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Research Article

Evaluation of Voice Codecs of VoIP Applications for MANET

Ekta Agrawal[Á*] and Kanojia Sindhuben Babulal[Á]

[Á]Department of Computer Science and Engineering,, UIT, Allahabad, India

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Abstract

Over Internet (VOIP) is an emerging popular Internet application that provides good services through Mobile Adhoc Network(MANET).MANET provides a suitable platform for the deployment of voice and multimedia session over IP network in many application scenarios that range from safety to comfort related services. But this network also faces many challenges on QoS issues due to packet loss because of transmission errors, short battery life and dynamic changing topologies. WiMAX is a wireless 4G technology, currently using by various industries &institutions which provides many advantages due to its high speed and large coverage area. Combining these technology together, in this paper we estimated the performance of various VOIP codecs with WiMAX in different scenarios (sparse and dense) over MANET. Voice codecs are evaluated with some QoS metrics such as average MOS, average jitter, average throughput, average delay and signal received with error. The performance & quality of VOIP applications using H.323 signaling protocol in Qualnet 6.1 simulator are studied in order to determine their effect on QoS and find out most efficient network specific voice codec .

Keywords: *VoIP, QoS, Codecs, WiMAX, MANET*

1. Introduction

Wireless communication enables a user to access the communication services at anytime from anywhere around the world. Now a day's user can move around all over the world while maintaining the connectivity with the rest of the world. This type of communication is categorized as Mobile Computing Network. Existing mobile computing network can be classified into – a) Infrastructure based network. b) Mobile Adhoc network(MANETs).MANETs are decentralized network consisting of mobile nodes equipped with wireless communication without any access point or any existing infrastructure(Hekmat Ramin,2006).This self organizing capability of MANET makes it suitable for various circumstances like to set up a network on emergency basis, for military environment, for civilian environment such as taxi cab network, meeting rooms and for personal area networking like cell phone, laptop, earphone ,wrist watch etc. MANET applications are basically concerned for voice transmission over IP network like tele-emergency system that needs voice communication.

Voice over IP is a very popular technology that allows the communication over packet switched network instead of circuit switched network (D.Minolli *et al*,2002).This facilitates the voice call using internet telephony along with additional capabilities instead of analog telephone network.

In order to ensure the QoS on VOIP network a suitable voice codec is required. It is the choice of CODEC determines the quality of voice communication with limited end to end delay and low packet loss rate. Primary function of codec is to perform voice digitization (analog/digital signal conversion) and compression.

The main objective of this paper is to evaluate the performance of several voice codecs those are used in VOIP transmission over MANET. The main contributions of this work are as follows-

- To design a model for MANET with WiMAX network for providing high data rates and broad coverage area.
- To study different Qos metrics like Mean opinion score(MOS),Jitter, Throughput, Delay, Energy Consumption, Signal received with error in order to evaluate Codec performance in small scale network and in large scale network.
- Encapsulating WiMAX connectivity to achieve accurate simulations.
- Testing different voice codecs using QoS metrics under the effect of small scale and large scale architecture with table driven routing protocol(Bellmen ford)
- The performances of simulated network have been analyzed using Qualnet simulator 6.1.

The rest of the paper is organized as follows-Section II describes voice transmission over IP over Manet.Section III overviews the details of the simulation methodology. Result and performance analysis are described in section IV. Then section V overviews the related work. Finally section VI describes future work and conclusion.

2. Related Work

(El Brak *et al*,2012) has done study on VOIP applications over VANET taking only urban scenario. (Said El Brak *et*

*Corresponding author: **Ekta Agrawal**

al,2013) evaluated the speech quality evaluation for VOIP in WiFi network.(Francesca Martelli *et al*,*2012*)measured the VoIP performance over IEEE 802.11p in V2I network in terms of throughput and jitter.(S. Alshomrani *et al*, 2012)have discussed the QoS of VoIP over WiMAX but only three codecs are evaluated under small scale seven scenario.(M. Imran Tariq *et al*,2013) has analyzed the capacity of WiMAX network, but VoIP performance regarding throughput & delay was not analyzed. (Tucker *et al*,2006) discussed how WiMAX deals with the various factors that affects the network performance. The main goal of this research work is to analyze the performance VOIP in MANET through QoS metrics. Different voice codecs are simulated under the effect of sparse and dense scenarios through both network level (such as loses) and user lever (MOS) metrics.

3. Voice over IP over MANET

Manet provides a suitable platform for the deployment of VOIP applications in many scenarios as the quality of service requirements is a major challenge for real time applications. This section provides a brief overview of VOIP implementation, voice codecs, H.323 signaling protocol

A. *VOIP*-Now a day's subscriber wants to communicate via e-mail, instant messaging, video etc. in addition to the voice traffic. For this type of multimedia communication VOIP technology is most appropriate that allows communication over packet switched network between two parties. It is a methodology that allows the transmission of voice and multimedia sessions over Internet that reduces the cost of long distance voice calls or sometimes offers free of cost service irrespective of the distance (R.G.Cole *et al*,2009).Voice communication in a VOIP system can be illustrated by a block diagram fig 1. Initially human voice remains in the analog form so before transmitting this voice over packet switched analog signal has to be digitized at the sender side and the reverse process (depacketized and decoded) is performed at the receiver side. Digitization process is composed of sampling, quantization and encoding. The quality of voice communication over IP is mainly influenced by the choice of voice codec.

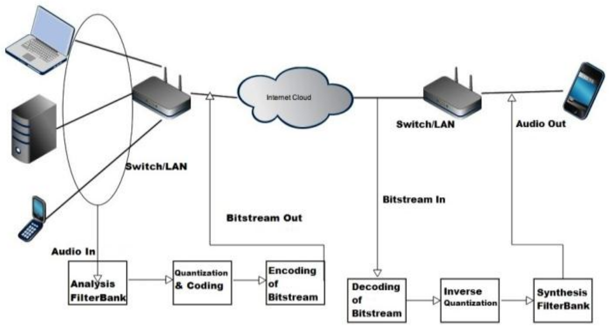

Figure1

B. *CODECs*- The application of voice codec process is the initial step for voice communication. It's primary function is convert the incoming analog voice pattern into digital stream and vice versa .International Telecommunication Unions (ITU-T) has developed various encoding techniques which are classified into waveform coders (eg-G.726,G.711), Vocoders and Hybrid Coders(eg-G.723,G.728,G.729).As the number of mobile nodes varies quality of the voice also gets affected. So the choice of codec is the significant factor whose main objective is to perform voice digitization and compression ensuring the lowest bit rate possible without degrading the signal quality.

C. *Signaling protocol*- The encoded speech in the form of packets is when transmitted over IP network requires appropriate signaling protocol like H.323 or SIP(J. Rosenberg *et al*,2002).In this paper we had used H.323 multimedia signaling protocol and WiMAX network.H.323 is a ITU-T's recommended standard protocol for set up and tear down calls of IP telephony. It is responsible for encoding, decoding, packetizing, signaling & control as well as capabilities exchange of audio and video signals (Davidson Jonathan *et al*; ITU-T Recommendation, 1998).

4. Simulation Methodology

For the evaluation of VOIP applications performance over MANET Qualnet simulation tool 6.1 is used which determines the behavior of system in virtual computational world.

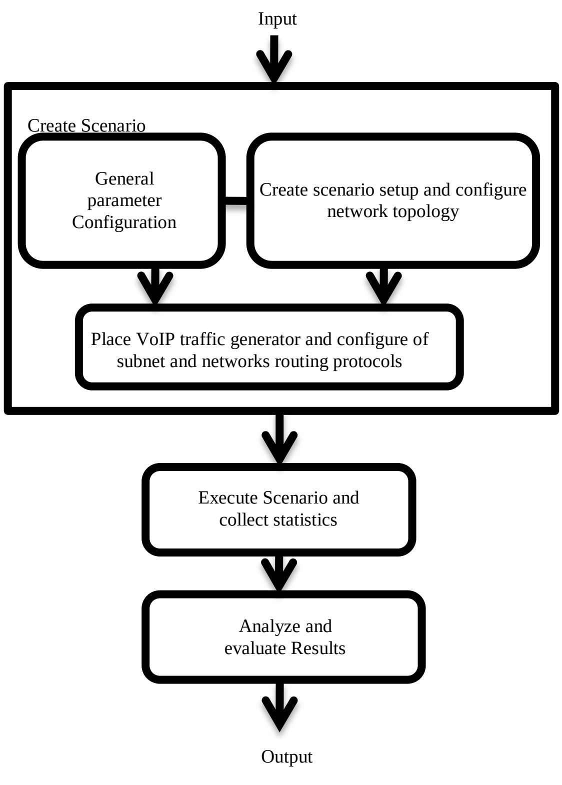

Figure 2

Qualnet is a network simulator which provides a comprehensive environment for developing protocols and analyzing the performance of network scenario (QualNet 6.1, 2013).

In order to analyze the performance of VOIP application over MANET we proposed framework approach.

Step1- First step involves the scenario creation which encapsulates the following stages-

a. Node configuration and general parameters- This includes configuration of parameters like simulation time, area, terrain as well as placement of nodes and connecting them to the wireless network through links.
b. VOIP Application- In the next stage VOIP applications are placed from node to node according to the requirement for transferring the multimedia data.
c. Configuration of subnet and network protocol- In this stage properties of wireless subnet are configured which encapsulates the configuration of each layer like PHY, MAC, NETWORK, and Routing protocol. Phy and Mac layer are set to 802.16(WIMAX),Network layer is set to IPV4 protocol and Bellman-ford(Raghavendra Ganig*a et al*,2012) as routing protocol.H.323 is set as multimedia signaling protocol in Application layer.
d. Channel Configuration- Channel configuration depends upon the no of base stations used connected through wired links. Then configuration will be changed through scenario properties. No of channel will be set via array editor and channel frequency (1.95-2.5) will be assigned to each channel individually.

Step2- Second step involves the creation of station in which particular nodes will be selected and their property will be changed to 'set as base station' in MAC layer. Then remaining nodes will be selected and their Mobility and Placement property will be set to Random way point.

Step3- This step includes the setting of scenario properties in which Battery model is enabled via statistics and tracking property. Then scenario is compiled and executed which encapsulates real time and execution time. This turns the simulator from design mode to visualization mode.

Step4- After the execution of scenario statistics of various metrics are collected from Analyzer. Statistics are the simulation result according to the layer properties.

Step5- This step specifies the result of different statistics of various scenarios with varying voice codec. Results with varying metrics values are compared that helps in concluding the best voice codec.

5. Simulation and Result Analysis

A. Simulation Setup

We have created two scenario's one for sparse network one for dense network and one for sparse network. The scenarios can be described as follows.

a. *Scenario 1:* The sparse network scenario is illustrated in figure 3. The 1000 x 1000 m^2 area is taken for creating the scenario in which 50 nodes are distributed over the area all connected through wireless link. Their node configuration and general properties are set. Three nodes connected with wired link are taken as base stations. Each node has own subnet connected with a group of nodes, The VoIP application are taken as traffic generator according to the requirement for transferring the multimedia data.

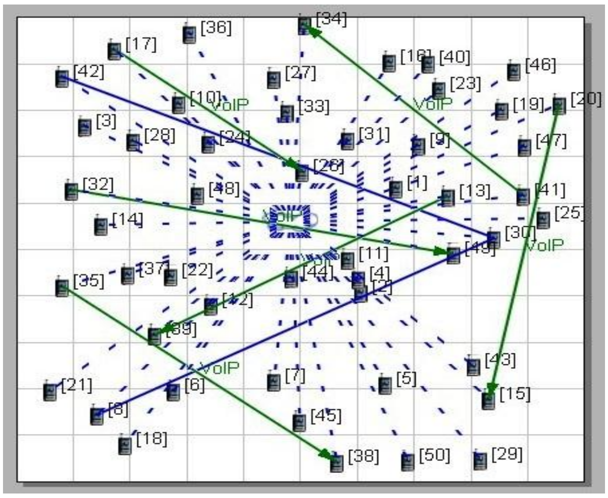

Figure 3

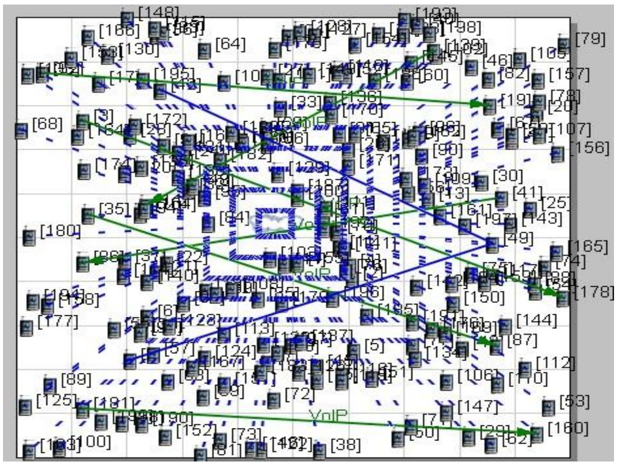

Figure 4

b. *Scenario2*: Dense network is described in figure 4. The same area 1000 x 1000 m^2 is taken for creating the scenario in which 200 nodes are distributed and three base stations are taken with their own subnets, each connected with a group of nodes. VoIP is used as traffic generator. The main parameters used in the simulation are given as follows:

Table 1: Simulation Parameters

Area	1000 X 1000 meter square
Simulation Time	300 sec
Bandwidth	20MHz
Transmission Power	Min: 20dBm, Max: 50dBm
Antenna Type	Omni directional
Traffic Source	VoIP
Physical Layer Protocol	802.16 Radio
MAC Layer Protocol	802.16

Four VoIP codecs are compared here: G.711, G.723.1ar6.3, G.728ar16 and G.729. These VoIP codecs one by one are evaluated in Qualnet 6.1 simulator tool. VoIP is use to simulates IP telephony sessions. We have taken VoIP traffic generator with 20 second average talking time and packetization of 20 millisecond interval. Then scenario is executed and information of statistics of various metrics are collected from Analyzer for the analysis of result.

B. Result Analysis

The performance of several voice codecs those are used in VOIP transmission over MANET are analyzed through different QoS metrics in small scale network and large scale network. The result analysis of the various metrics can be described as follows.

1. *Mean Opinion Score (MOS):*ITU-T P800 specifies MOS as a subjective metric which estimates the user satisfaction by means of a score which scales from 1.0 (poor) to 5.0(best). It is used to express the human opinion about QoS (ITU-T Recommendation,1996). So, for the communication network high value of MOS is considered.

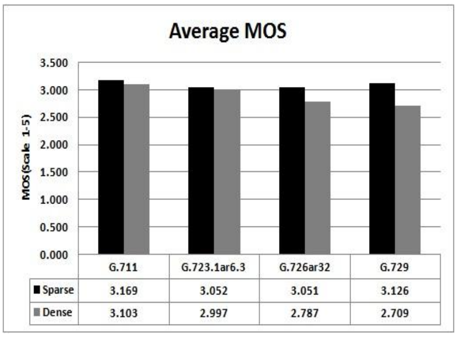

Figure 5

From the figure 5 it can be seen that sparse network has better MOS in MANET as compared to dense network. In case of codec G.711 has high MOS with value of 3.169 in both dense and sparse network.

2. *Throughput***:** It is defined as the average number of packets delivered successfully on communication network. It is measured in bits per sec. It specifies the number of messages that can be processed by a system in a given interval of time. So in any communication network high throughput is desired.

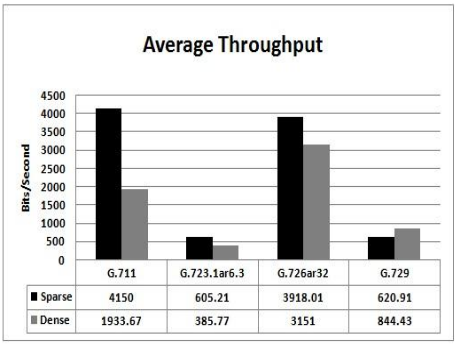

Figure 6

Figure 6 represents graph of throughput which shows that sparse network has high throughput. In comparison of codecs G.711 has high throughput in sparse network and G.726ar32 has high throughput in dense network.

3. *Average Jitter***-** It is calculated in seconds or in milliseconds. It is the undesired deviation caused by different data packets that can introduce undesired effects in audio signals and loss of transmitted data on reaching the destination. So low jitter is always desirable. Figure 7 represents the graph of average jitter which shows that dense network experience less jitter compared to sparse network. Codec G.729 has lowest jitter in dense network and codec G.711 has lowest value in sparse network.

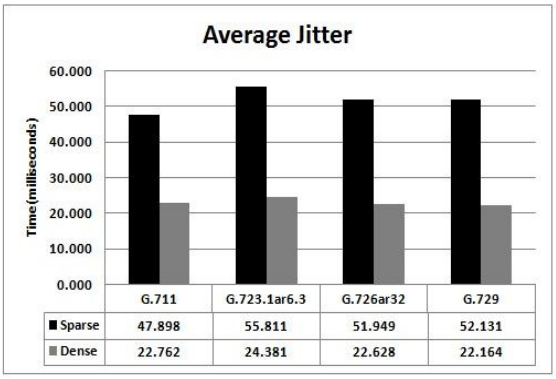

Figure7

4. *Average Delay*: It specifies the time taken by a bit of data to travel across the network from source to destination. It is usually measured in milliseconds. The average delay is classified into Processing Delay, Queuing Delay, Transmission Delay and Propagation Delay. For good network performance minimum delay is always desirable.

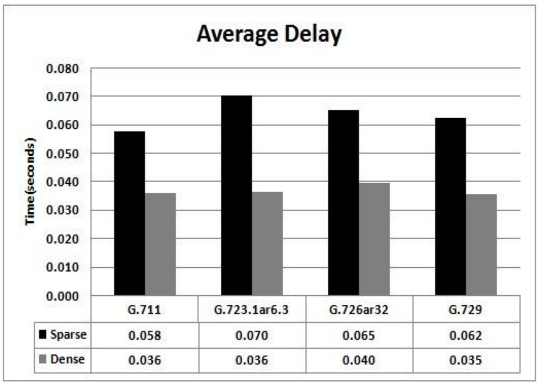

Figure 8

Figure 8 shows that dense network experience less delay compared to sparse network and codec G.729 has lowest delay in dense network and G.711 has lowest delay in sparse network. In sparse, mobility is more due to which packet consumes more delay and vice versa for dense networks.

5. *Energy Consumption*: It is the energy consumed by a system, total energy consumption is measured by summation of energy consumed in Transmit mode, Receive mode, idle mode and Sleep mode. It is calculated under physical layer and measuring unit is mWh.

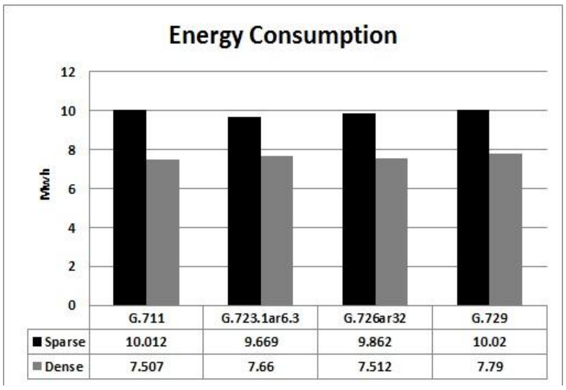

Figure 9

Figure 9 represents that graph of energy consumption in which dense network consumes less energy. In comparison of codec G.711 consumes lowest energy in dense network and G.723.1ar6.3 consumes lowest energy in sparse network.

6. *Signal Received With Error*: It is calculated under Physical layer and specifies the number of incoming signals that are failed to reach at the destination.

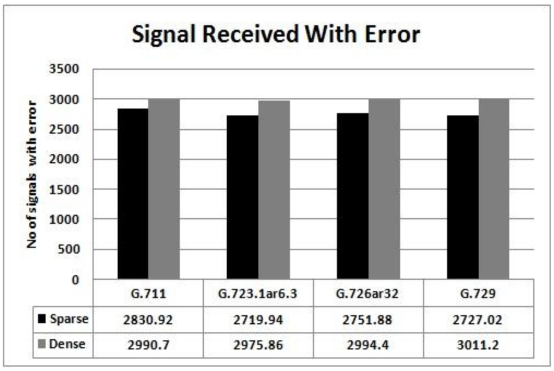

Figure 10

Figure 10 represents graph of signal received with error in which it can be seen that sparse network receives less error compared to dense network and codec G.723.1ar6.3receives lowest error in both sparse and dense network.

Conclusion

After the evaluation of the result we have concluded the performance of different CODEC's for VoIP with MANET in WiMAX network using H.323.From simulation results, we observed the performance with accordance to the dense and sparse scenario that *in terms of network*-sparse network is efficient to perform voice calls over MANET in case of average MOS, average throughput and signal received with error because of containing less traffic while there are three metrics average delay, average jitter and energy consumption which are in favor of dense network. In accordance to CODEC's, we concluded that codec G.711 performs best in case of average MOS, average throughput, average delay and energy consumption. While G.729 codec perform better in case of average jitter and G.723.1ar6.3 performs better in case of signal received with error.

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