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Improving Quality of VoIP over WiMAX

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Abstract

Voice over Internet Protocol (VoIP), also called ipTelephony, is rapidly becoming a familiar term and technology that is invading enterprise, education and government organizations. VoIP is designed to replace the legacy TDM technologies and networks with an IP based data network. Digitized voice will be carried in IP data packets over a LAN and/or WAN network. Installing and testing the VoIP network of IP phones, gateways and servers requires new tools and expanded knowledge. In parallel, a remarkable increase is happening in the deployment of IEEE 802.16 standard based WiMAX networks. wimax is really will known term in telecommunication .in this paper we investigating the performance of VoIP traffic over WiMAX networks. OPNET modeler simulator have been used to design various scenarios by three codecs G.711, G.723 and G.729 to determine how it affect the jitter in wimax network also different topology have been applied " dividing area into cells" to enhance the performance of voip over wimax.

Keywords: *WiMAX, VoIP Codecs, OPNET.*

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. Mobile WiMAX is an all IP network. The use of OFDMA on the physical layer makes it capable of supporting IP applications. It is a wireless solution that not only offers competitive internet access, but it can do the same for telephone service.

Voice over Internet Protocol (VoIP) offers a wider range of voice services at reduced cost to subscribers and service providers alike .VoIP is expected to be one of the most popular WiMAX applications. Its value proposition is immediate to most users. While WiMAX is not designed for switched cellular voice traffic as cellular technologies as are CDMA and WCDMA, it will provide full support for VoIP traffic because of QoS functionality and low latency. IPTV enables a WiMAX service provider to offer the same programming as cable or satellite TV service providers. IPTV, depending on compression algorithms [2], requires at least 1 Mbps of bandwidth between the WiMAX base station and the subscriber.

1.1 How WiMAX works?

The WiMAX network is just like a cell phone. When a user send data from a subscriber device to a base station then that base station broadcast the wireless signal into channel which is called uplink and base station transmit the same or another user is called downlink. The base station of WiMAX has higher broadcasting power, antennas and enhanced additional algorithms. WiMAX technology providers build a network with the help of towers that enable communication access over many kilo metres. The broadband service of WiMAX technology is available in coverage areas.[3] The coverage areas of WiMAX technology separated in series of over lied areas called channel. When a user sends data from one location to another the wireless connection is transferred from one cell to another cell. When signal transmit from user to WiMAX base station or base to user (WiMAX receiver) the wireless channel faces many attenuation such as fraction, reflection, , wall obstruction etc. These all attenuation may cause of distorted, and split toward multi path.

The target of WiMAX receiver is to rebuild the transmitted data perfectly to make possible reliable data transmission [3].

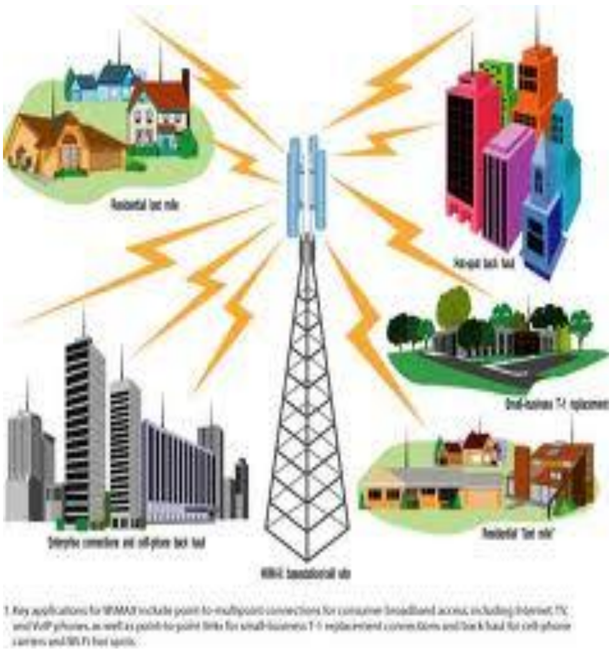

Figure 1: WiMax Model [14]

1.2 VoIP Protocol Overview

The voice over IP (VoIP) protocol suite is generically broken into two categories, control planeprotocols and data plane protocols. The control plane portion of the VoIP protocol is the traffic required to connect and maintain the actual user traffic. It is also responsible for maintaining overall Network operation (router to router communications). The data plane (voice) portion of the VoIP protocol is the actual traffic that needs to get from one end to another[4].

1.3 CODECs

There is a wide range of voice CODECs (coder/decoder or compression/decompression) protocols available for VoIP phone implementation. The most common voice CODECs include G.711, G.723, G.726, G.728, and G.729. A brief description of each CODEC follows.

G.711 – Converts voice into a 64 kbps voice stream. This is the same CODEC used in traditional TDM T1 voice. It is considered the highest quality.

G.723.1 – Two different types of G.723.1 compression exist. One type uses a Code Excited Linear Prediction (CELP) compression algorithm and has a bit rate 5.3 kbps. The other type uses an Multi-Pulse - Maximum Likelihood Quantizer (MP-MLQ) algorithm and provides better quality sound. This type has a bit rate of 6.3 kbps.

G.726 – Allows for several different bit rates, including 40, 32, 24, and 16 kbps. It works well with packet to private branch exchange (PBX) interconnections. It is most commonly deployed at 32 kbps.

G.728 – Provides good voice quality and is specifically designed for low latency appli-cations. It compresses voice into a 16 kbps stream.

G.729 – One of the better voice quality CODECs. It converts voice into an 8 kbps stream. There are two versions of this CODEC, G.729 and G.729a. G.729a has a more simplified algorithm over G.729, allowing the end phones to have less processing power for the same level of quality [4].

1.4 WiMAX Cell Site Design

One of the most important technical and business issues of any wireless technology is efficiently (cost and performance) providing coverage and capacity, while avoiding the build-out of a large number of new base stations. The first step in designing a wireless system is to develop a link budget. Link budget is the loss and gain sum of signal strength as it travels through different components in the path between a transmitter and receiver. The link budget determines the maximum cell radius of each base station for a given level of reliability and is comprised of two types of components: system related components and non-system related components [5].These components are important factors when evaluating the complexity and speed in deploying at higher frequency bands, especially in unlicensed bands such as 5.8 GHz (licensed in some countries such as Russia). Other factors like interference from other surrounding networks will also impact network performance and quality of service. Path loss, shadow margin, environmental effects, and morphology are important factors when planning for an optimum coverage. The morphology and physical surroundings of a cell site play a very important role in determining the cell footprint. A cell site footprint can shrink from 7km in a mostly flat area with light tree densities to 3 km in a hilly terrain with moderate–to heavy tree densities [17]. Traditional RF planning remains the fundamental limiting factor in system performance in WiMAX. With adaptation of Erceg Model [18], the cell size for several carrier frequencies from 450MHz to 3.5GHz is estimated for WiMAX systems using path loss propagation models for flat rural, hilly rural and urban environment

Figure 2: Real Image for Wireless Tower [14]

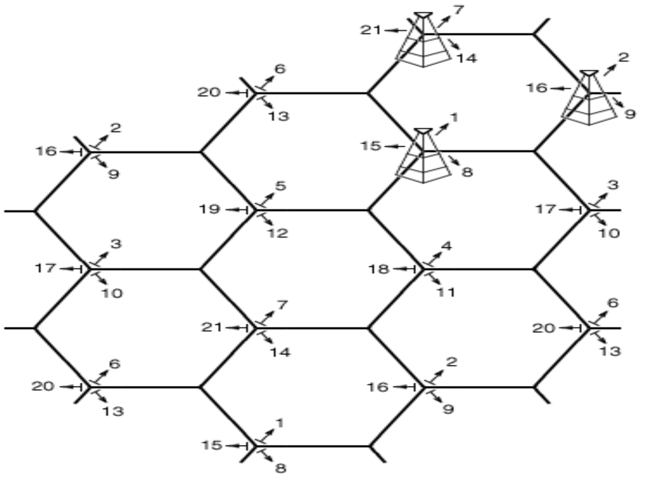

Figure 3: Cell Site Design [14]

2. Related Work

A rapid growth has been noticed in various wireless technologies in recent years. This has resulted in an increase in demand for wireless data services and multimedia application such as VoIP, streaming audio and video [6]. In order to provide good service and to meet the user demands, research has been in progress both in wireless technologies and VoIP network system. VoIP is becoming more and more popular especially after the deployment of WiMAX network in many countries [7]. Different aspects of VoIP over WiMAX have been addressed by researchers. The authors in [8] have investigated the data and voice support in the WiMAX network. The aim of their work was to examine the QoS deployment over WiMAX network and compare the performance obtained using two different WiMAX services classes i.e. UGS and ertPS. The author in [9] has pointed out different factors like delay, jitter and packet losses and discussed how WiMAX network can deal with them. In [10], the authors have considered a fixed WiMAX network in order to evaluate the performance of VoIP. They have measured the performance of different transmission schemes in term of cumulative good put, packet rate, sample loss rate and Mean Opinion Score (MOS) using R-score specified by ITU-T. In [11], the authors have proposed a traffic- aware scheduling algorithm for VoIP applications in WiMAX networks. They have studied the performance of their proposed method and compared it with that of some conventional methods. They have discussed the trade-off between delay and bandwidth efficiency and it is shown that using their scheduling methods enhances the efficiency of VoIP over WiMAX. The authors in [12] have discussed different issues related to VoIP and voice quality measurement models. They have outlined a new methodology for developing models for nonintrusive prediction of voice quality. The researchers in [13] have presented a voice quality measurement tool based on the ITU-T E-model. They have tested the tool in some calls generated through the RNP backbone, between two endpoints located at different Brazilian cities.

3. Simulation Setup

To evaluate the performance of VoIP over the WiMAX network the OPNET Modeler version 14.5 with WiMAX Module capability was used to design scenario to determined jitter with the assumption that the only traffic generated in this network model is VoIP. There are only peer-to-peer voice calls throughout the simulation, which means there is no voice conferencing and the workstations are considered as fixed during the simulation runs. Fig. 1 illustrates the WiMAX network model considered in the simulations. The WiMAX network consists of seven Cells and an IP backbone. The cell radius is set to1 Kilometers. Each cell consists of four workstations and one Base Station (BS) in each cell. We use an application configuration and profile Configuration to insert the VoIP application in work stations shown in figure (5).we use point –to- point (PPP DS3) as a connection between the backbone (internet) and every base station in cell and also as connection between the voice server and backbone there is a server backbone containing only one Voice Server. Our experiment was performed to show Voice quality is important for VoIP system because of the users' high demands for good quality voice services. In these scenarios, we considered the use of various voices codecs in the same WiMAX network in order to investigate the performance of voice codecs for VoIP. G.711 (64kpbs), G.729 (16kbps) And G.723 (5.8kbps) were considered for this experiment shown in figures blew.

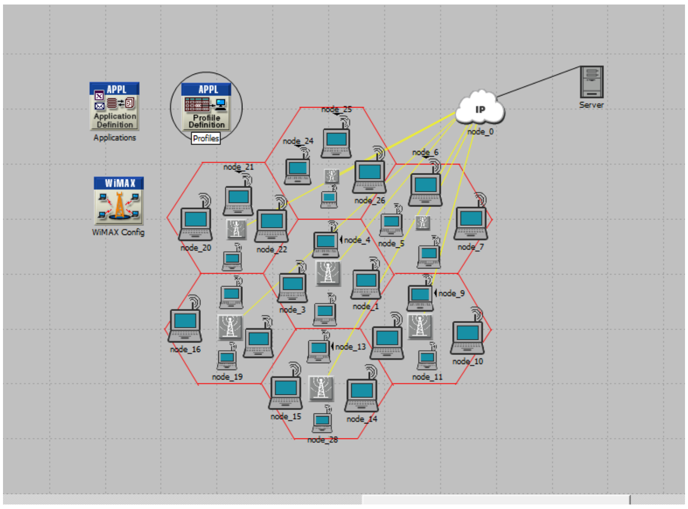

Figure 4: WiMax Network Model

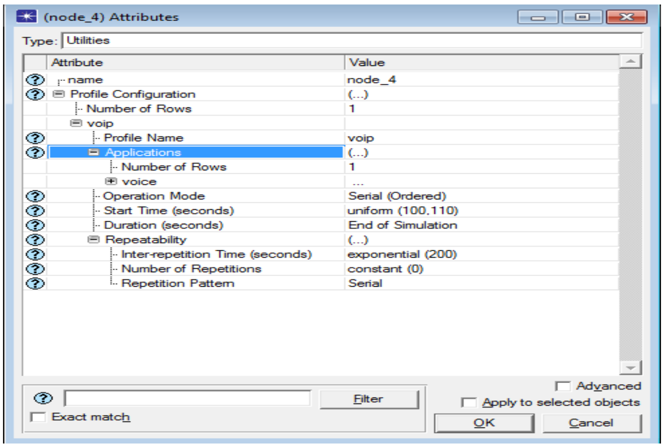

Figure 5: VOIP Application Profile

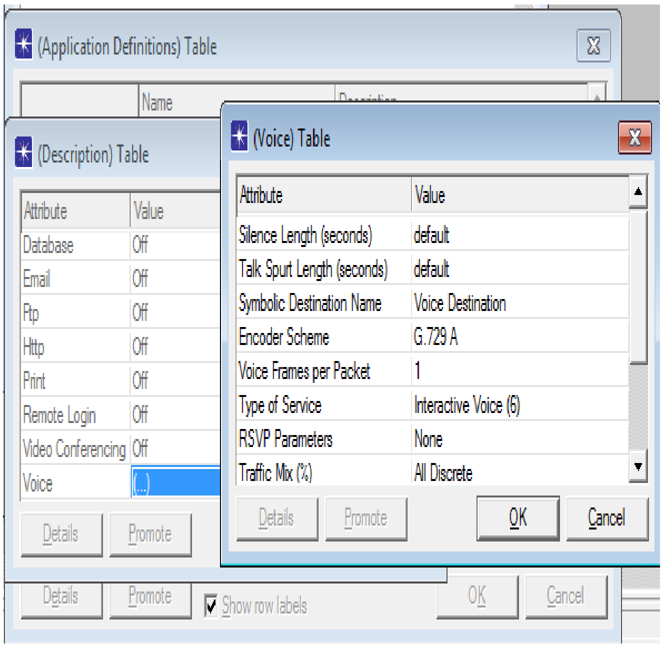

Figure 6: Configurations for ITTs' G.729 Encoder Schemes

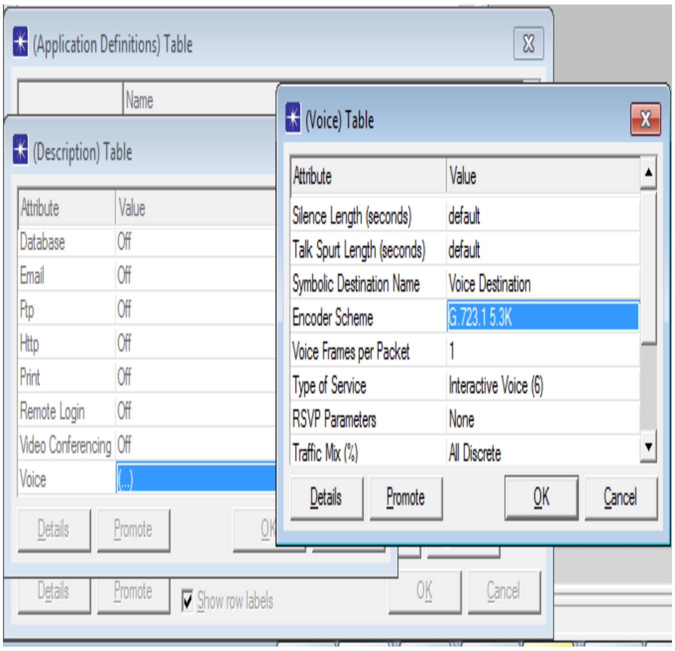

Figure 8: Configurations for ITU-Ts' G.723 Encoder Schemes

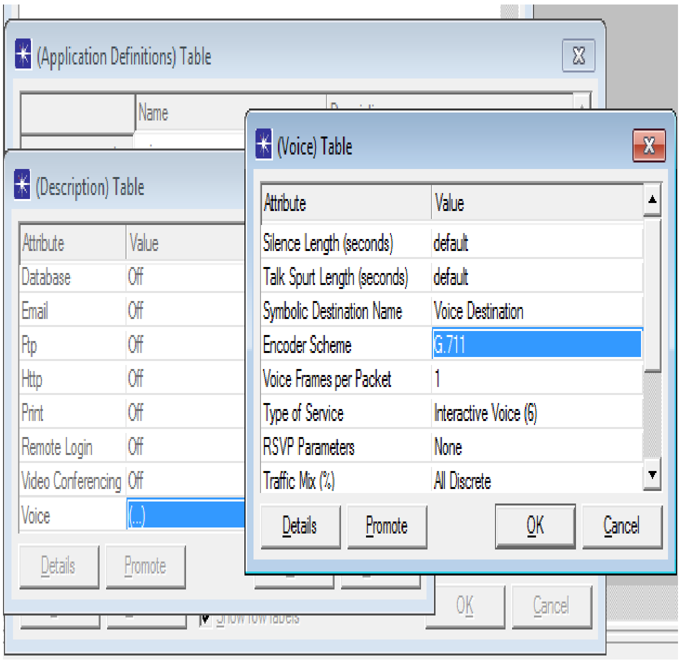

Figure 7: Configurations for ITU-Ts' G.711 Encoder Schemes

4. Simulation Result

This part compared the performance of voIP over wimax through three scenarios with different codecs and fig 9,10 shows the jitter value in each case also the comparison between the vice jitter values is clearly understood from the table 1 blew .and we found that codec G.711 and G.729 have the lower voice jitter value equals to is (-5.4814371004E009) ms and codec G.723 have large voice jitter value equals to of (-6.30714106421E-007) ms

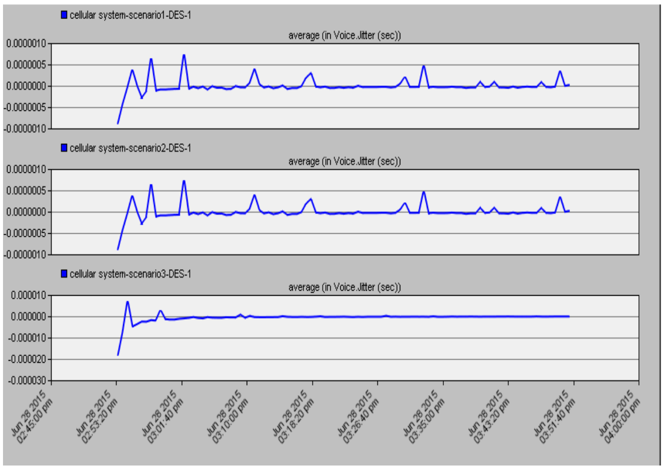

Figure 9: Voice Jitter for G.711, G.723 and G.729

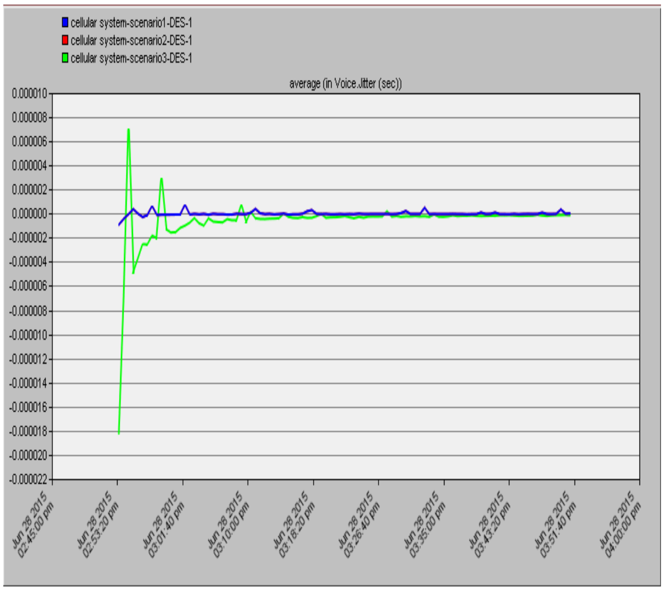

Figure 10: The Jitter Values of Three Codec Together

Table 1: Summary of Experimental Results (Voice Codec)

Codec scheme	Voice jitter value
G.711	-5.4814371004E-009 ms
G.723	-6.30714106421E-007 ms
G.729	-5.4814371004E-009 ms

5. Conclusion

In this paper we discuss the performance analysis of voIP over wimax by using three different voice codecs G.711, G.723 and G.729 depending on our simulation results we discover that the voice QoS became better if the jitter value is the lowest. also different topology have been used and this definitely improving the QoS because the area have been divided into seven cells each one have its own base station and continue a certain number of workstation in instead of one base station cover all the area and this has positive effect on decreasing voice jitter value.

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