International Journal of Current Engineering and Technology
ISSN 2277 - 4106

Research Article

Performance Analysis of Different Voice CODECs in Integrated VANET-UMTS Wireless Network by using H.323

Kalpana Gurung[Ȧ*] and Hari Mohan Singh[Ȧ]

[Ȧ]Department of Computer Science & IT, SHIATS, Naini,Allahabad, India

Accepted 05 November 2013, Available online 01 December 2013, **Vol.3, No.5 (December 2013)**

Abstract

Today's smart vehicles are embedded with computer, GPS receivers, short range wireless network interfaces make capable vehicles to communicate with their neighbour vehicles. Thus, they form a self organizing, self managing with low bandwidth network. This type of network called vehicular ad hoc network (VANET). VANET is a wireless communication area in which vehicles acts both as node and router. VANET provides variety of interesting applications ranging from safety to comfort levels. Therefore, it is one of the influencing areas for the improvement of Intelligent Transport System (ITS) in order to provide real time messages to the occupant of vehicles via communication. In this paper, our study is based on inter-vehicle communication over VANET. For this task, various voice CODECs behaviour was tested and analyzed the impact of varying traffic condition on the performance of QoS of VoIP. We designed simulation modules by using simulator QualNet6.1 and carried out extreme simulations. Results of simulations are presented in the terms of average jitter, average end-to-end delay, average throughput, average delay and mean opinion score (MOS).

Keywords: *VANET, VoIP, H.323, UMTS, CODECs, QualNet6.1.*

1. Introduction

Mobile ad hoc network (MANET) is an autonomous network consisting mobile nodes, which are communicates through wireless network without any access point, centralized administrator or any existing infrastructure (Ming Yu Jiang). MANET was originally used in military projects, increasing in tactical networks and Defence Advanced Research Projects Agency (DARPA). But now days, the existence of such Network supports various applications such as conferencing, emergency operations (such as natural disaster, terrorist attack, etc.), wireless sensor networks and inter vehicle communication (VANET) etc.

VANETs is one special type of MANET, such network composed group of moving vehicles that are exhaustively capable of providing communication among nearby vehicles and the roadside infrastructure. Vehicles are equipped with computing devices, antennas and GPS receiver making VANET realizable. VANET includes two types of communication (Mohammad Azouqa *et al*). First, Vehicle-to-Vehicle (V2V) communication, this helps in exchanging the traffic related information to make drivers better aware of their surroundings. In case of emergency, event-driven messages can be generate and disseminate to vehicles in the zone of danger. Other is, Vehicle-to-Infrastructure (V2I) communication, it is concerned about transmitting the information between vehicles and the fixed road side units (RSU). Such RSUs include gateways or base station. They provide infotainment services like on-road advertisement, accessing e mails, automatic tolling, streaming of audio and videos. Some characteristics of VNAET that distinguish it from other type of ad hoc networks are no power limitations, vehicles are aware of their position and regular movement restricted by both road topologies and traffic rules.

Voice over Internet Protocol (VoIP) is a methodology and group of technologies that are capable of transmitting voice communication and multimedia sessions through Internet connection. VoIP reduces call charges for long distance voice calls or may offer the service free of charge regardless of the distance (Ismail Dalgic *et al*, 1999). The quality of voice communication over an IP network based on the suitable voice CODECs (Hong Xiong *et al*, 2003).

With the deployment of wireless technology like Universal Mobile Telecommunication System (UMTS), a part of 3G network, high- quality service and higher downlink/uplink speed can provide to user in anywhere anytime. It also provides wider coverage area (William Lehr *et al*, 2003).The scope of this paper concerns the characterization and analysis of VoIP QoS over VANET with UMTS under varying traffic conditions of city and highway scenarios. For the implementation of this task, QualNet6.1 simulator was used to run several simulations. VoIP sources were employed according to city and highway scenarios to generate voice traffic.

*Corresponding author: **Kalpana Gurung**

The rest of this paper is organised as follows. Section II gives brief background of VoIP, H.323 signalling protocol, UMTS-3G wireless cellular network and related work. Section III presents the simulation methodology Section IV discusses the simulations and results. Finally, conclusions are given in section V.

2. Background & Related Work

In this section, we begin with a brief overview of VoIP implementation, H.323 signaling protocol and UMTS. At last, brief of review of some literatures that have been worked in this field are also discussed.

(A)VoIP over VANET- The speech source alternates between talking and silence period which is typically considered to be exponentially distributed. As illustrated in fig.1 at the sender's side, when a call or voice initiate over IP network first it converted from analog into digital signal through digitization process, it composed of sampling, quantization and encoding. After that those digital signals are packetized with equal size and then transmit it over IP network appended with source and destination network address. At the receiver's side, the reverse process is performed.

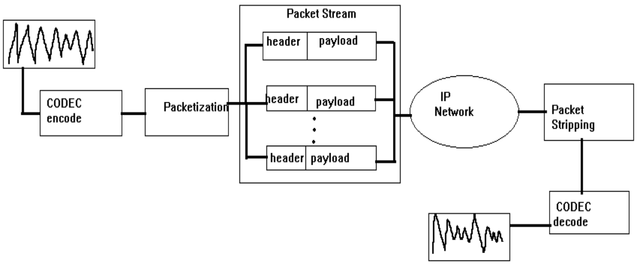

Fig.1 VoIP system

Table I show some of the commonly used ITU-T standard CODECs, and list their attributed. Generally, the primary function of CODECs is converts the incoming analog voice pattern into digital stream and converting that digital stream back to analog voice pattern at the ultimate destination.

Table I: CODECs Description

CODEC	Bit Rate (kbps)	Sample size (bytes)	Packets per second	Payload size (bytes)	License
G.711	64	80	50	160	Open source
G.729	8	10	50	20	Patented
G.729	6.3	24	33.3	24	Proprietary
	5.3	20	33.3	20	
G.726	32	20	50	80	Open source
	24	15	50	60	

(B) H.323 Protocol- It is an ITU-T's recommended standard protocol for signalling and call control of IP telephony (James Toga *et al*, 1999). It describes terminals, equipments and services for multimedia communication over packet based network. The H.323 protocol is a tightly coupled of sub protocol (Markku Korpi *et al*, 1999) that are responsible for encoding, decoding and packetization audio and video signal as well as negotiate the capabilities.

According to (Asoke Talukder) a H.323 system comprises the following entities: terminal. Gateway, gatekeeper and multipoint control unit (MCU). A terminal may be a multimedia PC, video telephone, IP phone that provide real time, two-way communication in point-to-point or multipoint conferences. Terminals are required to support H.245, Q.931, RAS (Registration Admission Status) and RTP (Real time Protocol). Gateway provides the connectivity between packet switched network and circuit switched network through the translation between transmission formats, translation between audio and video CODECs, call set up and call clearing on both network. Gatekeeper is the central point for all calls within its zone for all registered end points provide services like address translation, admission control, call signalling, call authorization, bandwidth management. MCU supports multipoint conference between three or more terminals and gateways.

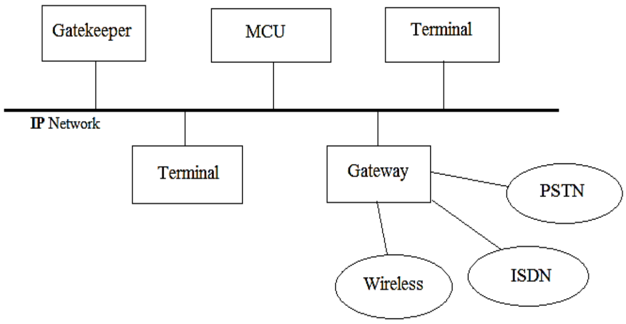

Fig. 2 H.323 component

In H.323, a call session is perceived as consisting of five phases:

1. Call Setup
2. Initial Communication and Capability Exchange
3. Establishment of Audio Visual Communication
4. Call Services
5. Call Termination

H.323 makes use of three different signaling channels in order to complete these five phases. These are:

(1) H.225.0 RAS Channel- It defines the mechanism for communication between an endpoint and its gatekeeper. The functions provided by this channel are gatekeeper discovery, endpoint registration, endpoint location and admission, bandwidth change, status and disengage. These functions are done over an unreliable transport protocol such as UDP.

(2)H.225.0 Call Signaling- This channel carries information related to call control and supplementary service control. The initial messages exchange between two endpoints by using the gatekeeper's RAS transport

address. The information carries by using the protocol like H.225.0 and H.450.x. the call signalling is done *over TCP (reliable channel).*

(3)H.245 control Channel- H.323 system utilizes this channel after the completion of call establishment phases. This channel carries the H.245 protocol messages for media control with capability exchange support.

(4) Logical channel for Media- This channel carry the audio, video and other media information. Each media type is carried in a separate uni-directional channels, one for each direction, using RTP/RTCP.

(C) UMTS: UMTS is proposed to converge packet switched and circuit switched network. It provides much greater level of functionality and flexibility than previous generation and high mobile data ranging from 144kbps to 2mbps. With availability of UMTS, users can access different wireless network. One example is Fring (D.Shinder *et al*, 2007) that uses VoIP over UMTS.

As illustrated in fig.3, the designed UMTS simulation module consists of core network (CN), UMTS territorial radio access network (UTRAN) and user equipment (UE). CN and UTRAN collectively called as Public Land Mobile Network (PLMN). A UE can be a mobile handset, laptop, desktop or any device can provide access to the network. CN provides routing, switching and network management. UTRAN provides radio access to UEs. The radio interface for UE and CN are provided by UTRAN (Harri Holma *et al*).

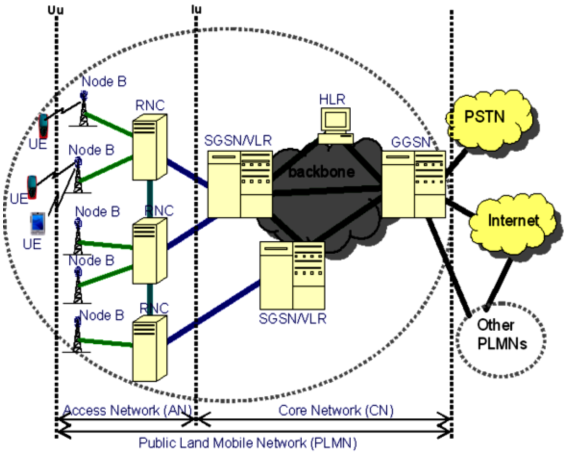

Fig. 3 UMTS model

Related Work : The authors in (Sheetal Jadhav *et al*, 2011) conducted simulation study to evaluate the QoS performance of WiMAX and UMTS for supporting VoIP traffic. The result of simulation showed that WiMAX outscores the UMTS with a sufficient margin and is better technology to support VoIP applications compared with UMTS. The work in (Said El Brak *et al*, 2013) examines the performance of various voice CODECs on inter-vehicle voice streaming rely on multi-hop fashion by the mean of simulations. Also test the impact of network environment on QoS metrics and analyzed that G.723.1 CODEC worked well in the urban VANET environment. In paper (Sajal K. Das*et al*, 2003), authors analyzed the performance of the H.323 call set up procedures over the wireless link by using two call modes of operation for this analysis namely regular and fast connect. They proposed a model with assumption of a transport radio link protocol (RLP) without RLP retransmission used for VoIP packets while non-transparent (regular) RLP is used for H.323 control Packets. This model proved that VoIP call setup performance can degrade significantly even for moderately high air-link frame error rates (FERs). Furthermore, the call setup performance can be improved significantly using the robust radio link layer such as RLP with the modifications. The authors in (Abderrahim Benslimaneal, 2011) envisioned an integrate network architecture of VANET and UMTS by forming clusters of minimum number of vehicles equipped with IEEE 802.11p and UTRAN RF (air) interfaces, are selected as vehicular gateways to link VANETS to UMTS. Issues pertaining to gateway selection, gateway advertisement and discovery, service migration between gateways (i.e., when serving gateways lose their optimality) are all addressed and an adaptive mobile gateway management mechanism is proposed.

3. Simulation Methodology

In order to evaluate the performance of QoS of VoIP over VANET by implementing the technologies described above, we choose the simulator QualNet in its version 6.1 to design mobility model. As discussed earlier that the performance of VoIP depends upon the suitable voice CODEC so different voice CODECs are deployed systematically both in city and highway scenario at varying traffic densities (30 to 100 vehicles) to analyze QoS of VoIP with help of five performance metrics which are describe below under this section.

Simulation Setup – In this paper we used QualNet 6.1 simulation tool. (Qual Net)The QualNet communication simulation platform is a planning, testing and training tool that mimics the behaviour of real communication network. Simulation is cost effective method for developing, deploying and managing network centric system throughout their entire life cycle. User can evaluate the behaviour of the network and test the combination of network features that are likely to work. QualNet provides a comprehensive environment for designing protocols, creating and animating the network scenarios and their performance.

In this paper to analyze the performance of VoIP application over VANET we build up two architecture basis on the environment i.e. city and highway scenario to transmit the audio signals. Also apply high and low traffic condition on them to check its impact on VoIP services.

(1) City Sparse and City Dense Network Scenarios: The city sparse network scenario is created in 1500 X 750m^2 areas with some streets with bidirectional lane which sometime creates traffic problem at intersection point or on road to provide as real city environment. For sparse city network scenario 30 nodes as vehicles are distributed over the scenario. The city dense network scenario is also

created in same area as city sparse network, but it contains 100 nodes as vehicles distributed over the area. In both network scenarios we applied 9 VoIP applications.

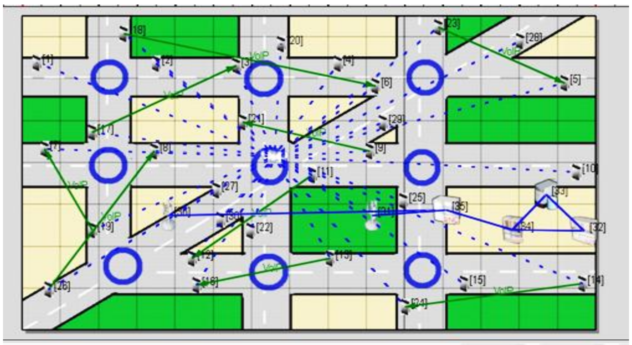

Fig.4 City Sparse Network Scenario

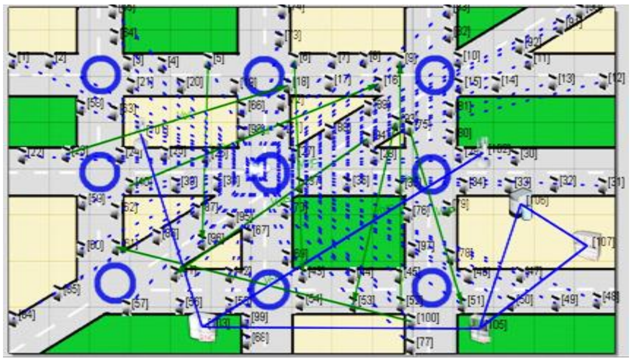

Fig.5 City Dense Network Scenario

(2)Highway Sparse and Highway Dense Scenarios: In highway sparse network scenario 2000 X 500 m^2 area has taken to create two lane roads as in live highway environment; the roads are wide and long. So here, vehicles are more dynamic. It contains 30 nodes as vehicles distributed over highway while in highway dense network scenario contains 100 nodes as vehicle. In sparse and dense scenario of highway, we applied 6 and 12 VoIP sources respectively.

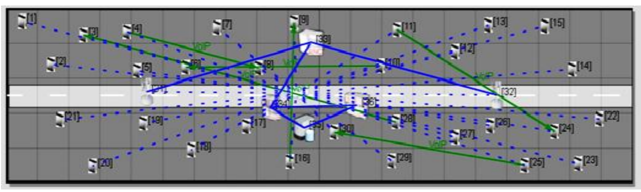

Fig.6 Highway Sparse Network Scenario

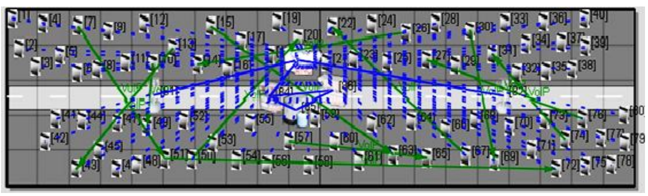

Fig.7 Highway Dense Network Scenario

The network layer employs Bellman ford routing protocol as default to compute the routing paths among the VANET nodes but in city dense network we used reactive routing protocol AODV (K.K.Singh *et al*) for utilizing the bandwidth efficiently as in dense city scenario, the topology is not so high dynamic and the network workload increased by generating the voice traffic with the duration of time. In Qqualnet6.1 simulator we configured the properties of each VoIP source and examined the performance of varying voice CODECs: G.723.1ar5.3, G.723.1ar6.3, G.711, G.726.ar24, G.726ar32, G.728ar16 and G.729. Table II shows simulation set up.

Table II: Simulation Parameter

Parameter	Values or protocols
Simulation Tool	QualNet 6.1
Simulation Time	200s(highway), 240s (city)
PHY/MAC layer	Cellular PHY/Cellular MAC
Network layer	Cellular layer 3(UMTS layer 3)
Transport layer	TCP/UDP
Application layer	H323
Voice CODECs	G.711,G.723.1ar5.3&6.3,G.726ar24&32, G.728ar16 & G.729
VoIP Duration	60s
No. of Channels	Two (1.95 GHz & 2.15 GHz)

Performance Metrics – In our simulation, to evaluate the performance of VoIP over VANET with UMTS, we use the following metrics:

(1)End-to-End Delay: It is the time taken to a packet to be successfully delivered from source to destination. This type of delay includes propagation and queuing delay. According to ITU recommendation, delay should not more than 150 ms and if it exceeds 300ms then the quality of VoIP degrade significantly. Mathematically, it can be defined as:

D end-end = N [D trans + D prop + D proc + D queu]

Where,

D end-end – end-to-end delay
D trans – transmission delay
D prop – propagation delay
D proc – processing delay
D queu – queuing delay
N – Number of links (no. of routers + 1)

(2)Mean Opinion Score (MOS): MOS provides a numerical measure of quality of human speech in voice telecommunication. ITU-T P800 defines MOS score with value ranging from 1 to 5, where 1 is the worst quality and 5 is the best quality. MOS score is found by converting R-Factor scale obtained by the following expression:

$$R = R_0 - Is - Id - Ie + A$$

Where:

- R represents the result voice quality (from 0 to 100),

- R_0 refers to noise ratio,
- *Is* characterizes the simultaneous impairment factor such as too load speech level,
- *Id* represents mouth-to-air delay,
- *Ie* is the equipment impairment factor, and
- *A* is the advantage of access.

(3)Average Throughput: It is defined as the amount of received data per second. Mathematically throughput is shown as follows:
*Throughput (Bytes/sec) = (Total number of received packets * packet size) / (Total simulation tool)*

(4)Average Jitter: It is the variation in the time between packets arriving, caused by network congestion, timing drift or route changes. It is measured in seconds or milliseconds. For VoIP, jitter should be 20 to 30 ms.

*(4)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.

4. Results and Discussion

Results of City scenarios: Average end-to-end delay is presented in fig.8 From the graph we can depicts that in city sparse scenario G.723.1ar5.3 presents the best performance compared with other CODECs. Whereas in city dense scenario G.723.1ar6.3 showing less delay in

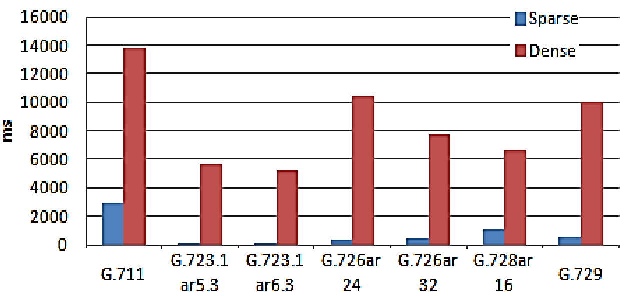

Fig.8 Avg. End-to-End Delay for different audio CODECs in City scenario

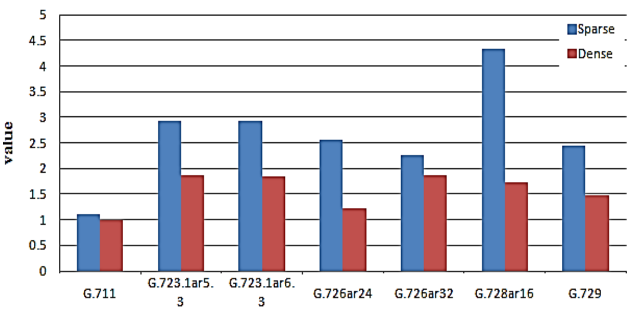

Fig.9 MOS score for different audio CODECs in City scenario

comparison to other CODECs. The MOS is one of the most widely used QoS metric in the VoIP application, which helps to compute a predictive estimation of the subjective voice quality. These results are due to packet size. The larger the packet size, the more time is required to process them. Due to the low packet size and transfer rate of G.723.1 make it ideal codec and due to the larger packet size of G.711, G.711 suffer high end-to-end delay.

However MOS is fundamentally affected by packet loss and delay. In figure we plotted MOS for different CODECs. The best MOS value is 4.33 for G.728ar16 in city sparse environment whereas in city dense environment G.723.1ar5.3 shows acceptable MOS value i.e.1.8573.

Result of Highway Scenario: During simulating the scenario for highway sparse environment, the CODEC G.729 didn't execute as nodes are not generating enough traffic with that CODEC.

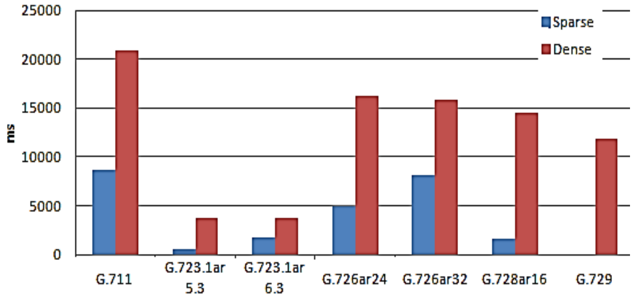

Fig.10 Average end-to-end delay for different audio CODECs of Highway scenario

Fig.10 shows the analysis of average end-to-end delay. VoIP CODEC G.723.1ar5.3 has lowest end-to-end delay for both highway sparse scenario & highway dense scenario.

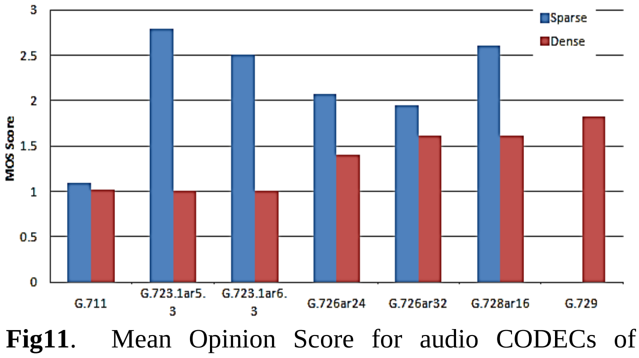

Fig11. Mean Opinion Score for audio CODECs of Highway scenario

In fig.11 we plotted MOS for different CODECs for sparse and dense traffic densities. From figure the best MOS value is 2.786 for G.723.1ar5.3 in sparse which seems quietly acceptable but in dense all CODECs have poor MOS value but G.729 have little better value than others.

The rest simulation results of both city and highway scenarios in terms of performance metrics for CODECs analysis are given in a tabular form. The Table III shows that which CODEC performed best with respect to which performance metric.

TABLE III Summarized result of simulation

	City Scenario		**Highway Scenario**	
Performance Metrics	**Sparse**	**Dense**	**Sparse**	**Dense**
Avg. Delay	G.723.1ar5.3	G.723.1ar5.3	G.723.1ar5.3	G.723.1ar
Avg. Throughput	G.711	G.726ar32	G.711	G.726ar32
Avg. Jitter	G.726ar32	G.711	G.726ar24	G.726ar24

5. Conclusion

In this paper, the performance of different CODEC's has been evaluated for VoIP with VANET, designed with UMTS and H.322. After evaluating the results of all designed scenarios, results are concluded with accordance to the city and highway environment with varying traffic condition. The performance of different CODECs showed a strong reference for deployment of VoIP services through VANET. From simulation results, we observed that:

- In terms of throughput, G.711 presents the best performance in sparse condition of both city and highway scenario but as the network becomes dense G.726ar32 presents high throughput.
- In term of End-to-End Delay, G.723.1ar5.3 presents optimal performance in both city and highway with sparse condition and also in highway dense network.
- G.723.1ar5.3 shows less average delay in both city and highway network with both traffic condition.
- With regard to MOS metric, G.723.1ar5.3 only in sparse network of highway provide quite acceptable quality whereas in dense network of all CODECs have poor quality. In city sparse network G.728ar16 has high quality whereas in dense network G.723.1ar5.3 has better MOS value comparison to other CODECs.
- In both scenario of highway G.723.1ar5.3 is suitable for jitter whereas in city, G.726ar32 for sparse and G.711 for dense are suitable for jitter.

References

Ming Yu Jiang, An Introduction to Mobile Ad hoc Network.

Mohammad Azouqa, Ali Jamoos, and Samer Bali, Investigation Of Vehicle-To-Vehicle Communication In A Road Intersection, available: http://www.comesafety.org

Ismail Dalgic and Hanlin Fang (1999). Comparison of H.323 and SIP for IP Telephony Signaling, Multimedia Systems and Applications, *Proc. SPIE* 3845; doi:10.1117/12.371194; 106(2).

Hong Xiong, Yuan Guo, Fang Zhu, Simple Voice over IP (VoIP) Implementation, *EEL6586, Spring* 2003.

William Lehr and Lee W. McKnight (2003) Wireless Internet access: 3G vs. WiFi?, *Elsevier Science Ltd*

James Toga and Jorg Ott (1999) ITU-T standardization activities for interactive multimedia communications on packet-based networks: H.323 and related recommendations, *Elsevier Science;* B.VAll rights reserved. PII: S0169- 7552_98.00267-0

Markku Korpi.; Siemens AG.; Vineet Kumar and Intel Corporation (1999) Supplementary Services in the H.323 IP Telephony Network, *IEEE Communications Magazine;* 0163-6804/99/$10.00.

Mobile Computing, by Asoke Talukder

D.Shinder (2007). Mobile VoIP [online]. Available: http://www,fring,com/

WCDMA for UMTS, Third Edition, Harri Holma & Antti Toskala, Page no.8

Sheetal Jadhav.; Haibo Zhang and Zhiyi Huang (2011) Performance Evaluation of Quality of VoIP in WiMAX and UMTS, *International Conference on Parallel and Distributed Computing, Applications and Technologies*, pdcat;375-380

Said El Brak, Mohammed Bouhorma, Mohamed El Brak and Anouar Bohdhir (2013) Speech quality evaluation based CODEC for VoIP over 802.11P, *International Journal of Wireless & Mobile Networks (IJWMN)* ;5(2).\

Sajal K. Das, Enoch Lee, Kalyan Basu, Naveen Kakani and Sanjoy K. Sen (2003)Performance Optimization of VoIP Calls over Wireless Links Using H.323 Protocol.52(6):742-752.

Abderrahim Benslimane.; Tarik Taleb and Rajarajan Sivaraj (2011) Dynamic Clustering-Based Adaptive Mobile Gateway Management in Integrated VANET – 3GHeterogeneous Wireless Networks, *IEEE Journal on selected area in communication*: 29(3)

Qual Net 6.1 Simulator; http://www.scalable-networks.com (2013-07-08)

K.K.Singh & Akansha Singh, Mobile Computing, page no.230.