Investigation on VoIP for a WiMAX Communication Network

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***Abstract*— Voice over internet protocol refers to the transmission of telephone traffic over IP-based networks. Voice transmission over internet is also called internet protocol (IP) telephony. IP telephony has gained wide popularity and has become an important service on the internet. The use of IP telephony became real due to the high bandwidth available on the internet and its low implementation cost. Digital form of communication has further led to its vital use. Due to above reasons, voice communications using the IP, is also called Voice over IP or VoIP, has become attractive. The simultaneous transport of voice and data over the internet has already been demonstrated using 3G networks. This paper presents the performance evaluation of a WiMAX network. In this paper, only static conditions for the application voice over IP has been considered.**

I. INTRODUCTION

As according to the need of modern telecommunication system Wireless networking has become an essential part the demand of high speed data transfer with high quality is being the leading factor for the evolution of technologies like WiMAX and WLAN and is still increasing day by day. Therefore, new ways to improve quality and speed of connectivity are being searched for. Moving towards the fourth generation communication networks, integrated networks are coming into operation. In same manner voice over IP is expected to be a low cost communication medium. The voice codecs are big constraints which influence the quality of the voice in a high data rate communication network. Therefore, before real time deployment of VoIP over a network it is essential to evaluate the voice performance over altering networks for various codecs.

VoIP have been widely accepted for its cost effectiveness and easy implementation. A Voice over internet protocol (VoIP) system is divided into three indispensable components, namely 1) codec, 2) packetizer, and 3) playout buffer. Analog voice signals which are to be transmitted compressed, and encoded into digital voice streams by the help of codecs. The output digital voice streams are then packed into constant-bit-rate (CBR) voice packets with the help of the packetizer. A two way conversation is very sensitive to packet delay jitter but could tolerate certain degree of packet loss. Hence a playout buffer must be used at the receiver end to smooth the speech by eliminating the delay jitter. Quality of noise sensitive VoIP is generally measured in terms of jitter, MOS and packet end-to-end delay.

II. COMPONENTS OF VOIP

When a phone call is placed through a telephone, the phone is picked up, the number is dialled, and the call signals travel through the phone line to the destination; along the line the phone service providers offer quality service to ensure the clarity of the call. Much like the telephone call, VoIP also provides call signalling, quality of service (Qos), and media transport. Most call signalling is provided through either H.323 or SIP protocol. The quality of service is provided by protocols such as RSVP and RTCP. The actual media transport is through CODEC and RTP.

As with any telephone implementations, there must be a signalling scheme that alerts users that there is an incoming call or the person that is trying to be reached is busy. In VoIP, this signalling scheme along with encodings schemes and packet transfer is provided by either the H.323 or the Session Initiation (SIP) protocol. Both of these protocols are implemented in different ways, but overall provide the same service. The H.323 protocol emerged in 1996 and was designed by the International Telecommunication Union (ITU) [1]. The SIP protocol later emerged in 1999 by the Internet Engineering Task Force (IETF) [1].

The H.323 protocol provides a standard voice and multimedia conferencing product that communicates over IP networks [2]. To establish real time voice or video over the IP network, H.323 uses several other protocols. There are several CODECs that are used to convert analog audio to digital audio: G.711, G.722, G.723.1, G.728, and G.729 [2]. The process is simple; the noise received on a microphone on the transmitting terminal is converted into a digital signal using one of the CODECs and is later (after transmission) decoded on the receiving terminal using the same codec. Video works in a similar fashion, except it uses the CODEC H.611. There are three types of signalling functions: Q.931 Call Signalling Channel, the H.245 Control Channel, and the H.225 Registration, Admission, and Status (RAS) Channel [2]. Each of these functions provides different functionality. When a connection is established between the two endpoints by Q.931, H.245 provides the endpoints information of flow characteristics and status [2]. RAS on the other hand, exchanges call admissions and bandwidth management functions between the endpoint and a Gatekeeper [2].

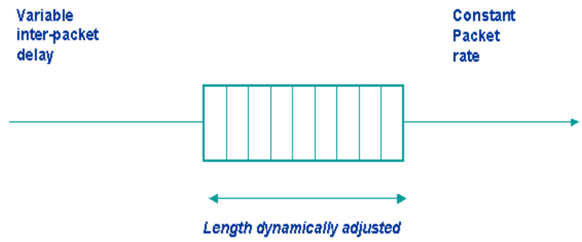
After the signalling process completes, a transport protocols takes care of all the data that needs to be transmitted through the network between the two parties. The Real-Time Protocol (RTP) provides end-to-end delivery of the audio and or video. Along with RTP, the Real-Time Transport Control Protocol (RTCP) provides feedback on the quality of the connection. SIP, on the other hand, works differently than the H.323 protocol. Like H.323, SIP provides a mean of signalling, setting up, and tearing down a VoIP session. SIP is a peer-to-peer protocol, where the peers in the session are called User Agents (UAs) [3]. A UA can function either as a client, where the client application initiates the SIP request or the UA can function as a server, where the server application listens to requests from clients and respond accordingly [3]. Things that are considered SIP clients include the phones and the gateways that provide call control. SIP servers consist of proxy servers, redirect servers, and registrar servers. A proxy server receives intermediate SIP requests and forwards the messages to the next SIP server on the network [3]. Redirect servers provide clients with information subsequent hops and registrar servers processes lookups for the UACs current location [3]. In SIP, users use SIP addresses to identify themselves. When a call first takes place, a request is made to a SIP server, which in turn find the end user or pass on the request to another SIP server [3]. Eventually, the end client will be found, and RTP will take place in the data communication between the two parties.

A well-designed voice network should make delay imperceptible regardless of the two calling parties' location. The people engaged in the call could be on the other side of the globe; their call signals may traverse thousands of miles; and the voice traffic may be transported through heterogeneous sub-networks. Yet the network should provide a fast response time so that the people engaged in the conversation feel they are right next to each other. To ensure quality of service before the call is set up, one group under IETF has developed the Resource Reservation Protocol (RSVP). It aims at ensuring each flow's QoS requirements through the complete path from the sender to the receiver. Each component involved along the path is responsible for the QoS support operation. The RSVP protocol defines a reservation procedure for real-time multimedia session. It is unique because the recipient of the traffic makes the reservation. The philosophy behind it is that the recipient should have the best knowledge of its limit. Like ICMP, IGMP, RSVP is an Internet control protocol. The key RSVP messages are Path message and Reservation message. The caller uses Path message to set up a path for the session. Once the path is set up, the receiver sends the Reservation messages, since the receiver has the best knowledge of the capacity on the receiving end. Before the call is connected, an application makes a request for QoS resources. Within the request, the application specifies the QoS requirements for the session. Both ends need to agree upon that requirement. Once the QoS is setup, the application is committed to deliver the quality of service it promised. In a sense, it is considered to be self-regulating.

III. SPEECH QUALITY AND CHARACTERISTICS

The quality of sound reproduction over a telephone network is fundamentally subjective, although standardized measures have been developed by the ITU. It has been found that there are three facts can profoundly impact the quality of the service. The two problems that results from high end-to-end delay in a voice are echo and talker overlap. Echo becomes a problem when the roundtrip delay is more than 50 milliseconds. Since echo is perceived as a significant quality problem, VoIP systems must address the need for echo control and implement some means of echo cancellation. Talker overlap (the problem of one caller stepping on the talker's speech) becomes significant if the one-way delay becomes greater than 250 milliseconds. The end-to-end delay budget is therefore the major constrain and driving requirement for reducing delay through a packet network.

Jitter is the variation in inter-packet arrival time as introduced by the variable transmission delay over the network. Removing jitter requires collecting packets and holding them long enough to allow the slowest packets to arrive in time to be played in the correct sequence, which causes additional delay. The jitter buffers add delay, which is used to remove the packet delay variation that each packet is subjected to as it transits the packet network.



$$R = \frac{Late - arrivals}{Successful - received}$$

Fig. 1 Controlling Delay and Jitter

A key requirement for successful VoIP deployment is the availability of an underlying IP-based network that is capable of real-time telephone and facsimile. As was noted above, voice quality is affected by delay, jitter, and unreliable packet delivery – all of which are typical characteristics of the basic IP network service. Most of today's data network equipment – routers, LAN switches, ATM switches, networks interface cards, PBXs, etc. – will need to be able to support voice traffic. Furthermore, VoIP-specific equipment will either have to be integrated into these devices or work compatibly with them. VoIP equipment must also accommodate environments ranging from private, well-planned corporate Intranets to the less predictable Internet.

IV. VOIP FOR STATIC ENVIRONMENT FOR WIMAX NETWORK

In the present work a wimax scenario with 19 base stations has been developed. Each base station contains 30 mobile stations hence resulting in 570 mobile stations. Each base station is connected to IP backbone as shows in fig 5.1.voice application has been defined using application definition icon and is included in profile definition. WiMAX configuration block has been configured as according to static and mobile node. IP backbone has been linked with IP cloud and IP cloud has been connected with voice server. Various parameter of voice server has been configured for voice applications.

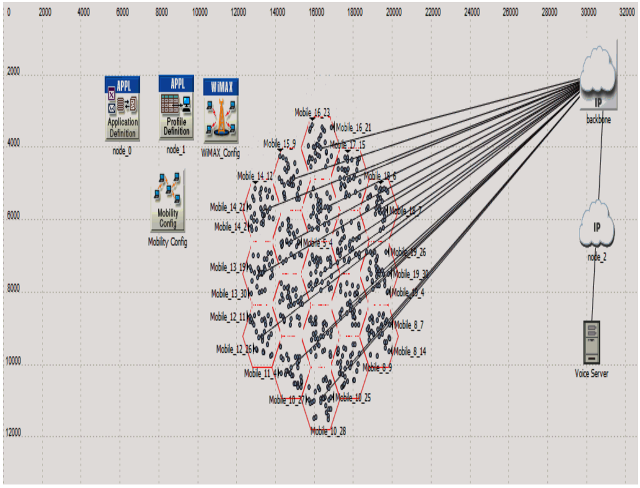


Fig. 2 VoIP-WiMAX Static Scenario

In the present work,G.711 codec has been used for coding and decoding of voice and performance of the system has been evaluated for rtps and ugs protocol.

Simulations findings for wimax static scenario

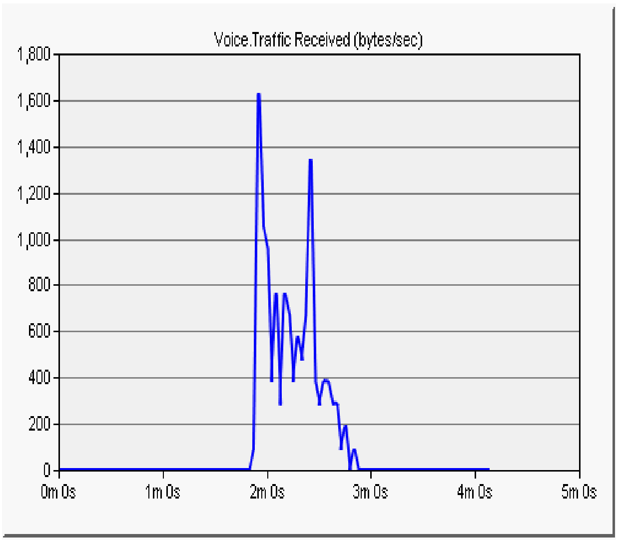


Fig. 3 Voice Traffic Received in bytes/sec for rtPS Protocol

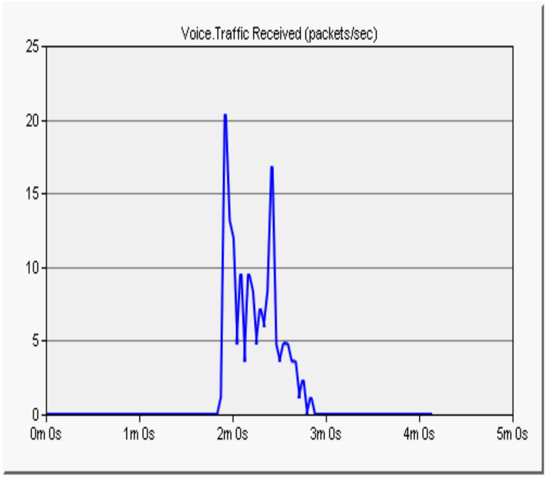


Fig .4 Voice Traffic Received in packets/sec for rtPS Protocol

Fig 3 and 4 shows variation of voice received in bytes per second and packets per second for rtps protocol respectively. From fig 3 and 4 it has been concluded that maximum value of voice traffic received is approx 1600 bytes per second and20 packet per second.

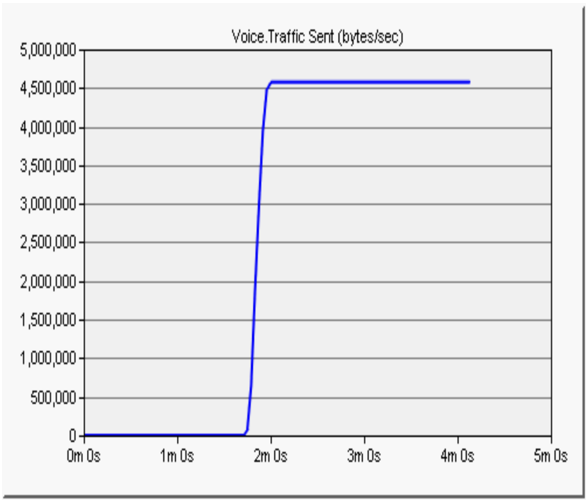


Fig .5 Voice Traffic Transmitted in bytes/sec for rtPS Protocol

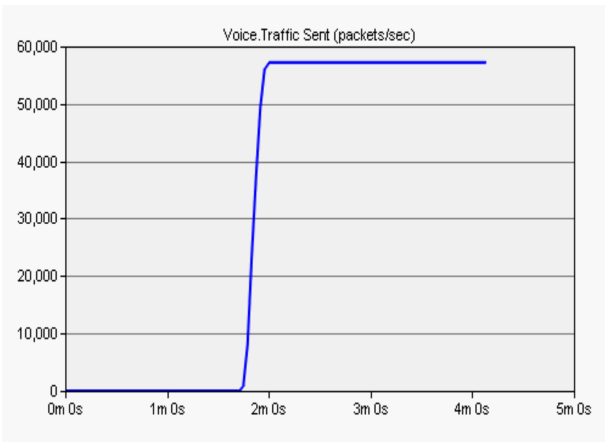


Fig .6 Voice Traffic Transmitted in packets/sec for rtPS Protocol

Fig 5 and 6 shows value of voice traffic sent in bytes /second and packets per second. From fig 5 and 6 it has been concluded that max value of voice transmitted is approx 45000 byte per second and 58000 packet per second respectively.

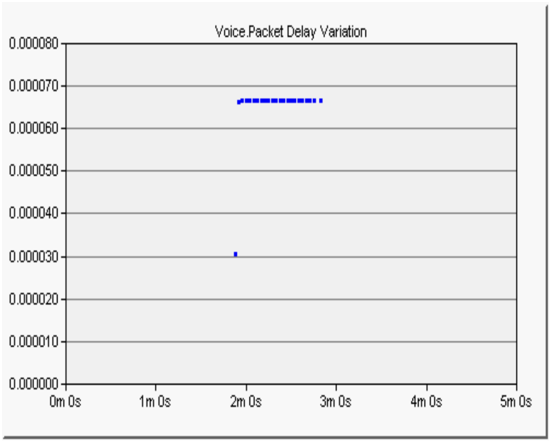


Fig .7 Variation of Voice Packet Delay for rtPS Protocol

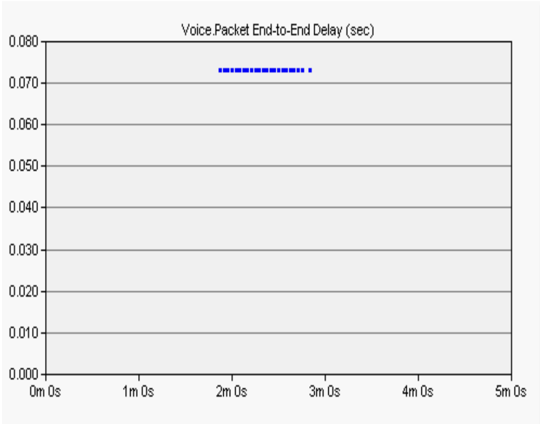


Fig .8 Variation of Voice Packet End-to-End Delay for rtPS Protocol

Fig 7 shows variation of voice packet delay for rtps protocol. The variation in voice packet delay is around 0.000068 seconds also voice end to end packet delay is around 0.072 seconds as shown in fig 8.

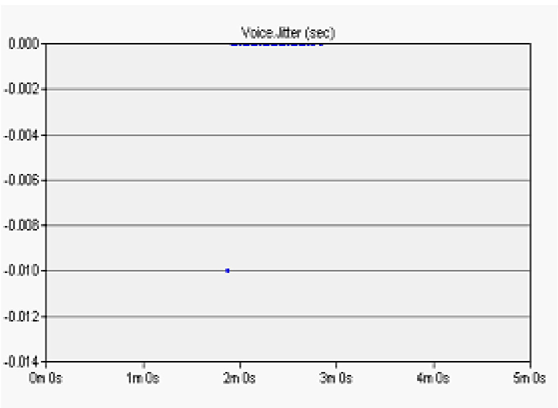


Fig .9 Variation of Voice Jitter for rtPS Protocol

Fig 9 shows value of voice jitter in seconds.fig shows that voice jitter remains almost zero during simulation session and maximum value of voice jitter has been noticed at -0.010 seconds for rtps protocol.

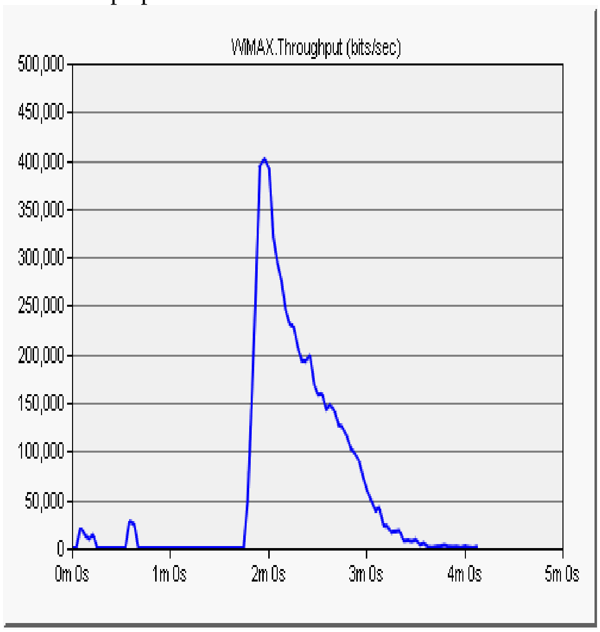


Fig. 10 Variation of WiMAX Throughput in bits/sec for rtPS Protocol

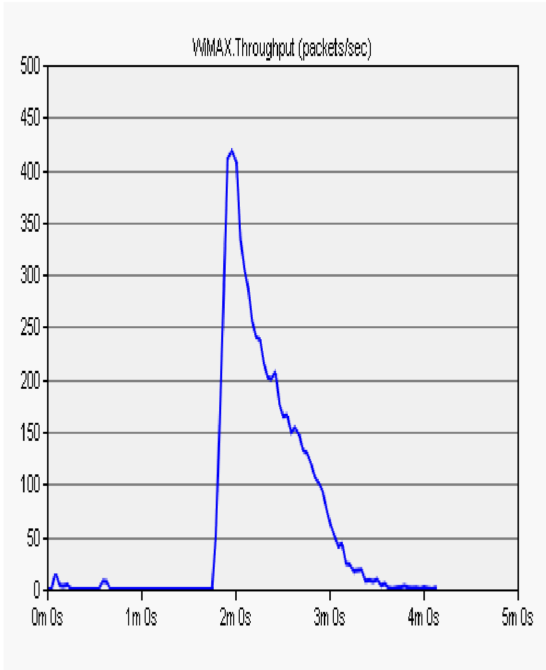


Fig. 11 Variation of WiMAX Throughput in packets/sec for rtPS Protocol

Fig 10 and fig 11 shows variation in throughput in bts per second and packets per second. for fig 10 and 11 it has been concluded that maximum value of throughput comes out to be 4*10 5 bits per second and 420 packets per second respectively for rtps protocol.

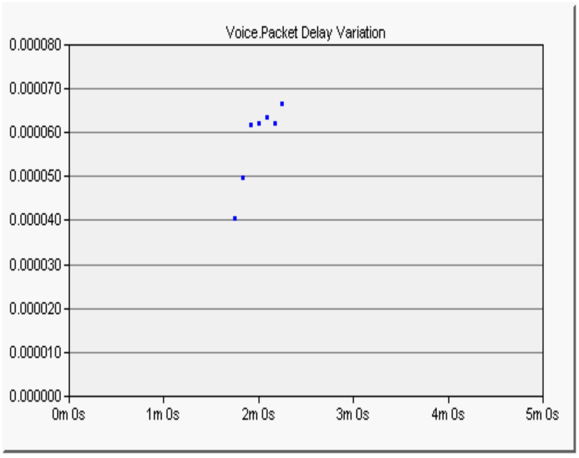


Fig. 12 Variation of Voice Packet Delay for UGS Protocol

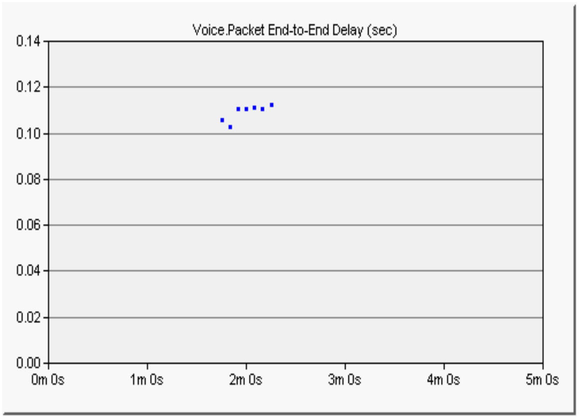


Fig. 13 Variation of Voice Packet End-to-End Delay for UGS Protocol

Fig 12 shows variation in voice packet delay for ugs protocol. The voice packet delay is not constant as per in case of rtps protocol and the value of maximum value of packet delay variation comes out to be 0.000070.also value of end to end packet delay as shown in fig 13 is not uniform for ugs protocol and is maximum value is around 0.11 seconds. Hence in terms of variation in packet delay and end to end delay rtps protocol sows better performance as compared to ugs protocol.

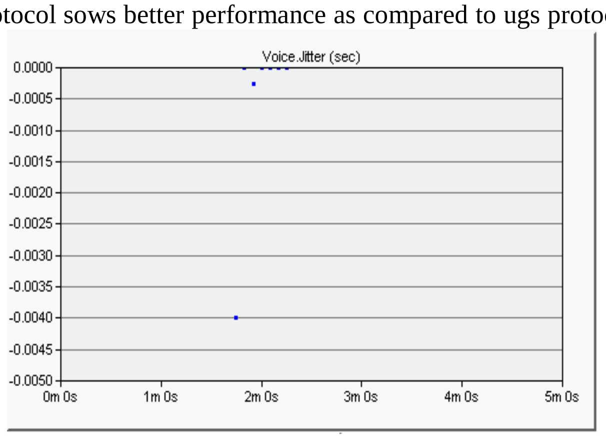


Fig. 14 Variation of Voice Jitter for UGS Protocol

The maximum value of voice jitter has been calculated as -0.0040 seconds for ugs protocol as shown in fig 14.hence in terms of voice jitter ugs protocol shows better performance over rtps protocol

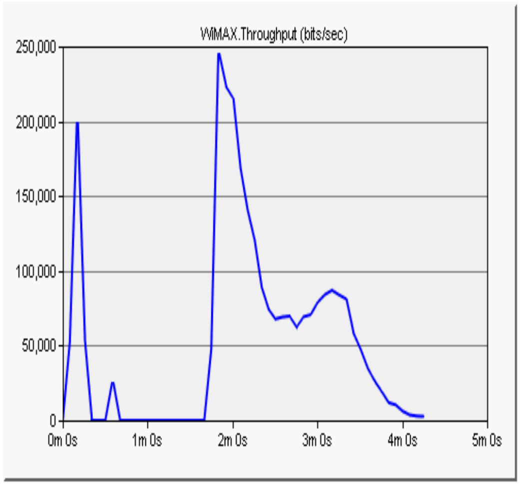


Fig. 15 Variation of WiMAX Throughput in bits/sec for UGS Protocol

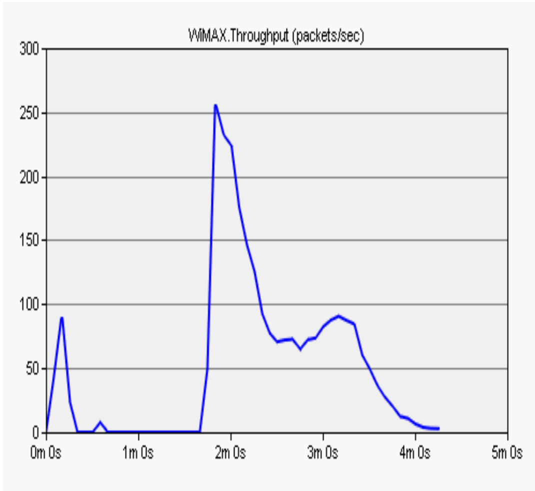


Fig. 16 Variation of WiMAX Throughput in packets/sec for UGS Protocol

Fig 15 and 16 shows throughput in bits per second and packets per second for ugs protocol. As compared to rtps protocol throughput of ugs protocol is very low and its maximum value is around 2.5 *10 5 bits per second and 250 packet per second. Hence throughput enhance in case of ugs protocol and perform much better as compared to rtps protocol.

REFERENCES

[1] Adam, F., Mateusz, M., and Marcin, P. (2010) QoE assessment of VoIP over IEEE 802.16e networks with DaVinci codes using E-model. *Future Network & MobileSummit 2010 Conference Proceedings,* IIMCInternational Information Management Corporation, 2010 ISBN:

978-1-905824-18-2,[Online]Available at:www.FutureNetworkSummit.eu/2010. [Accessed 22 08 2012].

[2] Alshomrani, S. Qamar, S. Jan, S. Khan I. and Shah, I. A. (2012) QoS of VoIP over WiMAX Acces Networks. *International Journal of Computer Science and Telecommunications*. 3 (4), p92-98.

[3] fBernardo, V.; Sousa, B. and Curado, M. (2009) VoIP over WiMAX: Quality of experienceevaluation," *Computers and Communications, 2009. ISCC 2009. IEEE Symposium on Communications*, vol. 13,no. 5, pp.42-47, 5-8 July 2009

[4] Yanfeng, G. and Aiqun, H. (2006) Bandwidth allocation algorithm of VoIP based on the adaptive linear prediction in the IEEE 802.16e system. *In Proceedings of 6th International Conference on ITSTelecommunications*, Jun. 2006, pp. 16–19.

[5] Salah, K; and Alkhoraidly, A (2006). An OPNET simulation approach for deploying VoIP. *International Journal of Network management, Vol 16, issue 3* Pp 159-183.

[6] Rajavelsang, R; Jeechgunta, V; Holur, B; Choudhary M; and Song O (2005). Performance evaluation of VoIP over 3G- WLAN internetworking system, *Wireless Communication and Networking. 2005 IEEE conference, Vol. 4 no. 5, P*p 2312-2317.

[7] Prokkola, J., Hanski, M., Jurvansuu, M. and Immonen, M. (2007) Measuring WCDMA and HSDPA delay characteristics with QoSMeT. In *Proceedings of IEEE International Conference on Communications (ICC)*, Glasgow, Scotland, June 2007, pp. 492–498.

[8] Huang, Z., Jadhav, S. and Zhang, H. (2011) Performance evaluation of quality of VoIP in WiMAX and UMTS in Parallel and Distributed Computing, Applications and Technologies (PDCAT), 2011 12th International Conference on 4G technology, PD-CAT 2011, pp. 375-380.

[9] Halepovic, E.; Ghaderi, M.; and Williamson, C. (2009) Multimedia Application Performance on a WiMAX network. In Proc. Of Annual Multimedia Computing and Networking Symposium, 2009. *IEEE.* 2 (5), pp.

[10] Inamura, H. Montenegro, G. Ludwig, R. Gurtov A. and Khafizov, F. (2003) TCP over second (2.5 G) and third (3G) generation wireless networks. RFC3481, February, 2003.

[11] Jin-Cherng, L., Chun-Lun, C. and Cheng-Hsiung L. (2008) Performance Evaluation for Scheduling Algorithms in WiMAX Network. *In Proceedings of Advanced Information Networking and Applications - Workshops, 2008.* AINAW 2008. 22nd International Conference vol.12 Issue 2, 25-28 March 2008, pp. 68 – 74 M.

[12] Gibson, J. D. and Wei, B. (2004) Tandem voice communications: Digital cellular, VoIP, and voice over wi-fi. In: *Global Telecommunications Conference. GLOBECOM'04.*

[13] Orfanus, D., Lessmann, J., Janacik, P. and Lachev, L. (2008) Comparative Study of Wireless Network Simulator. IEEE, *The Seventh International Conference on Networking, pages 517-523*, 2008.

[14] Rajavelsang, R; Jeechgunta, V; Holur, B; Choudhary M; and Song O (2005). Performance evaluation of VoIP over 3G- WLAN internetworking system, *Wireless Communication and Networking. 2005 IEEE conference, Vol.4 no. 5, P*p 2312-2317.

[15] Muntean, V., Otesteanu, M. and Muntean G. (2010) QoS parameters mapping for the e-learning traffic mix in LTE networks. *In proceedings of IEEE International Joint Conference on Computational Cybernetics and Technical Informatics,* (ICCC-CONTI), Timisoara, May 2010, pp. 299-304.

