International Journal of Scientific Research and Engineering Development-– Volume 2 Issue 3, May 2019

Available at www.ijsred.com

RESEARCH ARTICLE OPEN ACCESS

A Survey: Issues And Challenges Of VOIP traffic Over WiMAX

Anurag Saini[1], Avnish Kansal[2], Jasmeet Singh[3]

Department of Electrical & Electronics Engineering

[2, 3] Department of Computer Science Engineering

Calorx Teachers' University

Ahmedabad, India

Abstract:

With the introduction of different wireless communication technologies, the demand of users for real time services like voice along with data called VoIP is increasing tremendously. In order to provide high quality services, the new technologies like WiMAX has to consider all the quality parameters attached with a service. The paper thus discusses the WiMAX support to such services along with all the possible features of VoIP, its quality parameters like delay, jitter, throughput, etc. and emphasis on what work has been done in this perspective so as to improve the quality of the VoIP traffic by highlighting the various aspects, methodology and limitations of previous work in this field. It has been shown that, there still remain some issues and considerations, which could be taken into account so as to have high quality VoIP traffics.

Keywords— Quality of Service; VoIP issues; WiMAX

I. INTRODUCTION

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 [1]. 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. WiMAX (worldwide interoperability for microwave access) is type of broadband wireless access which offers the additional functionality of portability, nomadicity and mobility. WiMAX accommodates various applications such as Web surfing and quicker file downloads, but also enables several multimedia applications, namely real-time audio and video streaming, multimedia conferencing, and interactive gaming. WiMAX connections are also being used for voice telephony using voice over Internet Protocol (VoIP) technology [2]. More narrowly if we see that one of the most desired requirements of today's smartphone users is the need of always on connection, means the voice telephony along with the data. For example, one is downloading a data file and a call comes in between, the requirement is that the file downloading should not stop by the call interference it should go parallel with the voice and that too with high speed. The paper studies how such technologies are able to support such requirements, and what issues are there behind carrying such data.

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. Hence, before real time deployment of VoIP, it is essential to evaluate the voice performance over the wireless networks.

Thus, in the coming sections we have given the in-depth state of art of VoIP traffic over the WiMAX networks. Section II discusses the WiMAX technology, VoIP and its architecture; section III gives the details of the issues and challenges faced by the networks to carry VoIP traffic. The related work on different quality parameters of VoIP has been discussed in Section IV along with the conclusion, describing what work could be done in this area in section V.

II. WIMAX TECHNOLOGY

WiMAX is developed by the IEEE 802 group with the aim of providing the users a high internet speed at far range called 'last mile' coverage with an 'always on connectivity'. It provides voice telephony, along with faster Web surfing and quicker file downloads and also enables several multimedia all at the same time, such as real-time audio and video streaming, VoIP, video conferencing, and interactive gaming. It provides services with high quality [3]. WiMAX offers a variety of applications each with different traffic patterns and QoS requirements which must be fulfilled to get high user satisfaction. The QoS is a term that refers to the "collective effect of service," as perceived by the user or more narrowly it refers to meeting certain requirements such as throughput, packet error rate, delay, and jitter associated with a given application [4].

WiMAX architecture consists of broadly three subsections namely the Access Service Network, the Connectivity Service Network and the IP network. They are shown in Fig.1.

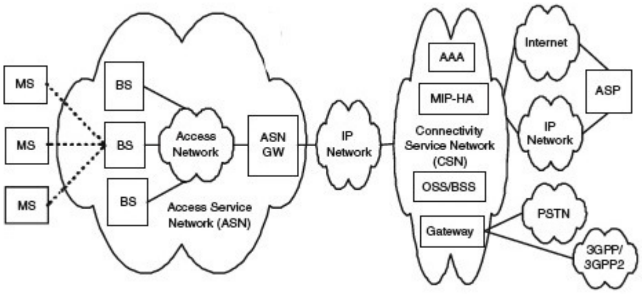

Fig.1 WiMAX Architecture.

Access Service Networks: It consists of number of transceivers called as base stations BS and the SSs. The communication is possible through the electromagnetic waves called radio frequencies, on which the information is carried, the BS are used for this purpose. The subscriber stations are the devices through which the user access the service, which could be a mobile phone, laptop, dongles etc.

Connectivity Service Network: It is the core network which is the basic wireless communication providing network or the standard network 2G, where all the calls procedures, requests, authentication, management and billing are done.

IP Network: It is the one which connects the WiMAX access network with the core network through internet protocol which further enables the users to get connected to the communication network along with the global internet which provides them real time and non-real time services.

In order to rapidly converge on a worldwide standard, a number of versions are there in the IEEE 802.16 family described in table 1 [5].

Table 1. WiMAX Specifications

Parameters	***802.16***	***802.16 - 2004***	***802.16e - 2005***
Frequency Band	10GHz – 66GHz	2GHz – 11GHz	2GHz – 11GHz for fixed, 2GHz – 6GHz for mobile applications
Application	Fixed LOS	Fixed NLOS	Fixed and mobile NLOS
MAC Architecture	Point to multipoint, Mesh Architecture	Point to multipoint, Mesh Architecture	Point to multipoint, Mesh Architecture
Transmission Scheme	Single Carrier Only	Single Carrier, 256 OFDM or 2048 OFDM	Single Carrier, 256 OFDM or Scalable OFDM with 128,512,1024 or 2048 subcarriers
Modulation	QPSK, 16 QAM, 64 QAM	QPSK, 16 QAM, 64 QAM	QPSK, 16 QAM, 64 QAM
Data Rate	32Mbps- 134.4Mbps	1Mbps- 75Mbps	1Mbps-75Mbps
Multiplexing	Burst TDM/TDMA	Burst TDM/TDMA/ OFDMA	Burst TDM/TDMA/ OFDMA
Channel Bandwidth	20MHz, 25MHz, 28MHz	1.75MHz, 3.5MHz, 7MHz, 14MHz, 1.25MHz, 5MHz, 10MHz, 15MHz, 8.75MHz	1.75MHz, 3.5MHz, 7MHz, 14MHz, 1.25MHz, 5MHz, 10MHz, 15MHz, 8.75MHz
Duplexing	TDD and FDD	TDD and FDD	TDD and FDD

A. *VoIP*

VoIP has been widely accepted for its cost effectiveness. VoIP converts the analog voice signal from a telephone or computer into a digital packetized signal that can be transmitted over the internet. Before transmitting the analog voice signals they are compressed and encoded into digital voice streams with the help of codecs. VoIP system is divided into three indispensable components, namely codec, packetizer and the playout buffer [6]. The general overview of VoIP architecture is shown as in Fig. 2.

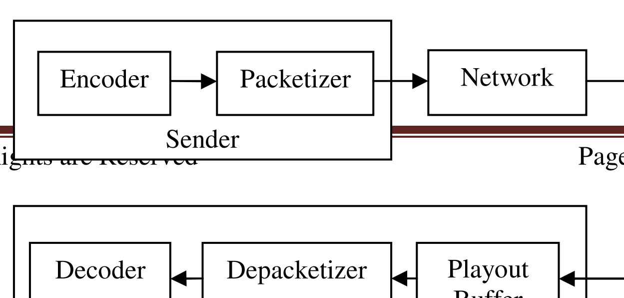

Fig.2.VoIP Architecture.

The output digital voice streams are packed into constant-bit-rate (CBR) voice packets using packetizer. A two way conversation is very sensitive to packet loss, delay and jitter. 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 [7]. Perceived voice with zero jitter, high MOS and low packet end-to-end delay is assumed to be the best. To obtain digital format of the analog signal process is utilized which is called encoding and the converse is called decoding and both are performed by voice codecs. As bandwidth is enormous concern, compression techniques are utilized to reduce bandwidth consumption. But problem related by using codecs is the overhead of algorithmic delay, thus codec is assumed to provide good quality even after compression with minimum delay. The different types of codecs with their specifications are described in table 2.

Table 2. Various Codes and their specifications.

Codec	***Data Rate***	***Datagram Size***	***A/D Conversion Delay***	***Combined Bandwidth (Bidirectional)***
G.711u	64.0 kbps	20 ms	1.0 ms	180.80 kbps
G.711a	64.0 kbps	20 ms	1.0 ms	180.80 kbps
G.729	8.0 kbps	20 ms	25.0 ms	68.80 kbps
G.723.1 (MPMLQ)	6.3 kbps	30 ms	67.5 ms	47.80 kbps
G.723.1 (ACELP)	5.3 kbps	30 ms	67.5 ms	45.80 kbps

III. ISSUES AND CHALLENGES OF VOIP

A. Bandwidth:

Bandwidth allocation is a major concern in VoIP as when it is split off between voice and computer data, it is necessary for a network to allocate bandwidth for the voice communication.

B. Securing:

Another big challenge for the VoIP is to secure the voice communication so that it cannot be overheard or blocked. The transmission is protected can be guaranteed by using a double encryption process which uses X.509 for authentication and 152-bit AES, 3DES or 56-bit DES for the data flow.

C. Link Failure:

Link failure is the loss of the series of packets during a period of time, which continue for the few minutes or seconds resulting into delay after re-establishing the link. Link failure can occur due to the various reasons. It can caused by a problem in the equipment, a cable wire being unplugged or disconnected, change in the configuration of the transport network or denial of service attack. Routers are capable enough in detecting a link failure and finding an alternative route.

D. Packet loss:

Bandwidth limitation and the way the packets move inside the network are the reasons for the packet loss on the network. When the IP packets are introduced via a router over the server, few packets may get lost, resulting in a period of muteness in the conversation as the multiple packets containing IP address would move in one direction or in another direction which leads to clipped-speech effect that is not satisfactory and acceptable by the users.

E. Network Design:

Network design affects the reliability of the VoIP communications. If the network is not designed to manage the combined transmission of voice and data packets, then the reliability of the system may get ruined. Moreover, the network design should support the new updated voice applications which are only possible by using the novel converged voice and data network.

F. Noise:

As voice communication is sensitive to noise. So, noise is the main cause due to which the signal reaches the destination with a lead or lag in the specific time period. The deviation in the signal characteristics is called jitter. The lead and lag both degrades the voice quality as lead causes the negative jitter and lag is responsible for the positive jitter. Total time taken by the packet to reach the destination from the source is called packet end-to-end delay. This delay should be minimum for voice communication. Perceived voice quality is normally measured using an arithmetic average of opinion score defined as mean opinion score (MOS). MOS of a specific codec is the standard mark provided by a panel of auditors paying attention to distinct recorded samples [8]. This will range from 1 (unacceptable) to 5 (excellent) described in table 3. It will depend on delay and packet loss by the network.

Table 3. Mean Opinion Score

Quality of Scale	*Range*	*Listening Effort*
Excellent	5	No effort required
Good	4	No appreciable effort Required
Fair	3	Moderate Effort Required
Poor	2	Considerable effort Required
Bad	1	No meaning understood with Reasonable effort

IV. Literature Survey

N.Nagarajan et al., (2009) defines the problem generates in the process of video conferencing and in various multimedia applications while delivering the video packets over the WiMAX. A model is formed using a different competing traffic sources over a point-to-multipoint topology. To handle the VoIP and flow of video traffic the performance investigation on the capacity of the WiMAX device was carried out. The parameters which describe the QoS such as throughput, average delay and jitter or the packet loss is examined for the multiple types of service flows that are defined in the WiMAX systems [9].

Bong-Ho Kim et al., (2009) focuses on a system which evaluates the performance and capacity of the mobile WiMAX systems. For the performance evaluation all the factors from the air link to the application are necessary. At the beginning they describe a mobile WiMAX systems briefly which specifically talks about the OFDMA/TDD systems of IEEE 802.16e and thereafter it explains subscriber and application profiles which contain data session attempts, traffic mix ratio, diurnal application traffic distribution and model of application-traffic. Along with this it also gives the simulation results of demand estimation and characteristics of network-traffic. Eventually, in the last portion it discusses the performance evaluation of end to end application with the examples of VoIP and also provides the performance enhancement method for TCP/IP which can be realized in the mobile WiMAX MAC or cross layer MAC/IP [10].

Jadhav, S. et al., (2011), Next generation wireless networks put light on conjunction of numerous Radio Access Technologies (RATs) supporting good Quality of Service (QoS) for the various applications like Voice over IP traffic (VoIP), video conferencing and video streaming. The voice applications are rising quickly due to their rapid increase in its popularity and cost. It is very much essential to develop a suitable model for QoS to accommodate the demand of giving VoIP service of high quality at any time, at any cost and from anywhere. For supporting the VoIP they organized the simulation study to find the quality of service performance of WiMAX and UMTS. The simulation results conclude that the WiMAX outperforms over the UMTS with adequate margins and is suitable technology, which can support applications of VoIP [11].

Henriques, J., et al., (2012) The momentous increasing of VoIP have raised various challenges in the deployment of WiMAX or Long Term Evolution (LTE), well known as novel broadband wireless access networks (BWA). To get the successful deployment the practical assessment of voice traffic evaluating the performance and quality is essential. In this paper, the capabilities of Mobile WiMAX (IEEE 802.16e) which supports VoIP traffic under distinct scenarios and engaging various Quality of Service (QoS) service classes were implemented. Further, the paper distinguishes the conditions of heterogeneity access within a city area by examining both conditions: Line of Sight (LOS) and Non-Line of Sight (NLOS). By evaluating the end-user perceived quality (Quality of Experience) and the network parameters of QoS, the obtained results show the correct QoS service classes management on the numerous well served users of VoIP [12].

Baig, M.T. et al., (2013), this states that as the users of real time applications such as VoIP is gaining so much attention so it is essential to attain effective and efficient service and mobile user must get the continuous connectivity with parallel node. This paper illustrates the influence of Vertical Handovers (VHOs) on the functioning of VoIP. The Results conclude that the existing protocols such as Realtime Transport Protocol (RTP) and TCP Friendly Rate Control (TFRC) do not support the Quality of Service (QoS) requirements during VHOs. A scheme Adaptive Vertical Handover Rate Control (AVHRC) is proposed which gets the VoIP's QoS requirements during vertical handovers. AVHRC provide an effective rate control mechanism and obtains the data of upcoming possible access technology. It also evaluates the link stability for AVHRC which supports the QoS requirements. The result summarizes that AVHRC shows the improvement in terms of packet loss, latency and throughput as compared to RTP and TFRC for the distinct mobile scenarios [13].

Ben Salem, A. et al., (2014) Long Term Evolution which is a latest wireless standard identified by the 3GPP uses the Voice over Internet Protocol to broadcast the voice services and data packets. In LTE technology the traffic scheduling plays a crucial role by authorizing the shared resources to the users in the most powerful manner. The main aim of this research is to develop an effective scheduling algorithm for this standard. It not only gets the high performance of the system, but also capable of maintaining fairness. Each user is assigned the requested resource with respect to the already defined QoS parameters. The simulation is carried out using MATLAB and the performance of the scheduling algorithms on the downlink side is evaluated [14].

V. CONCLUSION

From the exhaustive literature review, it has been concluded that Mobile WiMAX can not only be used to fulfill the demand for high internet speed, but can also be used to provide voice over- IP services. The low-latency design of mobile WiMAX makes it possible to deliver VoIP services more effectively and VoIP technologies may also be used to provide innovative services like voice chatting, push-to-talk and multimedia chatting. Several good works have discussed the capacity and performance of WiMAX networks. But there appears a scope for a comparative discussion of the performance of a WiMAX network with respect to the application of VoIP.

REFERENCES

[1] Li, B.; Qin, Y.; Low, C. P.; & Gwee, C. L., "A survey on mobile WiMAX", Communications Magazine, IEEE, vol. 45, no.12, pp.70-75, 2007.

[2] Rohani, B.; Ibrahim, M.; D. M. Ali, "Performance measurement of VoIP over WiMAX 4G network", Signal Processing and its Applications (CSPA), IEEE 8th International Colloquium on, IEEE, 2012.

[3] C.Eklund; R.B. Marks; K.L.Stanwood and S.Wang, "IEEE Standard 802.16: A Technical Overview of the WirelessMAN Air Interface for Broadband Wireless Access", IEEE Communications Magazine, pp. 98-107, June 2002.

[4] C. Cicconetti; A. Erta; L. Lenzini; and E. Mingozzi, "Performance Evaluation of the IEEE 802.16 MAC for QoS Support," IEEE Transactions on Mobile Computing, Vol. 6, no. 1, pp.26-38, 2007

[5] Pareit, D.; Lannoo, B.; Moerman, I.; Demeester, P., "The history of WiMAX: A complete survey of the evolution in certification and standardization for IEEE 802.16 and WiMAX", Communications Surveys & Tutorials, IEEE, vol. 14, no. 4, pp. 1183-1211, 2012.

[6] Bernardo, V.; Sousa, B.; Curado, M., "VoIP over WiMAX: Quality of experience evaluation", In Computers and Communications, IEEE Symposium on, pp. 42-47, July 2012.

[7] Lee, T. Y.; Pan, J. Y., "Improving R-score of adaptive VoIP codec in IEEE 802.16 networks", In Communications, APCC 2008, 14th Asia-Pacific Conference on, IEEE, pp. 1-5, October 2008.

[8] Shamik Sengupta; Mainak Chatterjee; Samrat Ganguly, "Improving Quality of VoIP Streams over WiMAX", IEEE Transactions on Computers, Vol.57, NO.2, pp.145-156, February 2008.

[9] Kaarthick, B.; Yeshwenth, V. J.; Nagarajan, N.; Rajeev, "Performance analysis of Video Conferencing and Multimedia application Services over WiMAX," Advance Computing Conference, IACC 2009, IEEE, pp.1109-1113, 6-7 March 2009.

[10] Bong-ho Kim; Jungnam Yun; Yerang Hur; Chakchai So-In; Jain, R.; Al Tamimi, A.-K., "Capacity estimation and TCP performance enhancement over mobile WiMAX networks," Communications Magazine, IEEE, vol.47, no.6, pp.132-141, June 2009.

[11] Jadhav, S.; Haibo Zhang; Zhiyi Huang, "Performance Evaluation of Quality of VoIP in WiMAX and UMTS," Parallel and Distributed Computing, Applications and Technologies (PDCAT), 12th International Conference on, pp.375-380, 20-22 Oct. 2011.

[12] Henriques, J.; Bernardo, V.; Simões, P.; Curado, M., "VoIP performance over mobile WiMAX: An urban deployment analysis," Future Internet Communications (BCFIC), 2nd Baltic Congress on, vol., no., pp. 148-155, 25-27 April 2012.

[13] Baig, M.T.; Shah, Z.; Baig, A., "AVHRC: A scheme to improve QoS for VoIP traffic," Telecommunication Networks and Applications Conference (ATNAC), IEEE, pp.100-105, 20-22 Nov. 2013.

[14] Ben Salem, A.; Bouallegue, S.; Sethom, K., "A QoS Based Resource Allocation in Femtocell Networks," Embedded and Ubiquitous Computing (EUC), 12th IEEE International Conference on, pp.299-303, 26-28 Aug. 2014.