th} technology IEEE 802.16 standard wireless communication, which can provide broadband up to 50Km for a fixed station and 10 miles (5-15km) for a mobile station.[13], high data rates, high number of users, convergence and availability, reliability and with preferable cost for the VoIP networks.

V. Mathematical model

Coder delay is the time for the digital signal processor to compress PCM (pulse code modulation) into sample [14]. Packetization delay is the time for data encapsulation into the packet

$$T_p = \frac{8\,P_s}{C_{BW}} \qquad (1)$$

Where: T_P : Packetization delay [ms]. P_S : Payload size [B]. C_{BW} :I codec bandwidth [Kb/s].[14]

Change in bandwidth is the information being added according to the technology protocol being used.

$$T_{BW} = \frac{H_L . C_{BW}}{P_s} + C_{bw} \qquad (2)$$

Where: T_{BW} : Total bandwidth [kb/s]. P_s : Payload size[b], H_L : Denotes header and tail length[b] , C_{BW} : Codec bandwidth [b] [14].

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Decompression delay

$$T_{DCD} = 0.1T_{CD} \qquad (3)$$

Where: T_{DCD} : Decompression delay [ms]. T_{CD} : Coder delay [ms]. [14]. Jitter known as variance of the RTP data packet inter-arrival time

$$D_{(i,j)} = (R_j - R_i) - (S_j - S_i) = (R_j - S_j) - (R_i - S_i) \qquad (4)$$

Where: $D_{(j,i)}$: Deference of relative transit time for the two packets. S_i : Timestamp for packet i. R_i : Arrival time for packet i.[15]. Serialization delay is a delay that depends on the transmission Bandwidth

$$T_{SER} = \frac{F_S}{L_S} \qquad (5)$$

Where: T_{SER} : Serialization delay[m]. F_S : Packet size[b]. L_S : Denotes line speed [kbit/s]. [14]

Propagation delay: Its transmission caused by physical media properties. Depacketization delay, in the receiver, when it receives the first block it can be decompressed and get zero delays but the other blocks have to wait in the buffer and the last of them gets the value of packetization delay, this value is the opposite to values of packetization delay [16]. The throughput of a connection or a link = total number of bit successfully transmitted during the period [t, t+T] divided by T.

VI. METHOD

The wireless deployment wizard was used to create three WiMAX cells, each one has five subscribers (WiMAX_wkstn_adv) in random positions in each cell, the cell radius is 1Km, the BTS used were (WiMAX_bs_router_adv) and one router was used (ethernet4_slip8_gtwy), in WiMAX config, the class service was gold and scheduling type was used is (UGS) with 384 as maximum reserved traffic rate and maximum sustained traffic rate.

Table 4 Trajectory parameter and fixed speed 50km/h

Wait time	0	10m,90s	10m,0s	10m,0s
	1	2	3	4

In the workstation, the service class was set into (gold), the traffic value was set into (interactive voice), and the uplink and downlink service class were set into (gold) and the duplicate scenario was made, the first for IPv4 and the second for IPv6.

For IPv4 subnet, The first scenario was used for IPv4, all devices in the network were given in IP, in BTS which the work station starts movement from, the BTS was given an IP (IP-IP routing parameter – interface information) the IF32 was given an address as class C and in BTS also in (mobile IP router parameter-mobile IPv4 parameter) the interface name was set as (IF32) which was (home agent), in the other BTS's, they were given an IP address as class C and in (mobile IP router parameter-mobile IPv4 parameter) the interface name was set as (IF32) which were a (foreign agent), the router was given a class C IP address, the router IP was declared for the four BTS's, (WiMAX parameter_ bs parameter_ ASN gateway address), OSPF was set as routing protocol, the workstation was given the trajectory, was given the same IPv4 address in the interface IF32 in its BTS

For IPv6 subnet, all interfaces (workstations) in the subnet were given an IPv6 (protocols-IPv6), all BTS's in the subnet were set (IP-mobile IP router parameters –mobile IPv6 parameters- number of rows (1) interface name IF24 as home agent), the OSPF was set as a routing protocol.

VII. RESULTS

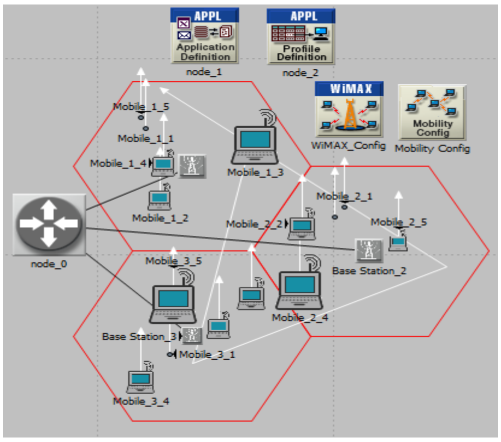

Figure 2: WiMAX network

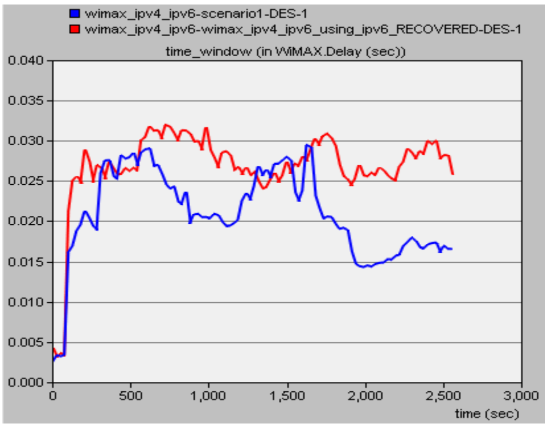

Figure 3: IPv4 vs. IPv4 (blue curve is IPv4, the red curve is IPv6) delay, the delay time mean of IPv4 =0.0207106209876 sec (20ms) and for IPv6 =0.0266530812287 (26.6ms)

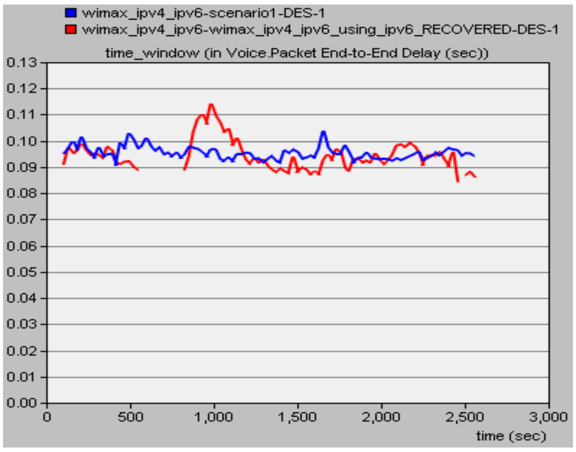

Figure 4: IPv4 vs. IPv6 (blue curve is IPv4, the red curve is IPv6) packet end-to-end delay, the delay mean for IPv4=0.0953489929312 (95.3ms) and IPv6=0.0943925674727(94.3ms)

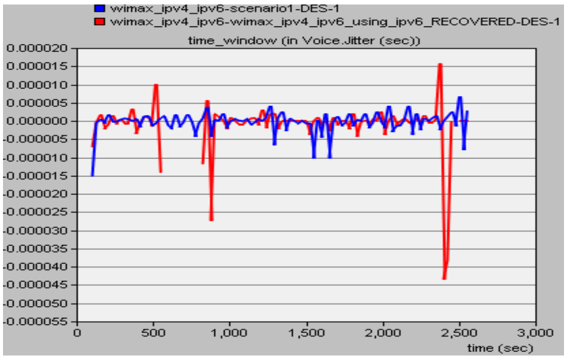

Figure 5: IPv4 vs. IPv6 (blue curve is IPv4, the red curve is IPv6) jitter, the jitter mean for IPv4=−3.64249390875E-007 and for IPv6=−1.39077861152E-006

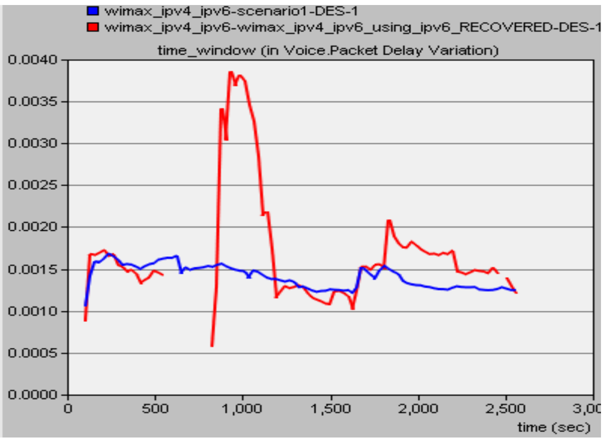

Figure 6: IPv4 vs. IPv6 (blue curve is IPv4, red curve is IPv6) packet delay variation, the packet delay variation of Pv4=0.00140361238041 (1.4036ms) and for IPv6= 0.00168223661794 (1.683ms)

VIII. CONCLUSION

A comparison between IPv4 and IPv6 delay in VoIP using WiMAX using OPNET simulator was made and it was found that IPv4 mean delay time is less than IPv6 mean delay time while end-to-end mean delay time of IPv6 is less than IPv4, in case of jitter IPv4 was found to have less variance meantime than IPv6 and in the case of packet delay variation IPv4 had less mean time than IPv6. Results where under consideration mentioned in the method and all of the delay, End-to-End delay and packet delay variation means were classified as good depending on ITU precept for voice quality as in Table 3.

REFERENCES

1. Sheetal Jalendry, S.V., *A Detail Review on Voice over Internet Protocol.* International Journal of Engineering Trends and Technology, 2015. 23.
2. http://www.trai.gov.in/, *Consultation Paper on Internet Telephone(VoIP),"* Telecom Regulatory Authority of India, 22.6.2016.
3. wikipedia.org, *Voice over IP - Wikipedia.*
4. Tucker, G.S., *Voice Over Internet Protocol (VoIP) and Security.* SANS 2004.
5. rutgers.edu, *H.323 TUTORIAL.*
6. Heyi, B.S., *Voice Over IP End-to-End Delay Measurments.* American Journal of Networks and communication, 2015. 4: p. 85-89.
7. H. Schulzrinne, S.C., R. Frederick V. Jacobson, *RTP: A Transport Protocol for Real-Time Applications.* The Internet Society, 2003. RFC 3550.
8. Qiu, Q., *Study Of Digest Authentication for Session Initiation Protocol(SIP)."* 2003.
9. Preetinder Singh, R.K., *VOIP over WiMAX: A Comprehensive Review.* International Journal Of computer science and information technology (IJCSIT), 2014. 5.
10. James Yu, I.A-A., *Call Admission Control and traffic Engineering of VoIP.* IEEE International Conference on Digital Telecommunications 2007.
11. F.Ohrtman, *WiMAX Handbook: Building 802.16 Wireless Network.* 2005.
12. Urjashee Shaw, B.S., *A Survey Paper on Voice over Internet Protocol.* International Journal of Computer Applications, 2016. 139.
13. Rasika.S.Ujjainkar, P.R.M.S., *A REVIEW PAPER ON WiMAX.* International Journal of Research In Science & Engineering. 1(1).
14. Ivan BAROŇÁK, M.H., *Mathematical Representation of VoIP Connection Delay.* Radio Engineering, 2007. 16(3).
15. TONCAR.CZ, *VoIP Basics: About Jitter.*
16. Bakyt KYRBASHOV, I.B., Matúš KOVÁČIK, Viktor JANATA *Evaluation and Investigation of the delay in VoIP Network.* Radio Engineering, 2011. 20(2).

