Evaluation VoIP over WiMax

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Abstract

Worldwide inter operability for Microwave access (WIMAX) is an 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. The VoIP is flexible and provides low cost telephony to customers over the existing IP infrastructure. However, there are still many challenges that need to be addressed to provide a steady and good quality voice connection over the best-effort Internet. In this paper we evaluate the performance of different VoIP codecs such as G.711, G.723.1 and G.729 over the best effort WiMAX network. The network performance metrics such as jitter used to evaluate the performance of VoIP codecs. The performance is analyzed by using OPNET Moeller. *Keywords: WiMAX, VoIP, OPNET 14.5*

1. 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.

1.1 WiMax Overview

WiMAX (Worldwide Interoperability for Microwave Access) is a new wireless digital communications system standard being Developed by the IEEE (Institute of Electrical and Electronics Engineers) That focuses on solving the problems of point-to-multipoint broadband Outdoor wireless networks, and it's one of the "4G" communication Networks[3].

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 [4], which added some improvement to 802.16 - 2004 standard, 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 a proved alternative to 3G or wireless LAN networks for providing ease of access, low cost and large coverage area [5].

1.2 VOIP Technology

A. VoIP Overview

Voice over Internet Protocol (VoIP) is also called as IP Telephony which allows users for real-time voice communication over the internet. The concept of VoIP goes back to 1970 but it started to be used since 1995 [3]. VoIP simply converts the voice analogue signals into digital signals to be carried over the internet to reach the other end. VoIP has significant benefits comparing to the traditional telephones. Some of the benefits are as follows [4]:

B. VoIP Benefits

- Obvious Cost Saving: it's the most valuable and attractive benefits comparing to the traditional dial-up telephony system especially in case of long distance or international calls. With VoIP, customer can call anywhere in the world with only internet subscription charge.
- Portability: VoIP provides the concept of mobility and portability so that the users can install a soft phone in their mobiles having the same number virtually and travelling anywhere.
- Integrated and Collaborated Services: VoIP is able to provide services other than voice the customers since it runs on the application layer such as voice mail delivery and messaging system while running real-time voice communication simultaneously.

C. VoIP Codecs

• 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.

• **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.

• **G.723.1** was once the recommended compression standard. It operates at 6.3 and 5.3 Kbps. High com pression with high quality audio. Although this standard reduces bandwidth consumption, voice is much poorer than with G.729 and is not very popular for VoIP [3].

1.3 Quality of Service

Quality of Service (QoS) has no specific definition but it can be described as the ability of a network to provide well services of delivering traffic from sources to destinations. QoS is without doubt very important to be used in VoIP communication which is considered as a critical traffic. QoS of specific traffic could be measured by several parameters such as end-to-end delay, delay variation and packet loss. The main objective of VoIP QoS is to meet the quality level of Traditional Telephony System [6]. In this paper jitter is the performance metrics QoS that is taken in consideration.

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 [7].

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

2. Previous Study

Rohani bakar [2] 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.

Elechi Onyekachi O. and Eze Elias C [8] Studied and analyzed voip over wimax network using different voip codec schemes. G711 came out to be the best codec in performance than the other G.729 and G.723 codec schemes and that the choosing a codec schemes affects that the QoS of the system.

N.Nagarajanet 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.

3. System Models

In this experiment, we used OPNET Modeler version 14.5 with WiMAX Module capability we designed four scenarios to determined jitter. Three of them do the jitter measurements by various codecs and this is the first part. The fourth scenario we increase the number of workstations to obtain its effect on jitter the second part. In first part we use an application configuration and profile configuration to insert the VoIP application in work stations. IP router is used to connect the base station and the ip cloud (internet), point –to- point (PPP DS3) as a connection between IP router and internet and Ethernet (100 baseT) connect the IP router and base station. Also fife work stations are used with one wimax subscriber station.



Fig (1): WiMax Network

Configuration mber of Rows P Profile Name Applications i-Number of Rows	node_4 () 1 voip
mber of Rows p Profile Name Applications	1 voip
p Profile Name Applications	voip
Profile Name Applications	
Applications	
Number of Powe	- \···/
"number or nows	1
voice	
Operation Mode	Serial (Ordered)
Start Time (seconds)	uniform (100,110)
Duration (seconds)	End of Simulation
Repeatability	()
- Inter-repetition Time (seconds)	exponential (200)
- Number of Repetitions	constant (0)
Repetition Pattern	Serial
F	Number of Repetitions

Fig (2): VoIP Application Profile

Three codec G7.11, G7.23 and G7.29 are used as shown in figure below.

	Name	- D		
(Description) Table	⊀ (Voice) Table		
Attribute	Value	Attribute	Value	
Database	Off	Silence Length (seconds)	default	
Email	Off	Talk Spurt Length (seconds)	default	
Ър	Off	Symbolic Destination Name	Voice Destination	
-ttp	Off	Encoder Scheme	G.729 A	
Print	Off	Voice Frames per Packet	1	
Remote Login	Off	Type of Service	Interactive Voice (6)	
Video Conferenc	ting Off	RSVP Parameters	None	
/oice	()	Traffic Mix (%)	All Discrete	
Details	Promote	Details Promote	<u>0</u> K	<u>C</u> ancel

Fig(3): codec G.729

	Name	(Voice) Table			
(Description) T	able	Attribute	Value		
Attribute	Value	Silence Length (seconds)	default		
Database	Off	Talk Spurt Length (seconds)	default		
Email	Off	Symbolic Destination Name	Voice Destination		
Ptp	Off	Encoder Scheme	G.711		
Http	Off	Voice Frames per Packet	1		
Print	Off	Type of Service	Interactive Voice (6)		
Remote Login	Off	RSVP Parameters	None		
Video Conferencing	Off	Traffic Mix (%)	All Discrete		
Voice	()	Details Promote	QK Cancel		

Fig(4): codec G.711

	Name		Ҟ (Voice) Table		X
(Description) Table		Attribute	Value	
Attribute	Value		Silence Length (seconds)	default	
Database	Off		Talk Spurt Length (seconds)	default	
Email	Off		Symbolic Destination Name	Voice Destination	
Ptp	Off		Encoder Scheme	G.723.1 5.3K	
Http	Off		Voice Frames per Packet	1	_
Print	Off		Type of Service	Interactive Voice (6)	
Remote Login	Off		RSVP Parameters	None	
Video Conferenc	ting Off		Traffic Mix (%)	All Discrete	•
Voice	()		Details Promote	<u>0</u> K	Cancel
<u>D</u> etails	Promote	<u>0</u> K			Galica

Fig (5): codec G.723

Part two is similar to the first scenario but increase the number of base stations from 5 to 35 as shown in figure blown:



Fig(6):Wimax Network Model

4. Results

Our results dividing into two parts the first part shows the different values of jitter gotten from three codec G.711, G.723 and G.729 as shown in table one.



Fig (7): shows the three codec

Table 1: The Jitter Value

Codec scheme	Voice jitter
G.711	0.014 ms
G.723	0.024 ms
G.729	0.032 ms

We found that codec G.711 is the best one

- Part two view the result if we increase the number of workstations as shown in the table (2).



Fig (8): Shows the Value of Jitter in Case of Increase Number of Workstation to 35

Table 2: Workstations Vs Voice Jitter

Number of work stations	Voice jitter
5	0.015 ms
35	0.020 ms

The jitter has lower value in a less number of work stations.

5. Conclusion

In this paper, the voice service classes using different VoIP codecs have been simulated and analyzed in terms of average jitter. Three voice codecs such as G.711, G.723 and G.729 were simulated in order to find the most appropriate voice codec for VoIP over WiMAX network. We found the lowest jitter value gotten by codec G.711 so that it is the best of the three and also increasing the number of workstations causes increasing the jitter value.

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