Performance Analysis of VoIP over Mobile WiMAX

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***Abstract*-** **In recent years, Voice over Internet Protocol (VoIP) has become a popular technology as an alternative to public switched telephone network (PSTN). Skype, Viber and KaKaoTalk are the examples of VoIP applications. Worldwide Interoperability for Microwave Access (WiMAX) is a network technology that supports VoIP. The target of this project is to evaluate the VoIP performance over mobile WiMAX. The performance metrics such as jitter, delay and Mean Opinion Score (MOS) value are critically analyzed using different codecs. The comparison is also being made between the pedestrian and vehicular environment. OPNET 14.5A is chosen as the simulation tool since it resembles real environment simulation. It can be concluded that VoIP performance is affected by the codecs used as well as the network environment.**

Keywords — VoIP, WiMAX, MOS, jitter, delay, throughput, codecs.

I. Introduction

While VoIP has been around for years, it has not been viable alternative for most applications due to technology constraints. Recent technology enhancement have improved quality and now VoIP service providers are positioned to offer an affordable alternative to traditional circuit-switched voice services[1] . Mobile WiMAX IEEE 802.16e offers large coverage area approximately 50 km and high data rates up to 75 Mbps [2]. WiMAX provides wide range of applications such as VoIP, Internet Protocol Television (IPTV) and mobile data TV. In this paper, we analyze the required quality of service (QoS) for VoIP applications in mobile WiMAX technology. OPNET 14.5.A simulator is used to analyze the QoS of VoIP application under various codecs.

A. VoIP

VoIP is the real-time transmission of voice signals using the Internet Protocol (IP) over the public Internet or a private data network. The voice signal is converted into a digital packet and compressed for efficiency using compression algorithms/codecs. Then the signal transferred over the Internet [3]. One of the most significant advantages of VoIP over a PSTN is that user can make a long distance phone call and bypass the toll charge. VoIP allows communications on the existing IP networks without adding additional lines or bandwidth. VoIP traffic potentially suffers from performance issues like packet loss, communication delay, jitter and echo, which can greatly affect QoS.

B. Mobile WiMAX technology IEEE 802.16e

Mobile WiMAX is an IP-based architecture based on wireless metropolitan area networking (WMAN) standards developed by the IEEE 802.16 group and adopted by both IEEE and the ETSI HIPERMAN group. It supports various applications such as VoIP and video streaming. Furthermore, mobile WiMAX offers high data rates, large coverage area as well as low cost of deployment.

C. Performance Metrics

1) Mean Opinion Score (MOS)

MOS is a numerical measure of the quality of human speech in voice telecommunications, with value ranging from 1 to 5 where 1 is the worst quality and 5 is the best quality.

2) Jitter

Jitter is the undesired time delay from the packets sending end to receiving end in communication network, simply stated that the variation of packet interarrival time. The delay is inevitable and high levels of jitter leads to large numbers of packets to be discarded by the jitter buffer in the receiving IP phone or gateway.

3) Throughput

Throughput is the amount of data transferred from one source to destination or processed in a specified amount of time. Typically, throughput is measured in kbps, Mbps and Gbps.

4) End to End Delay

End to end Delay is the total transit time for packets in a data stream to arrive at the endpoint and it is inevitable in communication system.

D. QoS Service Class in WiMAX

The QoS is granted on the basis of type of application and service under consideration [3]. There are five main service classes known as Unsolicited Grant service (UGS), Extended Real-Time Polling Service (ertPS), Real Time Polling Service (rtPS), Non Real Time Polling Service (nrtPS), and Best Effort (BE). These services are prioritized in decreasing order. Table I briefly defines WiMAX service classes and its applications.

TABLE I

WIMAX SERVICE CLASSES

Service Class	Description	Applications
Unsolicited Grant Service(UGS)	For constant Bit rate and delay dependent applications	VoIP
Real Time Polling Service (rtPS)	For variable rate and delay dependent applications	Streaming audio , video
Extended Real Time Polling Service (ertPS)	For variable rate and delay dependent applications	VoIP with Silence Suppression
Non Real Time Polling Service (nrtPS)	Variable and non real time applications	FTP
Best Effort (BE)	Best effort	Email , Web Traffic

E. Codecs

VoIP uses Internet Protocol for transmission of voice over IP networks. The process involves digitization of voice, the isolation of unwanted noise signals and then the compression of the voice signal using compression algorithms/codecs [4]. Voice compression using a codec is expected to provide good voice quality even after compression, with minimum delay. The codecs under investigation includes G.711, G.729A and G.723.1. TABLE II shows several types of voice codecs.

TABLE II

COMMON VOICE CODECS

Codec	Data Rate(kbps)	MOS Value
G.711	64	4.3
G.723	5.3	3.6
G.726	32	4.0
G.728	16	3.9
G.729	8.0	4.0

F. Multipath Channel Model

OPNET has different multipath channel models defined for mobile WiMAX link. They are defined based on International Telecommunication Union (ITU) multipath models. In this paper, the ITU Pedestrian A and ITU Vehicular A multipath channel models are compared. The multipath channel model is defined on the Mobile Station (MS) and it applies to both the uplink and the downlink transmission between the Base Station (BS) and MS.

II. RELATED WORK

There are a number of studies on VoIP applications over WiMAX that has been conducted in order to assess and monitor the deployment of VoIP into the existing network [5]. Two ways of determining the performance of VoIP are through real measurement-based and via simulation. The simulations environment can be varied in order to obtain accurate results. Different VoIP codecs and WiMAX service classes can be used in order to investigate and analyze the behavior and performance of the model [5]. VoIP equipments used these compression/decompression methods for conversion of analogue audio signals to digital bit stream. These techniques reduce the required bandwidth with assured voice quality [6]. In G-series recommendations by ITU-T, the most popular voice coding standards for telephony including G.711, G.726, G.728, G.729A and G.723.1 [7]. VoIP performance also can be compared over different network technologies such as in Wireless Fidelity (Wi-Fi) and WiMAX in terms of throughput, jitter and packet losses. Simulation approach using NS2- network simulator is also can be used to evaluate the VoIP performance. In Wi-Fi, noticed that the capacity for voice can be quite low for Wireless Local Area Network (WLAN). VoIP traffic and traditional data traffic such as web traffic can mingle with each other that reduce the VoIP performance [8]

III. METHODOLOGY

In this project, two environments are analyzed, pedestrian and vehicular. The VoIP codecs used are G.711, G.729A and G.723.1. In order to evaluate the performance of VoIP over the mobile WiMAX network, a scenario was designed in the network simulator OPNET with the assumption that the traffic generated in this network model is VoIP only. The UGS service class is applied for the VoIP. Three MS are served by a BS in a campus network. For pedestrian environment, the speed of the MS is set to 5 km/h while for the vehicular environment is 10 km/h. Fig. 1, Fig. 2, Fig. 3 and Fig. 4 show the network model and parameters used in the simulation in OPNET respectively.

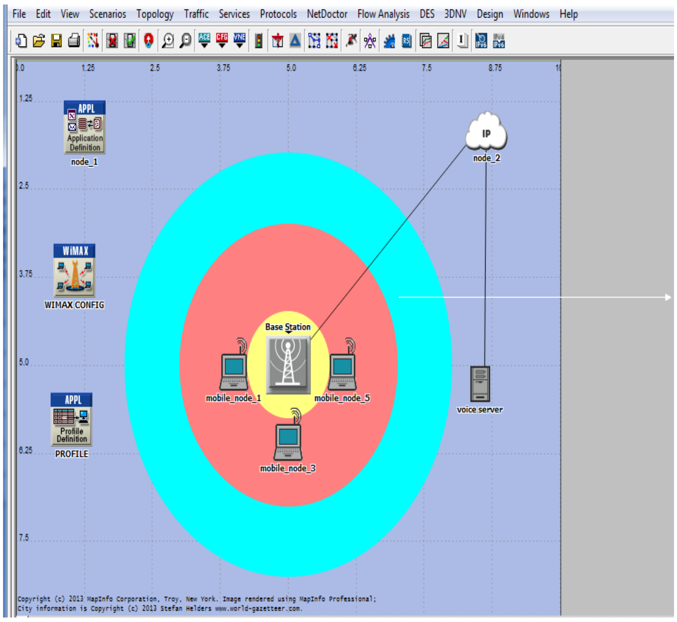

Fig.1 WiMAX Network Model

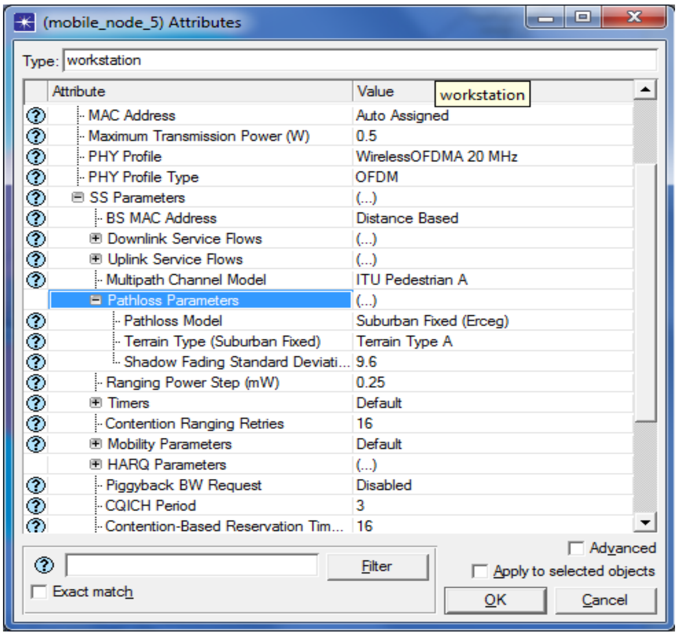

Fig.3 MS Parameters

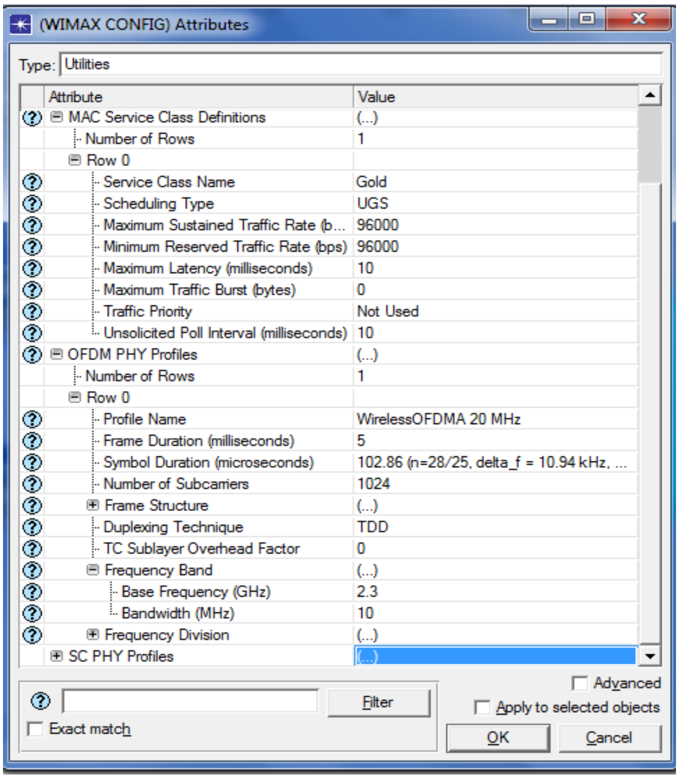

Fig. 2 WiMAX Configuration

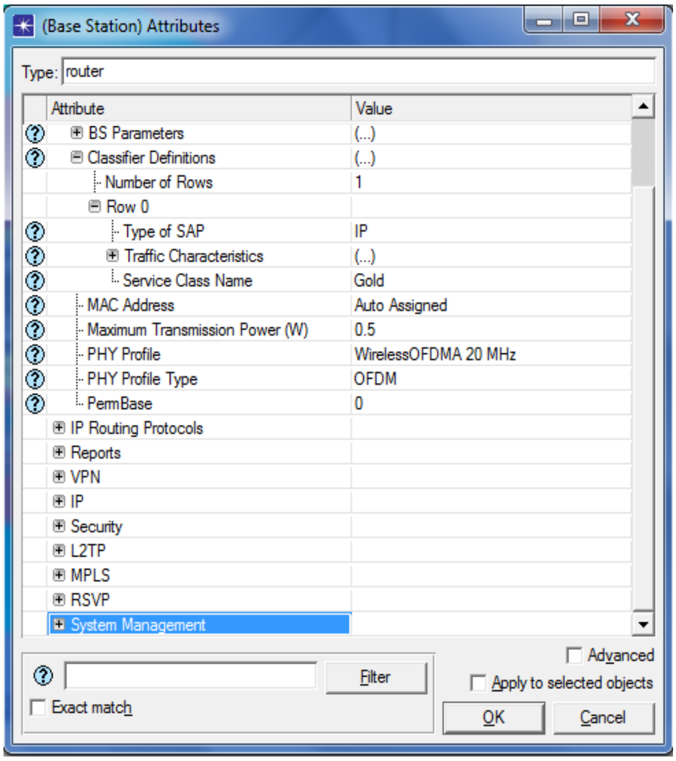

Fig. 4 BS Parameters

IV. RESULTS AND DISSCUSSION

A. Pedestrian Environment

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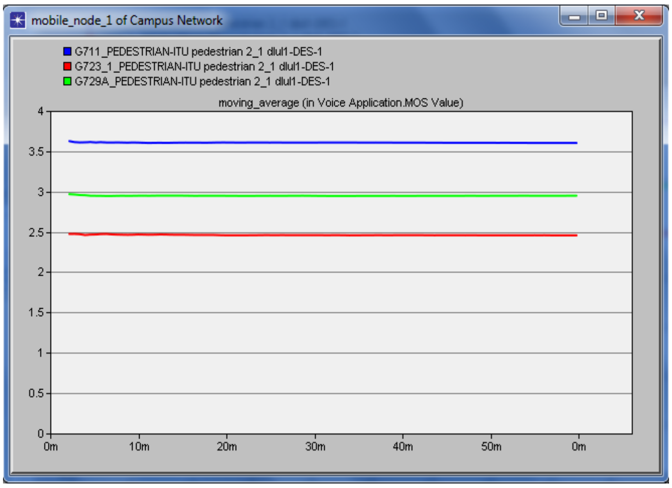

Fig. 5 MOS value for each codec

Fig. 5 shows the MOS value for the three codecs. G.711 gives the highest MOS value which is 3.7. The lowest MOS value 2.5 is given by G.723.1. G.723.1 generates higher number of packets with lower packet size as compared to G.711 which resulting network congestion and thereby packet drop. Hence the MOS value for G.723.1 is quite low. The jitter levels for the three codecs are represented in Fig. 6. G.723.1 codec scheme has highest value of jitter. The negative value of jitter means that the time difference between the packets at the destination is less than that at the source. The voice jitter value for G.711 is the lowest which means that there is a minimum delay variation between the VoIP packets.

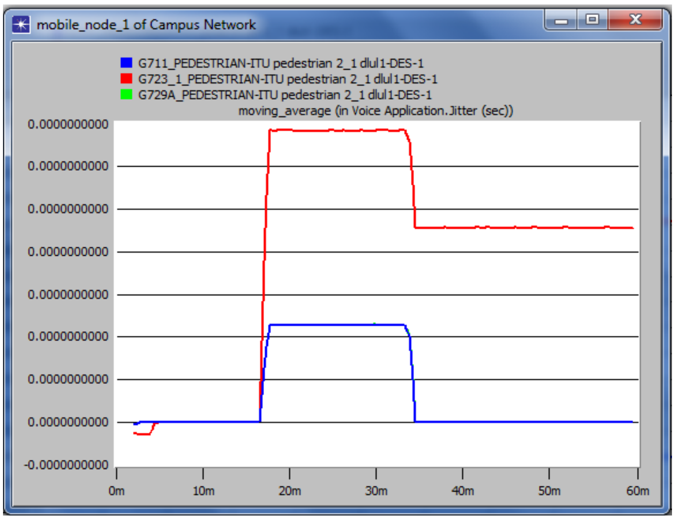

Fig. 6 Jitter

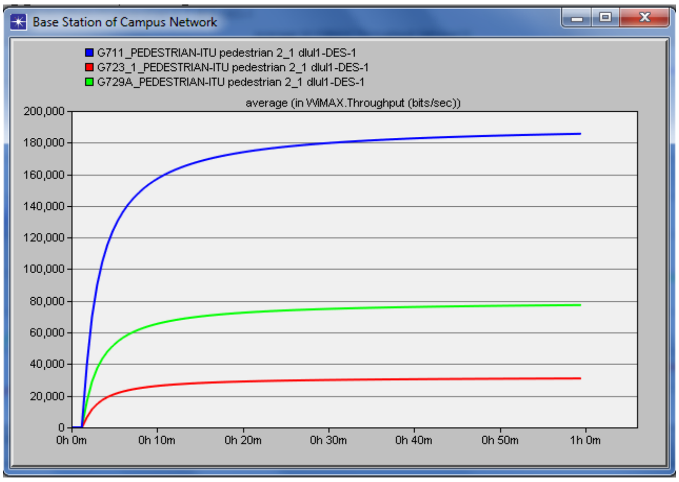

Fig. 7 Throughput

Considering throughput as one of the important parameter of WiMAX network, throughput represents the total data traffic in bits/sec forwarded from WiMAX layer to higher layers in all WiMAX nodes of the network. Fig. 7 shows that the voice G.711 codec has the highest throughput and least throughput for G.723.1 codec. In Fig. 8, G.723.1 codec gives the highest packet end to end delay as compared to G.711 and G.729A. This is due to coding rate of 5.3Kbps and 6.3Kbps used by G.723.1 which results in the formation of high number of packets with smaller size. As the number of packets increases in the network, the congestion in the network increases. Congestion leads to the network packet delay and thus increases packet end to end delay. The packet end to end delay is lowest for G.711. This implies G.711 can provide better VoIP services in terms of end to end packet delay.

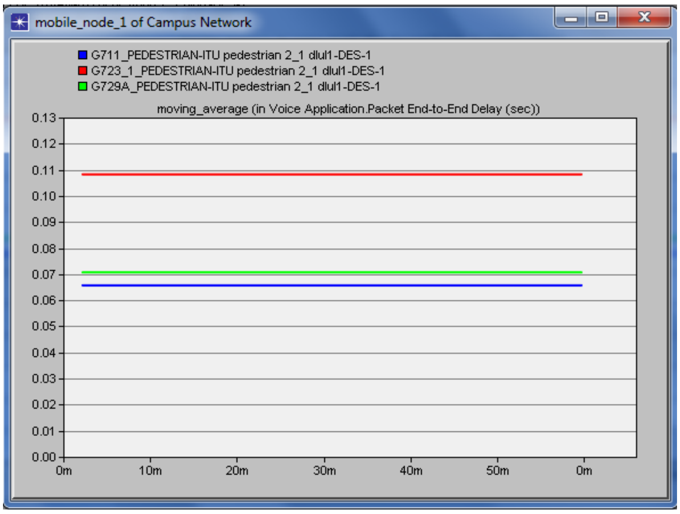

Fig. 8 Packet End-to-End Delay

B. Pedestrian and Vehicular Environment Comparison

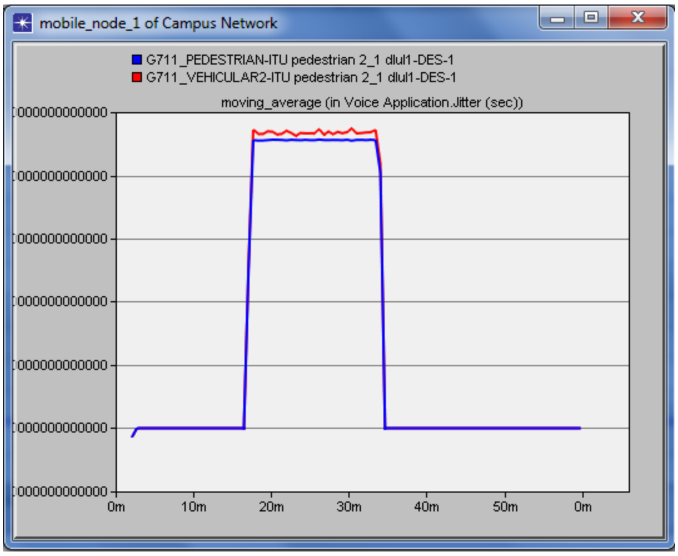

Fig. 9 Jitter for Pedestrian and Vehicular environment

For comparison purpose, each codec is simulated for both pedestrian and vehicular simulation environment respectively. But in this paper only G.711 codec performance represented for both scenarios. It shows that the mobility and speed affect the VoIP performance. The speed for vehicular is 10 km/h as compared to pedestrian which is 5 km/h. Figure 9 shows that the jitter in vehicular is higher than pedestrian. As the MS moving during communication, the voice quality also affected depends on speed and network environment. Figure 10 represents the MOS value for both scenarios. The MOS value slightly lower for vehicular environment as compared to pedestrian.

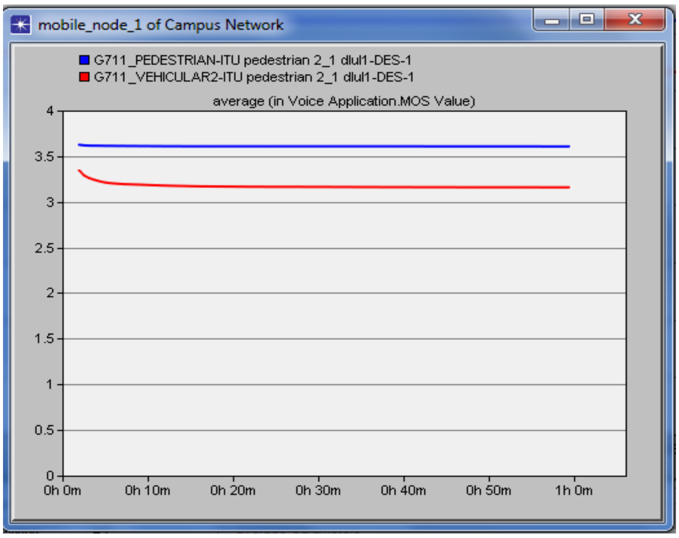

Fig. 10 MOS value for Pedestrian and Vehicular

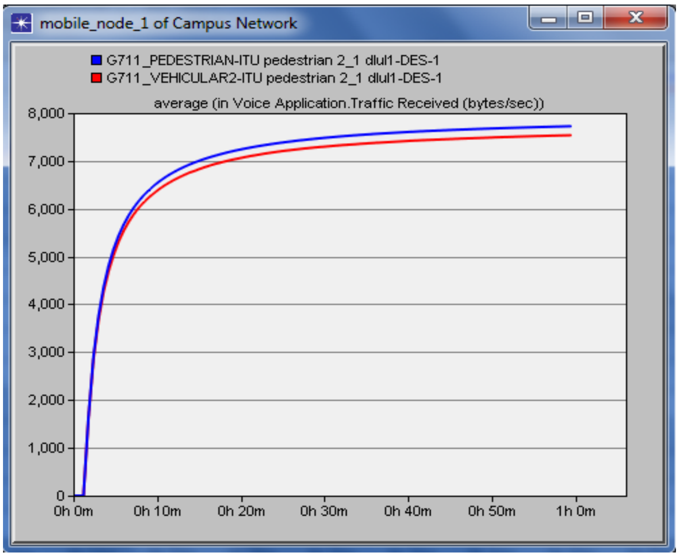

Fig. 11 Throughput for Pedestrian and Vehicular

As mentioned previously, the network environment and MS speed are the important factors that affecting the performance of VoIP. Figure 11 shows that the throughput for vehicular is lower as compared to pedestrian. Meanwhile, vehicular shows the higher delay variation as compared to pedestrian as represented in Fig. 12 below.

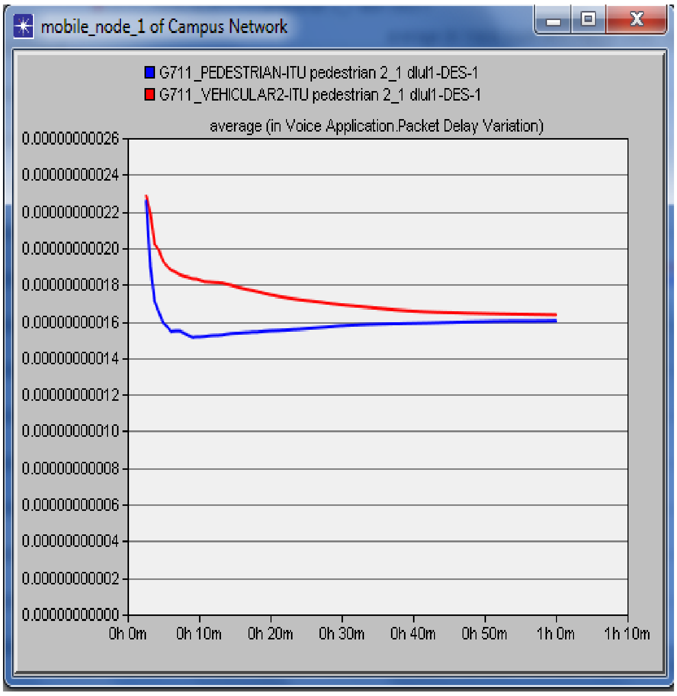

Fig. 12 Packet delay variation

V. CONCLUSION

In this paper, we have analyzed the performance of three VoIP codecs over Mobile WiMAX networks. The VoIP performance was simulated considering two simulation scenarios using OPNET 14.5.A. using three different codecs. The MOS value, average end-to-end delay, and jitter were used as performance parameters that define VoIP QoS. The G. 711 codec offers the best performance for VoIP over WiMAX. However, all three codecs G. 711, G. 723, and G. 729A show acceptable performance quality for VoIP over WiMAX. VoIP performance in vehicular is lower as compared to pedestrian network environment. Since the mobility and number of user also affects VoIP performance, its impact could be analyzed further by comparing different technology in future. WiMAX networks may also employ other VoIP codecs such as G. 722, G. 726, G.728 and GSM.

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