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APPRAISAL OF VoIP CODECS WITH WiMAX OVER VEHICULAR AD HOC NETWORKS

JAYA SINGH[1] & EKTA AGRAWAL[2]

[1]Department of Computer Science & IT, SHIATS, Allahabad, Uttar Pradesh, India

[2]Department of Computer Science and Engineering, UIT, Allahabad, Uttar Pradesh, India

ABSTRACT

Vehicular ad hoc network (VANET) is a very wide area of research due to various types of services it provides. It is considering as essential part of Intelligent Transportation System (ITS). It is a wireless technology that helps to improve the road safety and driving assistance. WiMAX is a 4G technology, currently using by various industries & institutions. WiMAX is a wireless telecommunication protocol which provides many advantages due to its high speed and large coverage area. VoIP is a methodology for voice communication & transmission of multimedia session over Internet Protocol (IP) networks. These three networks provides a high variety of services when combine with each other. But there are also many challenges that need to be pointed to provide a better quality of voice communication in any network. In this article, we estimate the performance of various VoIP codecs with WiMAX in different network conditions of Highway over VANET. There are some parameters such as End-to-End delay, average MOS, average jitter, throughput, average delay and signal received with error has taken to evaluate the performance and quality of voice connections and also helps to find out that which factor affect most the telecommunication media quality over VANET.

KEYWORDS: VANET, Wi MAX, Codec, IP, Bellman-Ford

1. INTRODUCTION

VANET is a wireless communication technology, which is considered as an extension of MANET [1]. Intelligent Transportation System (ITS) provides a set of standard for VANET to develop intelligent vehicular network in future [2]. In VANET, the moving vehicles are assumes as mobile nodes or routers in wireless networks to create a mobile network. It allows vehicles to communicate and create a network with wide range. VANET is useful for enhancing driving safety, providing many applications like vision enhancement, weather warning, collision avoidance, driver assistance, online gaming, infotainment etc. The main characteristics of VANET are high dynamic topology, unlimited battery power, sufficient bandwidth, high storage capacity, satellite navigation system, frequent disconnected network, communication, environment etc [3]. Voice communication is an important part of many applications of VANET.

WiMAX was developed by WiMAX forum in June 2001. WiMAX stands for Worldwide Interoperability for Microwave Access, which is a wireless communication standard. It comes under the IEEE 802.16 family, which provides Broadband Wireless Access (BWA), upto 30 miles for fixed stations and 3-10 miles for mobile stations [4]. The high bandwidth and speed of WiMAX makes it suitable for many services such as providing mobile broadband connectivity over a large geographical area, providing data, telecommunication and multimedia services. There are mainly two devices are used by WiMAX network to provide internet connectivity: Subscriber Stations & Base Stations. The Base Stations provide wireless coverage over a specific area, which is also known as cell. It is similar to the concept of cell phone towers. Any wireless device within coverage area would be able to access the internet while the subscriber stations are works as receiver, which gets services from base stations. WiMAX uses OFDMA (Orthogonal Frequency Division

Multiple Access) at physical layer, which is use to encode digital data on multiple carrier frequencies [5].

VoIP stands for Voice over Internet Protocol, which is a telecommunication technique to manage the delivery of voice information over the internetwork [6]. The VoIP is use to send voice information by discrete packets of digital form. The major benefit of VoIP is that it avoids the toll charges by ordinary telephone services. Currently, VoIP with WiMAX over VANET is emerging as infrastructure network, which provides telecommunication services over wireless broadband with reliability and cost effective way. Yet, it is facing many problems and difficulties due to many factors of VANET such as high velocity of vehicles, fast changing topology, frequent disconnections, security problem and limited communication range, which abate the performance and quality of Voice communication.

The purpose of this estimation is to test the performance of different VoIP codecs in WiMAX network with different traffic conditions of VANET with the help of QualNet simulation tool.

The rest of the paper is organised as follows: the Related Work is discussed in section II. In the section III, we explain the evaluation model & simulation tool. The simulation and result analysis of VoIP codecs with WiMAX are discussed in section IV. Finally, we conclude our work in Section V.

2. BACKGROUND AND RELATED WORK

The performance of VoIP applications using various networks and technologies have been pointed in this section. There are many research works have been done to analyze the VoIP performance in WiMAX network using Vehicular networks. Adhicandra [7] has find out that data transferring and telecommunication can be done in WiMAX networks. Tucker [8] discussed the various factors which affect the network performance and also find out that how WiMAX deals with them. Pentikousis et. al. [9] has taken a fixed WiMAX network for estimating the performance of VoIP in terms of MOS, packet rate, cumulative good put and sample loss rate. Imran Tariq et al. [10] presented the measurement of the capacity of WiMAX network, but they didn’t evaluate the VoIP performance regarding throughput & delay. Scalabrino et al. [11] focused on VoIP performance using test beds over WiMAX. Alshomrani e. al.[12] also discussed the QoS of VoIP over WiMAX but they all didn’t done these works in VANET. Although Martelli et. al. [13] measured the VoIP performance over IEEE 802.11p Vehicular network, but they didn’t used V2I networks to measure its performance.

3. EVALUATION MODEL AND SYSTEM DISCRIPTION

There are following evaluation model has proposed to measure the performance of different codecs of VoIP with WiMAX over VANET:

- **Routing Protocol**

We have used Bellman-Ford routing protocol for our simulation [14]. It is used as distance vector routing protocol. It is also known as Ford-Fulkerson Algorithm, in which every router has to maintain a distance table, which gives the information of distance of nodes in network and provides shortest path to send packet to each node. The information of table is update every time, when the information is exchange with neighbour nodes.

- **H.323 Signaling Protocol**

We have used H.323 protocol for multimedia signaling protocol at application layer of our network scenario. H.323 protocol was the first standard based VoIP technology which was introduced in May 1996 by ITU-T. It defines the protocols to provide audio-visual communication sessions on any packet network. The H.323 standard addresses call signaling and control, multimedia transport and control, and bandwidth control for point-to-point and multi-point conferences [15] [16].

- **Simulation Tool Description**

We use QualNet 6.1 simulation tool to evaluate the quality and performance of video and services [17]. It is a network simulator which examines the behaviour of system in virtual computational world. QualNet is a comprehensive suit for modelling large wired and wireless network. It is composed of the following tools: Architecture, Analyzer, Packet Tracer, File Editor and Command Line Interface. It provides an absolute environment for designing protocols, creating and animating network scenarios and analysing their performance. We have proposed an approach for this work. It is given as follows:

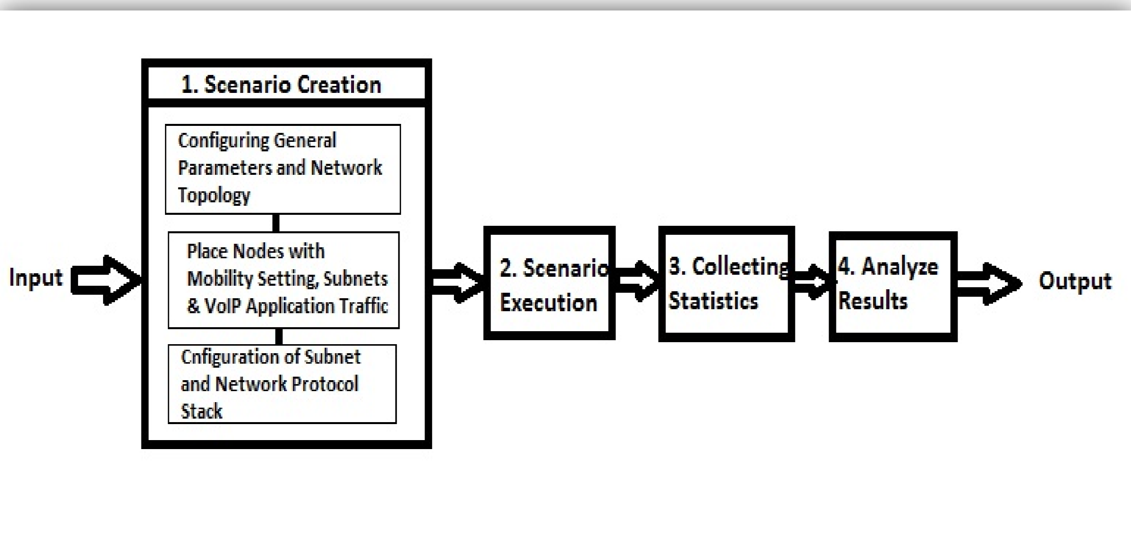


Figure 1: QualNet Simulation Framework for Proposed Approach

The simulation framework consist four steps:

Step 1: First step explains the scenario creation, which consist three parts. The first part describes the configuration of general parameters like area, simulation time, terrain etc.

Second part include node placement with setting mobility according to need, placing wired and wireless subnets and connected them to mobile nodes through links. Then set VoIP applications over the network from node to node.

In third part, the configuration of subnet and network protocol setting is done. The subnet are configure according to protocol stack. It also include channel configuration, which depend on the number of base stations we are using in our network. Then the listening and listenable channels are set at physical layer in accordance with channelization. Then physical layer protocol & MAC layer protocol is set to 802.16 & network layer set to IPV4 protocols. After it, set Bellman-ford as routing protocol. In application layer, the H.323 is set as multimedia signaling protocol.

Step 2: Second step is scenario execution, which contains the visualization part of QualNet, which includes simulation compilation and scenario execution. It shows both, simulation time & real time at the time of scenario running.

Step 3: The third step contains the collections of statistics, which has came after execution of scenario. The statistics collection are gives statistics results according to the layers.

Step 4: The last step include analysis of results. This step helps us to evaluate the results of different statistics files of various scenarios, and conclude the output.

4. SIMULATION AND RESULT ANALYSIS

- **Simulation Setup**

According to the approach, we have created two scenarios for two types of networks conditions: Sparse and Dense network. The scenarios are as follows:

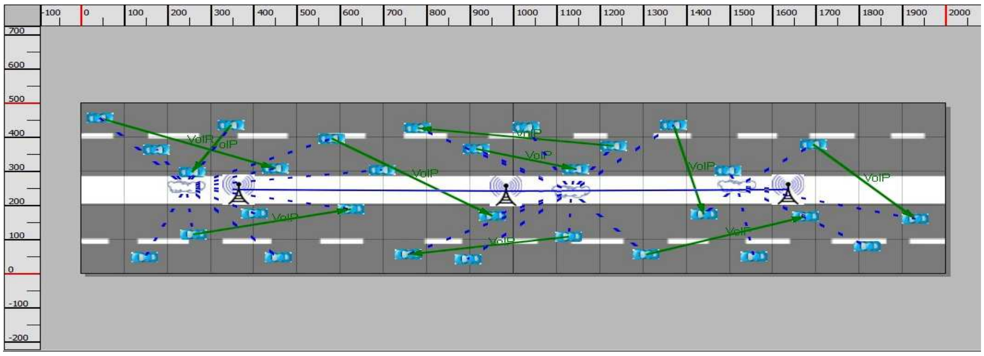


Figure 2: Highway Sparse Network Scenario

Scenario 1: The Highway sparse network scenario is illustrated in figure 2. The 2000 x 500 m^2 area has taken for creating the two lane highway in this scenario. It shows the distribution of 30 vehicles over 2km highway with three base stations, each has its own subnet connected with a group of vehicles. The VoIP application has taken as traffic generator.

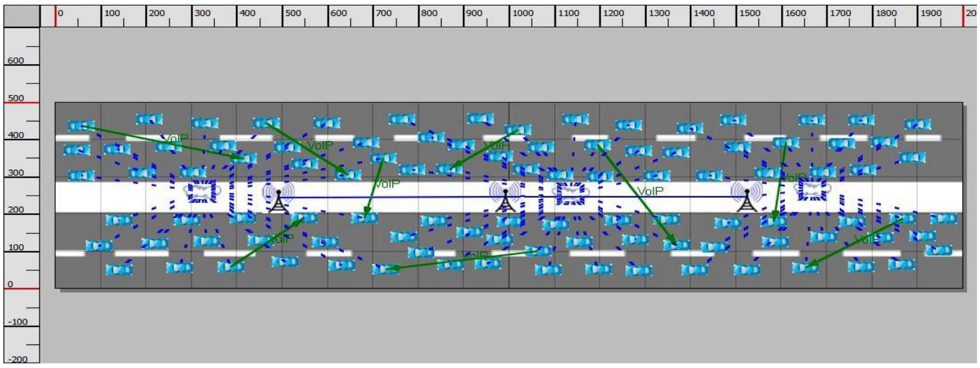


Figure 3: Highway Dense Network Scenario

Scenario 2: The 100 vehicles are distributed over 2km highway in dense network as shown in figure 3. There are 2000 x 500 m^2 area has been taken to create two lane highway in the scenario and three base stations with their own subnets, each connected with a group of vehicles are used in the scenario and VoIP has used as traffic generator. The main parameters used in the simulation are given as follows:

Table 1: Simulation Parameters

Area	2000 X 500 Meter Square
Simulation Time	240 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

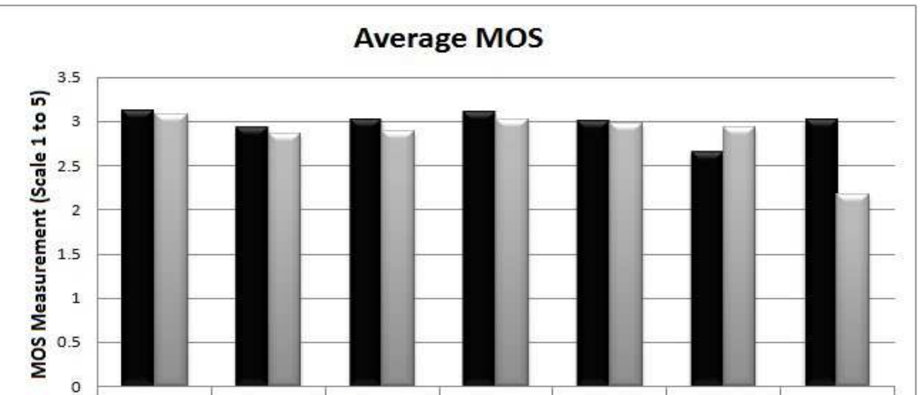
The Qual Net simulation tool, as explained in section 3, evaluates the VoIP codecs one by one in these scenarios. There are seven VoIP codecs has compared here: G.711, G.723.1ar 5.3, G.723.1ar 6.3, G.726ar 24, G.726ar 32, G.728ar 16 and G.729. VoIP is use to simulates IP telephony sessions. We have taken VoIP traffic generator with 20 second average talking time and packetization of 20 millisecond interval. After execution of scenarios, we record the information of network with collecting statistics for analysis of result.

- **Result Analysis**

The result analysis of following metrics is given as below:

- **Mean Opinion Score (MOS)**

MOS follows the measurement techniques specified by ITU-T P.800, in which, various people are made to listen the voice signals and rate the factors like distortion, delay, echo; noise etc on a scale of 1 to 5 where 1 is the minimum and 5 is the maximum. So, high value of MOS consider as better for the communication network.



	G.711	G.723.1ar 5.3	G.723.1ar 6.3	G.726 ar 24	G.726 ar 32	G.728 ar 16	G.729
SPARSE	3.12545	2.929928571	3.020861429	3.10692	3.0022	2.650084286	3.013925714
DENSE	3.07779	2.8545525	2.8899475	3.015675	2.97189	2.93769	2.1728875

Figure 4: Graph for Average MOS

The graph of figure 4 shows that sparse network of VANET has better MOS as compared to dense network. So, it is clear that sparse network performs well in case of MOS. According to codecs, the graph shows that G.711 codec is best in both, sparse & dense network.

- **End-to-End Delay**

The end-to-end delay is the amount of time taken for a packet to be transmitted across the communication network from the source to destination. The small end-to-end delay is good for any communication network. It is calculated as:

$$D_{\text{end-end}} = N\,[\,D_{\text{trans}} + D_{\text{prop}} + D_{\text{proc}} + D_{\text{que}}]$$

Where,

$D_{\text{end-end}}$ – end-to-end delay

D_{trans} – transmission delay

D_{prop} – propagation delay

D_{proc} – processing delay

D_{que} – queuing delay

N – Number of links (no. of routers + 1)

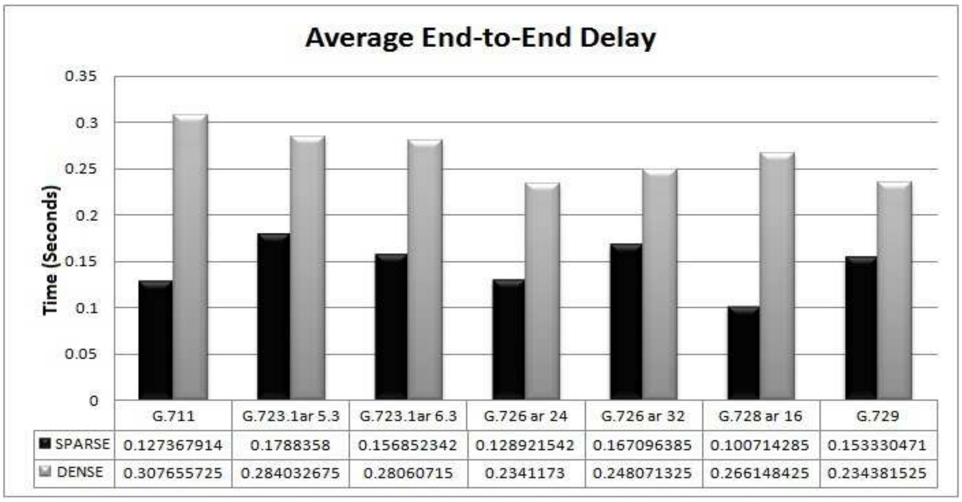


Figure 5: Graph for Average End-to-End delay

The graph of figure 5 presents that sparse network has less end-to-end delay as compared to dense network and in comparison of codecs; G.728ar 16 has smallest value in sparse network, while G.726ar 24 has smallest value in dense network.

- **Throughput**

Throughput is the measurement of the number of messages that a system can process in a given amount of time or it is the average rate of the delivery of successful message over a communication channel. In communication network, the throughput is usually measured in bits per second or transaction per seconds. The throughput can be analyzed mathematically by means of queuing theory, where the load in packet per time unit is denoted by arrival rate λ and the throughput in packet per time unit is denoted by the departure rate μ. High throughput is always acceptable in a communication network.

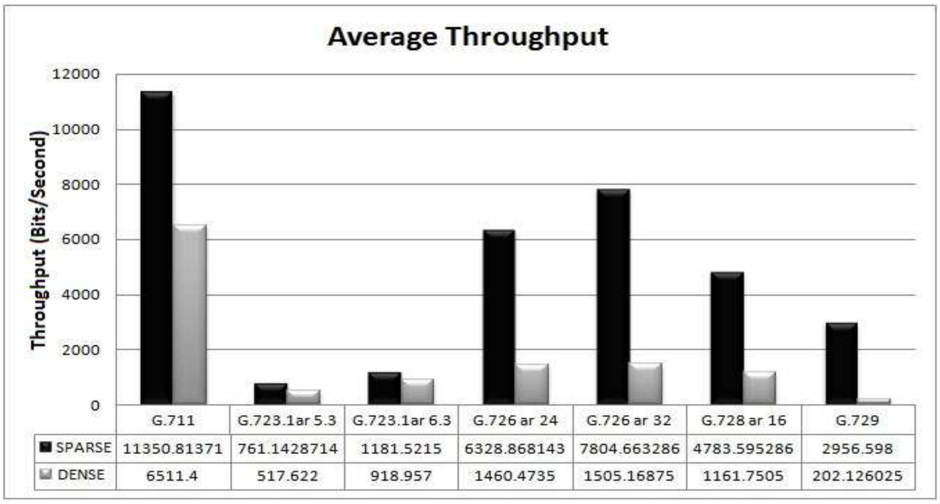


Figure 6: Graph for Average Throughput

The figure 6 present the graph of average throughput, which shows that sparse network has high throughput as compared to dense network & G.711 codec performs best in both networks.

- **Average Jitter**

Jitter is the amount of variation in latency or response time, which is calculated in seconds or most probably in milliseconds. Jitter shows up as different symptoms, which is depending on the application using by us. Web browsing is fairly resistant to jitter, but any kind of streaming media is quite susceptible to jitter. Less value of jitter is always desirable in a communication network.

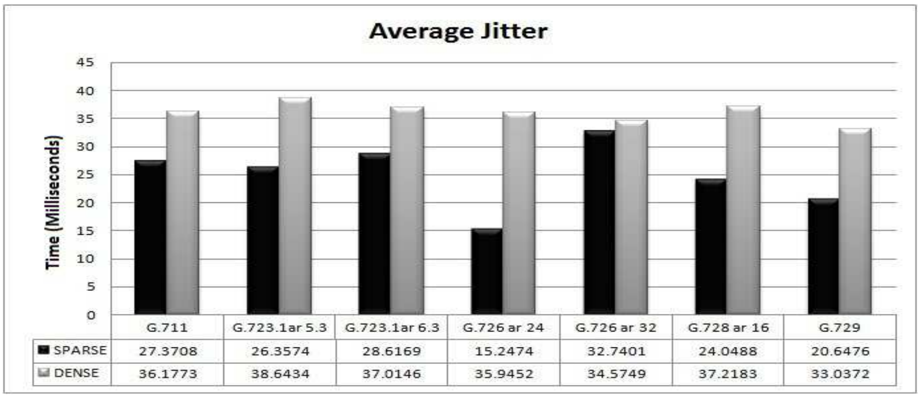


Figure 7: Graph for Average Jitter

The graph of figure 7 shows the evaluation of average jitter, which present that sparse network has less jitter as compared to dense network. In accordance with codecs, G.726ar 24 has lowest value of jitter in sparse network and G.729 has lowest jitter in dense network.

- **Average Delay**

The average delay of any network specifies the average of time taken by a network for a bit of data to transmit across the network from sender to receiver. It is typically uses multiple or fraction of seconds as measuring unit under the Network layer. The average delay is divided into the following parts: Processing Delay, Queuing Delay, Transmission Delay and Propagation Delay. Minimum value of delay is always requiring for good performance of network.

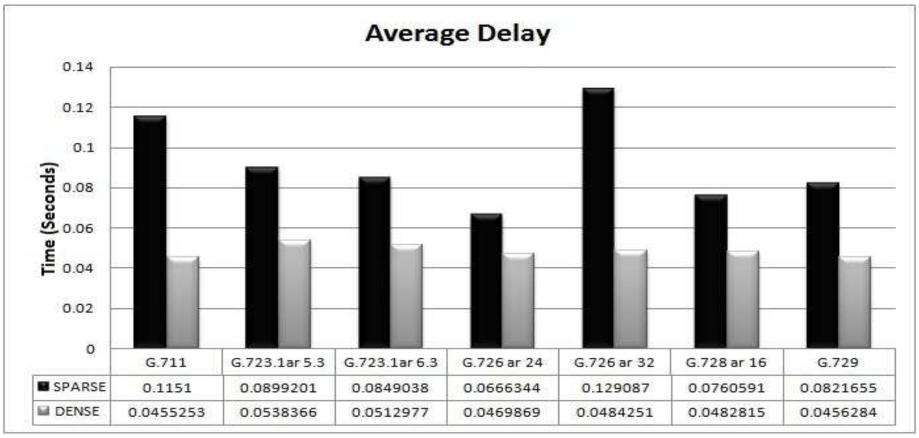


Figure 8: Graph for Average Delay

The figure 8 presents the graph of average delay, which shows that dense network has the less values as compared to sparse network. So, here dense network performs better. In case of codecs, G.711 has the lowest value of delay in dense network, while G.726ar 24 has lowest delay in sparse network.

- **Signal Received With Error**

It refers to the number of incoming signals that failed to receive by the radio channel. The value of this metrics is calculated under the physical layer.

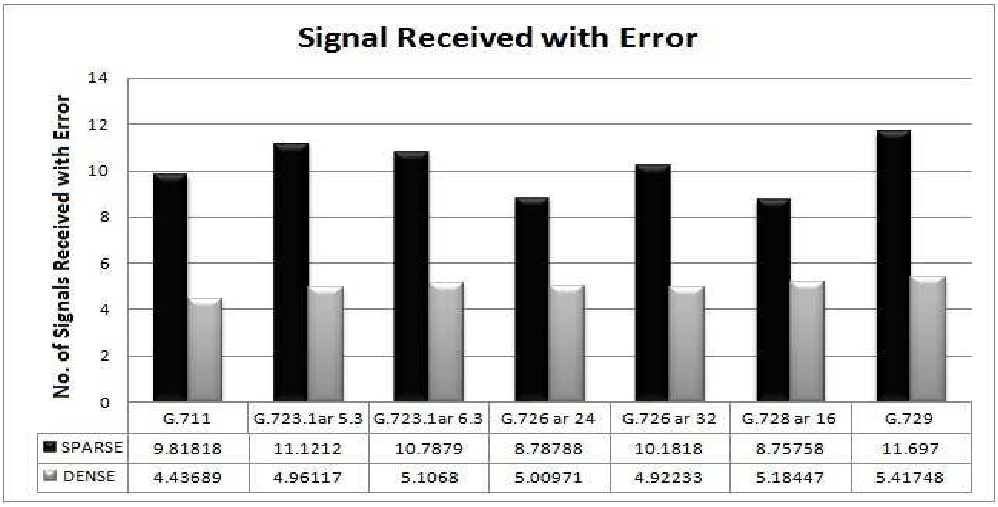


Figure 9: Graph for Signal Received with Error

From the graph of figure 9, it is clear that sparse network received more signals with error as compared to dense network, so here; again dense network performs better than sparse network. In accordance with codecs, G.711 has lowest signal received with error in dense network while G.728ar 16 has the lowest signal received with error in sparse network.

5. CONCLUSIONS

After the evaluation of the result, we concluded that mostly results are in favour of sparse vehicular network. So, sparse network is more efficient to perform voice calls over VANET as it gives best performance in case of average MOS, average end-to-end Delay, average throughput and average jitter because of containing less traffic as compared to dense network, while there are two metrics: average delay and signal received with errors, which are comes in favour of dense network. If we notice the results in accordance with codecs, then we get that codec G.711 performs best in case of average MOS, average throughput, average delay and signal received with error while in case of average end-to-end delay, G.728ar 16 codec perform better and G.726ar 24 codec performs best in case of average jitter.

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