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VOIP Over Wimax: A Comprehensive Review

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Abstract-Worldwide interoperability for Microwave access(WIMAX) is a 802.16 wireless communication standard that provides high speed, throughput and cover larger area. Besides quality of service support, ieee 802.16 standard offer data rate upto 100mbps and cover area upto 50km.VOIP through wimax is most prominent service and is a growing rapidly in world of telecommunication. Recent studies focussing on qos scheduling services and performance related metrices such as jitter,mos and packet end to end delay have been addressed in this paper.

Keywords-IEEE802.16, wimax, Mos,qos,jitter,voip

I. INTRODUCTION

The evolution of wireless communication systems and networks in recent years has been accelerating at an extraordinary pace and become an essential part of modern life style requirements [1]. The demand of high speed data transfer with high quality is being the leading factor for the evolution of technology like Wimax and is still increasing day by day .Wimax is a standard for wireless communication which supports higher number of users with higher data rates, coverage and availability.This technology is based on IEEE 801.16 standards and the most relevant versions are the Fixed Wimax, based on IEEE 801.16d, and the mobile Wimax, defined by IEEE 802.16e [2]. Moving towards the fourth generation communication networks, integrated networks are coming into operation. In same manner, VOIP is expected to be a low cost communication medium.

A. HISTORY OF VOIP

VOIP is a protocol that allows users to make calls over the internet..Voip is also referred as Internet telephony, IP telephony or voice over the internet.Few of commercial computer applications which make use of voip applications such are skype, yahoo messenger and google talk[3]. Advantages and disadvantages of voip are summarized in following table

Table 1 shows Advantages and Disadvantages of Voip

Advantages	Disadvantages
Low cost	Users cannot make calls during power outrages
Free voip calls from anywhere for long distances or international calls	Emergency calling is not provided by voip Ip network does not provide gurantee quality of service.
Easy to implement and install	
More scalable	
Provides free voice call and call forwarding	

VOIP transmits packet via packet switching in which voice packet follow most efficient path.On the other hand , public switched telephone network (PSTN) is a circuit switch network in which dedicated path is decided before the transmission starts.

Table 2 shows comparison between Voip and Pstn

Feature	Voip	Pstn
Cost	Cheap	Expensive
Switching	Packet switching	Circuit switching
Path	No dedicated path	Dedicated path
Reliability	Less reliable	More reliable
Connectivity	Internet connectivity	Dedicated telephone lines
Scalability	Update require more bandwidth and simple software update	Update require more hardware which can be more costly and complex
Call waiting	Voip offers free call services in skype	Available at extra cost
Emergency calling	Emergency calling is not provided by voip	Emergency calling is enabled
Call forwarding	Free in google talk but paid in skype	Available at extra cost

B. VOIP COMPONENTS

VoIP have been widely accepted for its cost effectiveness and easy implementation. A VOIP system is divided into three components,namely codec, packetizer, and playout buffer.

CODEC : The function of codec is to compress and encode the analog signals into digital voice signals. Codec provide good quality of voice even after compression, with minimum delay which is one of the main advantage of using codec.

PACKETIZER : With the help of packetizer output digital streams are packed into constant bit rate voice packets.

PLAYOUT BUFFER : A two way conversation is very senstive to packet delay jitter so to eliminate the delay, playout buffer is used at receiver end.

Quality of noise sensitive in VoIP is generally measured in terms of jitter, mean opinion score (MOS) and packet end-to-end delay. Perceived voice with zero jitter, high MOS and low packet end-to-end delay is assumed to be the best. Before transmitting voice over internet which is an analog signal should be converted into digital format. To obtain digital format of the analog signal process is utilized which is called encoding and converse is called decoding and both are performed by voice codecs. As bandwidth is enormous concern, compression techniques are utilized to reduce bandwidth consumption.

In recent days voice codecs are developed to detect talk-spurt within a conversation. In communication, silence is a period leads to packetization of the background noise and sending it through the network, which results bandwidth wastage. Normally, during a conversation client talk 35% of the time and remain quiet in the remaining time. By using the silence suppression technique during the silence period, the codec does not send data. This reduces the channel utilisation and therefore saves bandwidth.

As voice communication is noise sensitive, noise is the main cause due to which the signal to reach the destination either cause negative jitter or positive jitter .Both of these jitter will result in degradation of voice quality The time taken by voice to be transmitted from the source to the destination is called packet end-to-end delay [4]. This delay should be very less for voice communication .

QOS SERVICE CLASSES IN WIMAX

To fulfill all performance related parameters such as mos,end to end delay,jitter,wimax utilize different scheduling mechanism to allocate downlink & uplink transmission oppurtunities for different pdu.

Table 3 classifies different service classes defined in WIMAX and its description and qos parametes.

Table 3.QOS service classed in wimax

Service	Description	QOS parameters
UGS	Designed to support real time constant bit rate(cbr) traffic such as voip that periodically generates fixed size data packets.	Maximum sustained rate,maximum latency tolerance
RTPS	Designed to support MPEG video & teleconferencing that periodically generates variable size data packets.	Maximum reserved rate,maximum sustained rate,maximum latency tolerance,traffic priority.
NRTPS	Designed to support non real time application with minimum rate such as ftp.	Minimum reserved rate,maximum sstained rate,traffic priority.
BE	Designed to support data stream which do not require minimum service level gurantee.This QOS is used for internet service such as email and web browsing.	Maximum sustained rate,traffic priority.
ERTPS	Designed to support real time application ,such as voip with silence suppression that have variable data rate .	Minimum reserved rate,maximum sustained rate,maximum latency tolerance

C. VOIP (QOS) QUALITY OF SERVICE PERFORMANCE METRICS

Speech quality performance measurements are divided into subjective and objective methods [5].The methods are explained briefly in following paragraph :

SUBJECTIVE METHODS :The main aim of subjective method is to find perceived quality of speech based on user perceptions. ITU-T in Recommendation P.800 [6] introduced MOS based on user perception ranging from 1(poor) to 5(excellent).

OBJECTIVE METHODS: The E-model (online voice quality measurement), an analytical model is defined in ITU-T recommendation, provides a framework for an purpose on-line quality estimation based on network performance measurements like delay and loss .

MOS- MOS is a technique used to check the work of codecs which compress audio and video files.MOS of a particular codec is the standard mark given by a panel of auditors listening to various recorded samples. This will range from 1(unacceptable) to 5 (excellent). MOS values depend on several factors, not only the network parameters such as delay and packet loss but also on codec used [7]

Table 4 shows relationship between mos and quality of speech

Mos	Quality of speech
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

JITTER-Technically defined as measurement of variation in arrival time.If consecutive packets leave source node with time stamp t1 & t2 and are played at destination node at time t3 & t4 then

JITTER= (t4-t3)-(t2-t1)

PACKET END TO END DELAY-The time taken by voice to be transmitted from source to destination is called packet end-to-end delay.The total voice packet delay is calculated as

De2Ee=Dn+De+Dd+Dc+Dde

Where De2e represent end to end delay while Dn,De,Dc,Dde represents network,encoding,decoding,compression and decompression delays.

II. RELATED WORK

Rohani bakar [8] in his research paper stated that quality of voice in wimax network is satisfying with excellent jitter and good latency with 4.0 for mos scale.His work also show that bandwidth does not influence the jitter in the network.His findings also show that both jitter and latency values for wimax was low as compared to broadband network.

N.Nagarajan et al [9] addresses the problems concerning the delivery of video packets in video conferencing and other multimedia application services over WiMAX. Multiple competing traffic sources over a point-to-multipoint WiMAX topology is modeled. The performance analysis on the capacity of the WiMAX equipment to handle VoIP and video traffic flows was conducted. Parameters that indicate quality of service, such as throughput, packet loss, average jitter and average delay, are analyzed for different types of service flows as defined in WiMAX.

Jadhav, S. et al [10] focus on convergence of different Radio Access Technologies (RATs) providing good Quality of Service (QoS) for applications such as Voice over IP traffic (VoIP) and video streaming. The voice applications over IP networks are growing rapidly due to their increasing popularity and cost. To meet the demand of providing high-quality of VoIP at anytime and from anywhere, it is imperative to design suitable QoS model. They have conducted simulation study to evaluate the QoS performance of WiMAX and UMTS for supporting VoIP. They designed simulation modules in OPNET for WiMAX and UMTS, and carried out extensive simulations to evaluate and analyze several important performance metrics such as Mean Opinion Score (MOS), end-to-end delay, jitter and packet delay variation. According to results it was shown that WiMAX outscores the UMTS with a sufficient margin, and is the better technology to support VoIP applications compared with UMTS.

Chin-Ling Chen and Cheng-Yi Pan [11] presented an efficient way and the provision of Quality of Services (QoS) guarantee are the major issues in delivering delay sensitive traffic, like VoIP service in WiMAX (Worldwide Interoperability for Microwave Access). One well-designed scheduling algorithm is expected to coordinate QoS-related functional entities in WiMAX architecture. Existing downlink scheduling algorithms of WiMAX like DRR (Deficit Round-Robin) and WRR (Weighted Round-Robin) usually reserve minimum rate to each type of traffic and cannot consider the status of queue length of each connection, thus making it unsuitable for VoIP on-off traffic model. Author proposed an efficient downlink scheduling algorithm, which allocate the bandwidth based on queue-length estimation and compared the proposed scheme with DRR and WRR by estimating the system performance such as average delay, loss rate and throughput under several traffic scenario and system parameters value.

Henriques, J. et al [12] presented that the Mobile WiMAX (IEEE 802.16e) capabilities to support VoIP traffic under different scenarios and employing distinct Quality of Service (QoS), service classes were performed. Additionally, they characterized the heterogeneity access conditions within a city, by analyzing both Line of Sight (LOS) and Non-Line of Sight (NLOS) conditions. After examining the end-user perceived quality (Quality of Experience) and the network QoS related parameters, the attained results shown the impact of the correct WiMAX QoS service classes management on the number of well served VoIP users.

Adi Chandra [13] has investigated that extended real time polling service (ertps) scheduling service which initially support variable rate real time services imgsproves performance of voip over wimax.

Islam,Rashid and tariq[14] analyzed that voip over gsm enhanced full rate (gsm-efr) & gsm full rate (gsm-fr) codecs achieve desirable spech quality with les delay and jitter.

III. CONCLUSION

From Exhaustive literature review, it has been concluded that Mobile WiMAX can not only be used to fulfil the demand for high internet speed but can also be used to provide voiceover- 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. In this paper, extensive survey of paper published in field of wimax has been addressed.

REFERENCES

1. http://www.internationalgraduate.net/school/brunel-university-the-school-of-engineering-and-design/program/wireless-communication-systems-msc-M1898
2. Joao Henriques,Vitor Bernado,Paulo Simoes,Marilia Curado, "Voip Performance over Mobile Wimax: An Urban Deployment analysis" IEEE Communications ,2012
3. http://en.wikipedia.org/wiki/Voice_over_IP
4. Nataranjan Meghanathan, Selma Bourmerdassi , Napendu Chati & Dhinaharan Nagamalai, "Recent Trends in Networks & Communication" Springer-Verlag Berlin Heidelberg Publishers,pp 318
5. R.J.B Reynolds,A.W.RIX," Quality VOIP-An enginnering challenge,"BT Technology journal,vol.19,no.2.pp.23-32,2001
6. ITU-T Recommendation p.800,methods for subjective Determination of Transmission Quality,Augest 1996
7. Rohani Bakar,Muhammad Ibrahim and D.M.ali ,"Performance Measurement of Voip over wimax 4g network" International colloquium on signal processing and its applications,2012
8. Joao Henriques,Vitor Bernado,Paulo Simoes,Marilia Curado,"Voip Performance over Mobile Wimax: An Urban Deployment analysis" IEEE Communications ,2012
9. Nagarajan,N,Rajeev,Kaarthick,B,"Performance Analysis of Video Conferencing and Multimedia Applications Services over Wimax" International Advance Computing Conference (IAAC),2009
10. Jadhav , S., Zhang,H, & Huang," Performance Evaluation of Quality of Voip in Wimax and UMTS"12th International Conference on Parallel and Distributed Computing, Applications and Technologies, 2011
11. Chin-Ling Chen and Cheng-Yi Pan " A Downlink Scheduling based on queue length estimation for Voip in Wimax Networks" 5th International Conference on Biomedical Engineering and Information (BMEI),2012
12. Joao Henriques,Vitor Bernado,Paulo Simoes,Marilia Curado,"Voip Performance over Mobile Wimax: An Urban Deployment analysis"IEEE Communications ,2012
13. I.Adhichandra,"Measuring data and Voip traffic in wimax networks," Journal of telecommunications,vol.2,no.1,pp.1-6,apr.2010
14. Tariq,M.I.,Azad,M.A.,Beuran,R.and shinoda,Y."Performance analysis of voip codecs over BE Wimax network," 3rd International conference on computer and electrical engineering(ICCEE 2010),vol .4,no.4,pp.77-88,july2012