Analysis of VoIP over Ad hoc Networks

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***Abstract*— Voice Over Internet (VOIP) is a rising popular Internet application that offers good services by Mobile Ad-hoc Network (MANET). MANET offers a appropriate platform for the deployment of multimedia and voice session over IP network in various application scenarios that provides safety to comfort related services. But this network also faces many issues on QoS because of packet drop, due to transmission errors, small battery life and dynamic changing configurations. WiMAX is a next generation wireless technology, presently utilizing by several institutions and industries which offers various benefits because of its large coverage area and high speed. Aggregating these techniques together, in this paper we measured the performance of several VOIP codec's with WiMAX in many scenarios (dense and sparse) over MANET. Voice codec's are formulated with some QoS metrics i.e. average jitter, average MOS, average delay, average throughput and signal obtained with error. The quality and performance of VOIP applications employing H.323 signaling protocol in Qualnet 6.1 modeler are analysed in order to find their impact on QoS and discover most effective network particular voice codec .**

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

I. INTRODUCTION

Wireless communication enables a subscriber to access the communication services from anywhere, at anytime all over the world. Currently user can move all over the world while preserving the connectivity with the rest of the world. This kind of communication is classified as Mobile Computing Network. Available mobile computing network can be categorized into – a) Mobile Adhoc network (MANETs), b) Infrastructure based network. MANETs are not centralized network containing mobile nodes fitted with wireless communication without any available infrastructure or any access point. This self configuring ability of MANET makes it appropriate for many conditions i.e. to set up a network on urgent basis, for military environment, for personal area networking i.e. cell phone, laptop, earphone, wrist watch etc, and for civilian environment i.e. taxi cab network, meeting rooms. MANET applications are generally related to voice transmission over IP network i.e. tele-emergency system that requires voice communication. Voice over IP is a very appropriate technology that permits the communication over packet switched network rather than circuit switched network. This provides the voice call employing internet telephony along with extra capabilities rather than analog telephone network. In order to assure the QoS on VOIP network a appropriate voice codec is needed. It is the selection of CODEC finds the quality of voice communication with restricted end to end delay and less packet drop rate. Main function of codec is to perform voice modulation (analog/digital signal conversion) and compression. The primary goal of this paper is to measure the performance of various voice codec's those are utilized in VOIP transmission over MANET.

II. RELATED WORK

(El Brak *et al*, 2012) has studied VOIP applications over VANET considering only urban scenario. (Said El Brak *et al*,2013) measured the speech quality for VOIP in WiFi network. (Francesca Martelli *et al,2012*) evaluated the VoIP performance over IEEE 802.11p in V2I network by using jitter and throughput. (S. Alshomrani *et al*, 2012)have talked about the QoS of VoIP over WiMAX but only three codecs are measured under small scale seven scenario. (M. Imran Tariq *et al*,2013) has examined the capability of WiMAX network, but VoIP performance related to throughput & delay was not examined. (Tucker *et al*,2006) talked about how WiMAX deals with the several factors that influences the network performance. The primary objective of this research work is to examine the performance VOIP in MANET in terms of QoS metrics. Several voice codecs are evaluated under the impact of dense and sparse scenarios through both user level (MOS) metrics and network level (such as loses).

III. VoIP over MANET

MANET offers a appropriate framework for the deployment of VOIP applications in various scenarios as the quality of service needs is a main issue for real time applications. This section offers a short overview of VOIP implementation, H.323 signaling protocol, voice codec's.

A. VOIP- presently user wants to interact by instant messaging, e-mail, video etc. in summation to the voice traffic. For this kind of multimedia interaction VOIP

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technology is most popular that permits communication over packet switched network between two parties. It is a technique that permits the transmission of multimedia sessions and voice over Internet that decreases the cost of large distance voice calls or sometimes provides free of cost service regardless of the distance (R.G.Cole *et al*,2009).Voice communication in a VOIP system can be shown by fig 1. At first, human voice continues in the analog form so before transmitting this voice over packet switched analog signal has to be modulated at the sender side and the reverse process (demodulation) is done at the receiver side. Modulation process consist sampling, quantization and encoding. The quality of voice communication over IP is primarily affected by the selection of voice codec.

B. *CODECs*- The application of voice codec process is the first step for voice communication. Its main function is to convert the incoming analog voice pattern into digital form and vice versa. International Telecommunication Unions (ITU-T) has developed several encoding methods which are categorized into waveform coders (eg-G.726, G.711), Vo-coders and Hybrid Coders (eg-G.723, G.728, G.729). As the number of mobile nodes changes quality of the voice also gets influenced. So the selection of codec is the important factor whose primary aim is to do voice digitization and compression assuring the lowest bit rate possible without decreasing the signal quality.

C. *Signaling protocol*- The encoded speech in the form of packets is when transmitted over IP network needs suitable signaling protocol i.e. H.323 or SIP (J. Rosenberg *et al*, 2002). In this paper we had utilized H.323 multimedia signaling protocol and WiMAX network. H.323 is responsible for encoding, decoding, signaling, packetizing & control as well as abilities exchange of audio and video signals.

IV. SIMULATION METHODOLOGY

For the measurement of VOIP applications performance over MANET, Qualnet simulator 6.1 is employed which find out the behavior of system in virtual computational world.
Qualnet is a network simulator which offers a appropriate environment for developing protocols and examining the performance of network scenario (QualNet 6.1, 2013).
In order to measure the performance of VOIP application over MANET we suggested framework method.

*Step1***-** First step includes the scenario generation which encapsulates the following phases-

a. **Node configuration and general parameters**: This involves configuration of parameters i.e. simulation area, time, terrain as well as position of nodes and associating them to the wireless network by connections.

b**. VOIP Application-** In the next phase VOIP applications are positioned from node to node according to the need for transferring the multimedia data packet.

c. **Configuration of subnet and network protocol-** In this phase, features of wireless subnet are organized which encapsulates the configuration of every layer i.e. PHY, MAC, NETWORK, and Routing protocol. PHY and MAC layer are adjusted to 802.16 (WIMAX), Network layer is adjusted to IPV4 protocol and Bellman-ford (Raghavendra Ganig*a et al*,2012) as routing protocol. H.323 is adjusted as multimedia signaling protocol at Application layer.

d. **Channel Configuration-** Channel configuration based upon the no of base stations utilized linked by wired connections. Then configuration will be varied by scenario properties. No of channel will be adjusted by array editor and channel frequency (1.95-2.5) will be allocated to every channel separately.

*Step2***-** Second step includes the generation of station in which specific nodes will be chosen and their property will be modified "set as base station" at MAC layer. Then left nodes will be chosen and their Mobility and Placement property will be adjusted to Random way point.

*Step3***-** This step involves the adjusting of scenario properties in which Battery model is enabled through tracking and statistics property. Then scenario is collected and performed which encapsulates execution time and real time. This turns the modeler from design to visualization mode.

*Step4***-** After the execution of scenario statistics of several metrics is compiled from examiner. Statistics are the modeling result according to the layer properties.

*Step5***-** This step describes the result of different statistics of several scenarios with changing voice codec. Results with changing metrics values are compared that supports in concluding the best voice codec.

V. SIMULATION AND RESULT ANALYSIS

*A. **Simulation Setup***

We have generated two scenario's one for dense network and one for sparse network. The scenarios can be specified as.
a. *Scenario 1:* The sparse network scenario is described in figure 2. The 1000 x 1000 m2 area is considered for generated the scenario in which 50 nodes are scattered over the area all nodes are associated by wireless connections. Their node topology and normal properties are adjusted. Three nodes connected with wired connections are considered as base stations. Every node has own subnet associated with a group of nodes, The VoIP application are considered as traffic generator according to the need for transferring the multimedia data.

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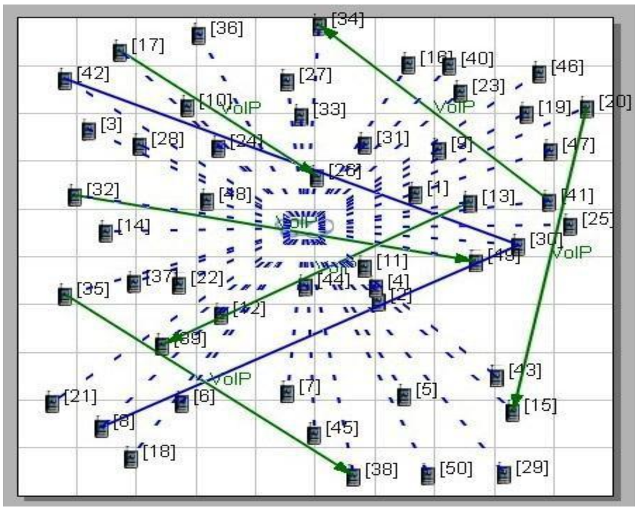

Figure 2 Sparse Network Scenario

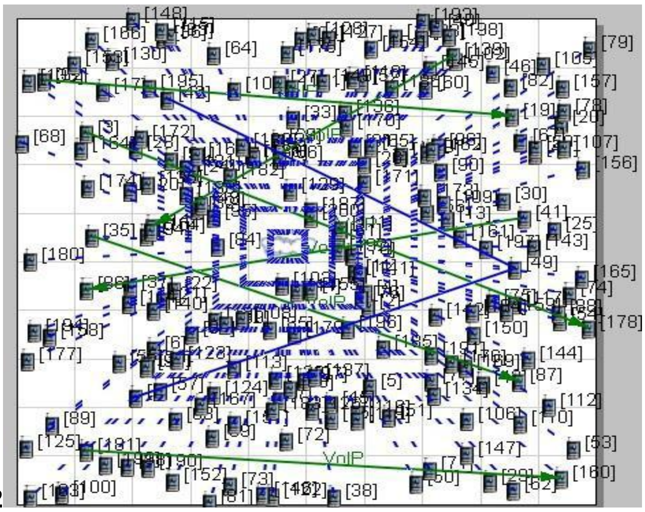

Figure 3 Dense Network Scenario

b. Scenario2: Dense network is specified in fig 3. The same area 1000 x 1000 sq m is considered for generating the scenario in which 200 nodes are scattered and three base stations are considered with their own subnets, each associated with a set of nodes. VoIP is employed as traffic generator. The main simulation parameters utilized in the simulation are described as follows:

Table 1: Simulation Parameter

Area	1000*1000 sq meter
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 codec's schemes are compared here: G.711, G.723.1ar6.3, G.728ar16 and G.729. These VoIP codecs one by one are measured in Qualnet 6.1 modeler tool. VoIP is utilized to analyze IP telephony sessions. We have considered VoIP traffic generator with 20 second mean talking time and packetization of 20 millisecond interval. Then scenario is performed and information of statistics of several metrics is compiled from Analyzer for the examining the result.

VI. RESULT ANALYSIS

The performance of various voice codec's those are utilized in VOIP transmission over MANET are examined by various QoS metrics in large scale network and small scale network. The result analysis of the different metrics can be specified as follows.

Mean Opinion Score (MOS):ITU-T P800 describes MOS as a subjective metric which evaluates the user satisfaction by means of a score which scales from 1.0 (poor) to 5.0(best). It is utilized to show the human view about QoS (ITU-T Recommendation,1996). So, for the communication network high value of MOS is taken.

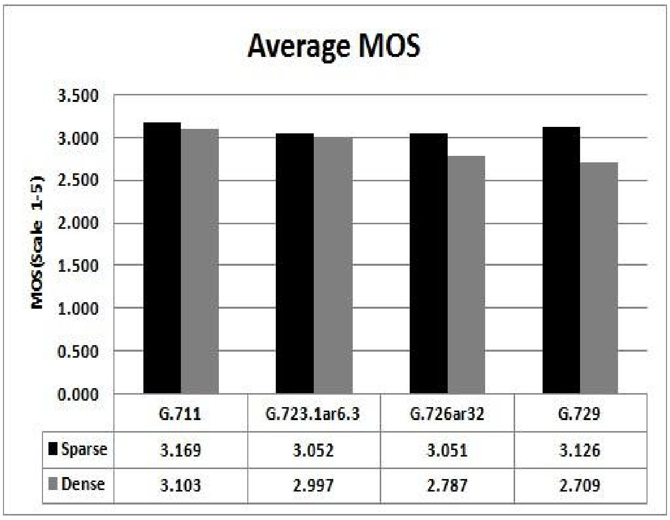

Figure 4 Average MOS for sparse and dense network

From the fig 4 it can be view that sparse network has good MOS in MANET in comparison of dense network. In case of codec G.711 has high MOS with value of 3.169 in both sparse and network.

Throughput: It can be defined as the average number of packets delivered successfully on communication network. It is evaluated in bits per sec. It describes the number of messages that can be treated by a system in a specified

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interval of time. So high throughput is required in any communication network.

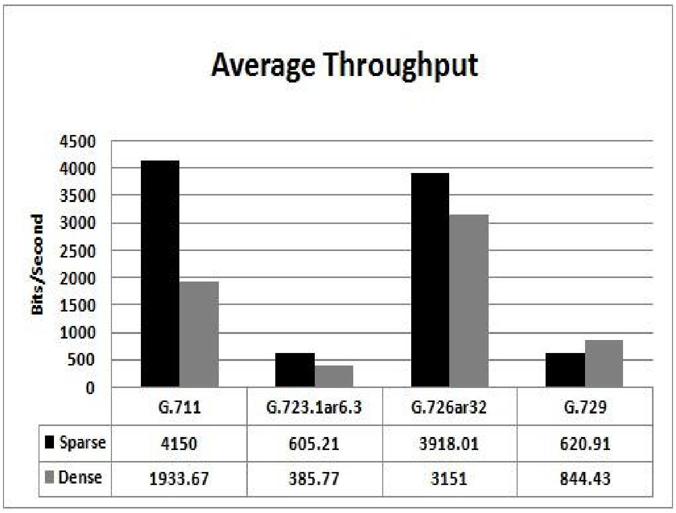

Figure 5 Average Throughput for sparse and dense network

Fig 5 shows graph of throughput which describes that sparse network has high throughput. Codec's G.711 has high throughput in sparse network while G.726ar32 has high throughput in dense network.

***Average Jitter*-** It is computed in seconds or in milliseconds. The deviation occurred by different data packets is not required that can present undesired impacts in audio signals and drop of transmitted data on arriving the destination node. So low jitter is always required. Fig 6 shows the graph of mean jitter which describes that dense network observes less jitter in comparison of sparse network. Codec G.711 has lowest jitter in sparse network and codec G.729 has lowest jitter in dense network.

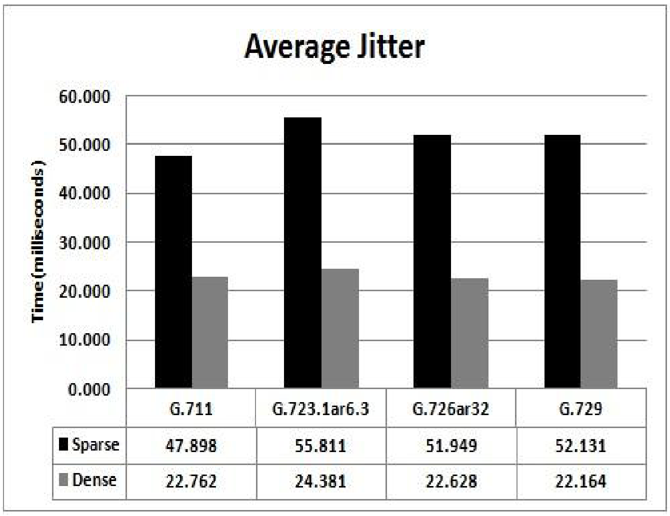

Figure 6 Average Jitter for sparse and dense network

***Average Delay*:** It describes the time considered by a bit of data to transmit all over the network from source node to destination node. It is normally evaluated in milliseconds. The mean delay is categorized into Queuing Delay, Processing Delay, Transmission Delay and Propagation Delay. For better network performance lowest delay is always required.

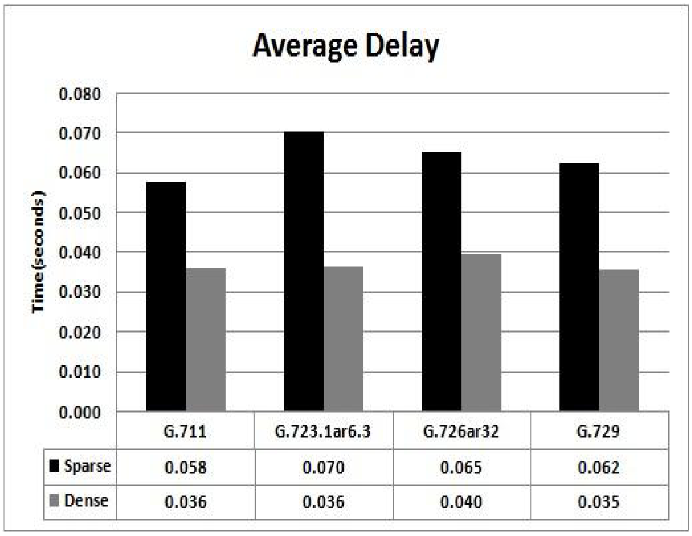

Figure 7 Average Delay for sparse and dense network

Fig 7 describes that dense network observes less delay in comparison of sparse network and codec G.711 has lowest delay in sparse network and G.729 has lowest delay in dense network. In sparse, mobility is more because of which packet consumes more delay and vice versa for dense networks.

Energy Consumption: The energy consumed by a system is known as energy consumption, total energy consumption is evaluated by sum of energy consumed in Receive mode, Transmit mode, sleep mode and idle mode. It is computed under physical layer and evaluating unit is mwh.

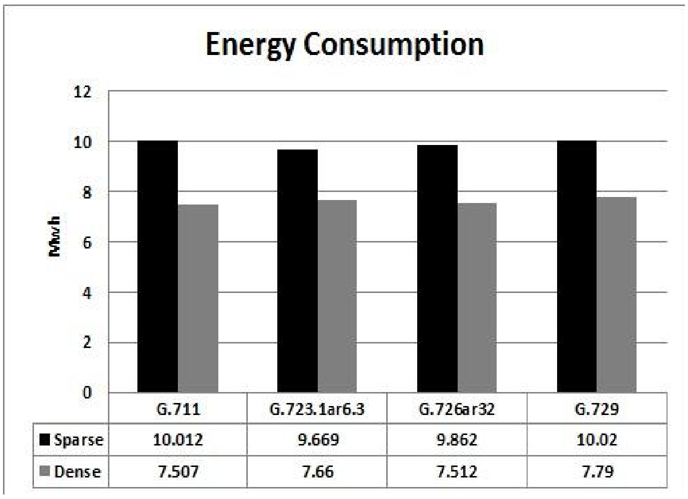

Figure 8 Energy Consumption for sparse and dense network

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Fig 8 shows that dense network consumes low energy. Codec G.723.1ar6.3 consumes lowest energy in sparse network and Codec G.711 consumes lowest energy in dense network and

***Signal Received With Error*:** It is computed under Physical layer and describes the number of incoming signals that are not capable to arrive at the destination.

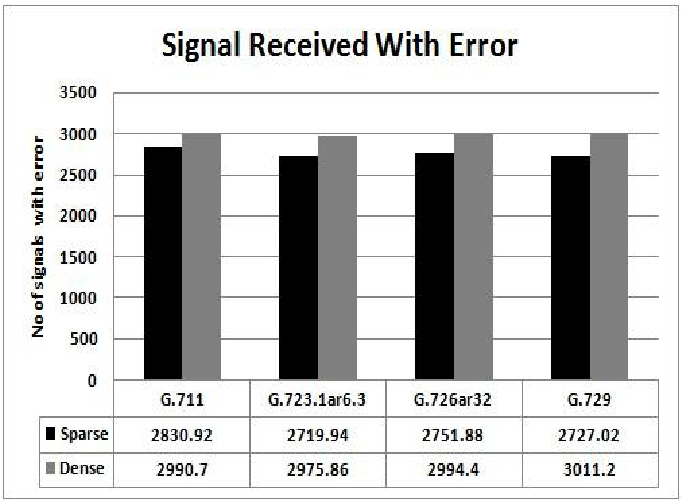

Figure 9 Signal Received With Error

Fig 9 shows graph of signal obtained with error in which it can be viewed that sparse network obtains less error in comparison of dense network and codec G.723.1ar6.3 obtains lowest error in both dense and sparse network.

CONCLUSION

After simulating the result we have concluded the performance of various CODEC"s for VoIP with MANET in WiMAX network utilizing H.323. From simulation results, we realized the performance of the sparse and dense scenario that in terms of network-sparse network is effective to perform voice calls over MANET in case of mean MOS, mean throughput and signal obtained with error due to containing low traffic while there are three metrics average jitter, average delay and energy consumption which are in support of dense network. According to CODEC"s, we observed that codec G.711 performs best in case of average throughput, average MOS, energy consumption and average delay. While G.729 codec has better performance in case of average jitter and G.723.1ar6.3 has better performance in case of signal obtained with error.

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