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# Audio Codecs Impact on Quality of VoIP Based on IEEE802.16e Considering Mobile IP Handover

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**Abstract:** A simulation model using OPNET tool is introduced for testing audio codecs impact on quality of VoIP based on IEEE802.16e with taking into account handover performance. Different parameters that indicate the quality of VoIP such as Throughput, MOS, End to End delay and traffic send and received. According to our simulation the acceptable MOS value is recorded for codecs G.711 and GSM-FR.

**Keywords:** Audio, Codecs, QoS, WiMAX, Handover

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## 1. Introduction

In the recent years the Worldwide Interoperability for Microwave Access (WiMAX) placement is growing rapidly. IEEE 802.16e (Mobile WiMAX) becomes a prevalent technology for holding mobile clients because the base station (BS) of mobile WiMAX has the vital benefit to serve considerable coverage areas. During data transmission, mobile WiMAX permits the user to move freely. The major consideration of mobile WiMAX is that when the user is moving from one BS to another there must be no loss in data, i.e. during the handover which represents the method when a mobile node (MN) alters the serving BS [1]. Serving a large number of MNs practically demands an effective scheme of handover. Presently, a considerable delay in handover of mobile WiMAX is contributed to the overall end-to-end communication [2]. In the recent researches more attention is paid to increase the handover schemes efficiency. In this research, a comparison to study the performance of Mobile WiMAX handover among various audio codec schemes will be examined. The rest of this research will be structured as follows: the next section will object to the related works. Section 3 refers to the WiMAX handover procedure. In section 4, voice encoding schemes of utilizing VoIP over IEEE802.16e network will be clarified. Section 5 presents network model implementation. In Section 6 the simulation results will be discussed and finally, conclusion will be stated in section 7.

## 2. Related Works

Our study recognizes a number of earlier researches in the same course. The main contribution in [3] is to assess and examine the performance of several audio encoding schemes employing RSVP in a VoIP based on WLAN. The results were compared among three audio codecs namely; G.711, G.723.1 and G.729 utilizing the services of RSVP. On the other hand, the researchers in [4] examined mobile WiMAX performance using voice codecs under only two conditions and showed that the performance of audio codecs stay fixed for random way point and group mobility manner.

The researchers in [5] introduced an investigation to study the performance of different audio codecs based on various service classes such as rtPS, ertPS, nrtPS, Best Effort and NOAH routing protocols. While the authors in [6] introduced the performance evaluation of VoIP in wireless networks considering packet end-to-end delay and throughput without taking into account the mobility of nodes. In [7] the study examined the handover in order to delay characteristic and throughput involving UDP and TCP protocols for two types of fading models. An efficient technique of forced handover to progress capacity handling and performance of base station in [8] was introduced; however, the study did not mention audio performance during handover. In [9] the researchers provided an evaluation for different scenarios of WiMAX handover with several movements. While in [10] the authors presented an examination of handover impact on the speech performance in VoIP but audio codecs did not take

into consideration. This study will provide an examination of the performance of WiMAX handover using four kinds of audio codecs namely; G.711, G.723.1, G.729A and GSM-FR.

### 3. IEEE 802.16e Handover Procedure

In general, due to weak signal received, cell capacity and radio channel condition the handover is occurring. The most important component before occurring handover process is the received signal strength. The MN will exchange to the adjacent BS if a significant drop in the received signal strength is observed [7, 8]. The procedure of handover can be made according to the following phases.

- Phase 1: Cell Reselection

Reselection of cell indicates to the procedure that MN is scanning and/or associating with one or more BSs according to fix handover suitability taking into account other performance issues. The MN might be used the acquired data of neighbor BS from a decoded data to arrange intervals of scanning or sleep intervals to scan the neighbor BS for the reason of evaluating the handover of MN to a potential target BS [8, 9].

- Phase 2: Handover Decision and Initiation

The decision to transfer the mobile station to the desired BS from the serving BS is known as initiation of a handover. Both BS and MN can be started this decision. Starting the actual handover requires sending a handover request that will activate a specific messages sequence of handover to be sent between MN and BS [8, 9].

- Phase 3: Synchronization

The synchronization of MN to its downlink channel is required to create the communication with the desired BS. In this stage, the MN starts receiving the uplink and downlink

transmission parameters. This process can be reduced if the MN earlier received data about the desired BS (during the acquisition of network topology) [8, 9].

- Phase 4: Ranging

The target BS and the MN must have handover ranging. Re-entry of network progress from the initial ranging steps in the network entry process negotiates basic capabilities, authentication stage, establishment stage, registration and optional network entry steps. Network re-entry may be reduced by target BS possession of information of MN, acquired from serving BS over the backbone network. Based on the amount of the acquired information, the target BS may choose to avoid one or several of network entry steps [8]. Regardless whether the serving BS received MN information or not the target BS will demand MN information from the backbone network [8, 9].

- Phase 5: Termination of Mobile Node

Termination of the MN context is expressed as the serving BS termination of the context of all connections belonging to the MN and the removal of the context associated with them [8, 9].

### 4. Voice Encoding Schemes

Voice codecs are used to convert an analog voice signal into digitally encoded version. The codecs may vary in sound quality, required bandwidth and computational requirements, etc. Table 1 shows the commonly used voice codecs with their algorithms, bit-rates and mean opinion score (MOS) [4]. Voice codec can be briefly defined as an algorithm used to code and decode voice stream. The voice quality experienced by the end-user depends on the performance of the codec [11].

**Table 1.** Features of the most common codecs.

IUT-T Codec	Algorithm	Frame (ms)	Bit Rate (kbps)	Bits per frame	Codec delay (ms)	Comments
G.711	Pulse code modulation (PCM)	0.125	64	8	0.25	Delivers precise speech transmission. Very low processor requirements.
G.723.1	Algebraic Code Excited Linear Prediction (ACELP)	30	5.3	159	67.5	High compression with high quality audio. Can use with dial-up. Lot of processor power.
G.729A	Conjugate Structure Algebraic Code Excited Linear Prediction (CS-ACELP)	10	8	80	25	Excellent bandwidth utilization. Error tolerant.
GSM FR	Regular Pulse Excitation Long Term Predictor (RPE-LTP)	20	13	260	40	High compression ratio. Free and available in many hardware and software platforms. Same encoding is used in GSM cell phones

### 5. Network Model Description

The configuration of the suggested Mobile WiMAX network model will be done using OPNET environment. OPNET simulator is chosen to its global recognition in terms

of accuracy and reliability to get the results which are much closed to the real life. Also, it provides the necessary modules to study the performance of Mobile WiMAX (IEEE802.16e) with respect to the impact of audio codecs, mobile IP, handover and other mobility features.

The suggested model will be implemented using four

different scenarios to evaluate the best voice encoding schemes of utilizing VoIP over IEEE802.16e network. It is taken into account the impact of mobile IP and a handover mechanism which occur during the movement of MN across the cells. In the proposed simulation, all scenarios have the same structure and topology and the comparisons will be conducted using different values of parameters.

The first scenario will implement with voice encoding

G.711 over mobile VoIP handover environment. The second scenario will employ with G.729A audio codec over mobile VoIP handover environment, while the third scenario will be examined using G.723.1 and finally, GSM audio codec over mobile VoIP handover environment will be used in scenario four. The network topology of the simulation environment is illustrated in Figure 1.

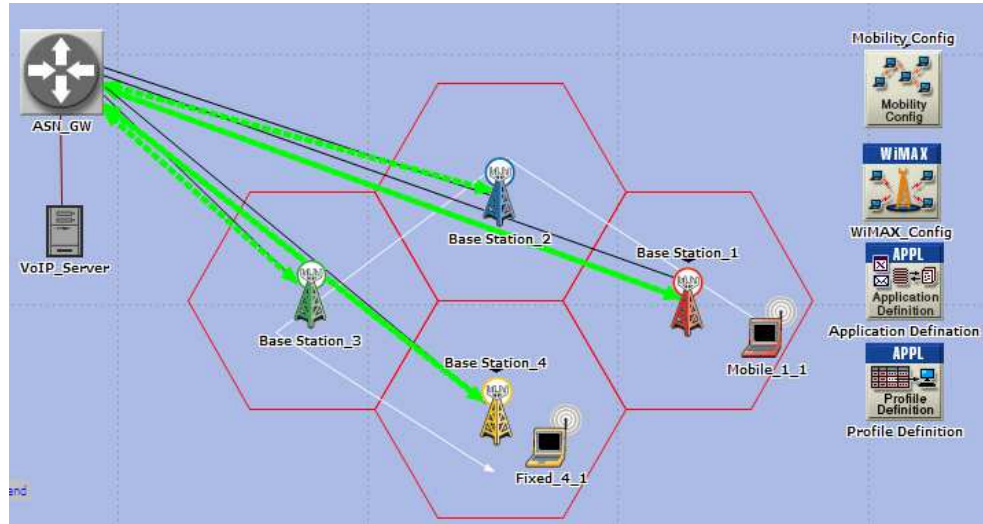


Figure 1. Network Topology of IEEE802.16e.

The scenario of the simulated network consists of four hexagonal cells of IEEE802.16e networks, i.e. single BS per cell. All BSs are connected to the core network (VoIP\_Server) via Access Service Network Gateway (ASN-GW). The ASN-GW supports the mobility in IEEE802.16e network and the VoIP\_Server handles the VoIP traffic. The researchers use two MNs with varying movements. The first one is (fixed\_4\_1) fixed and another node is (mobile\_1\_1) moved along with the trajectories indicated by a white color line around the cells. The green bidirectional dotted lines represent the generic routing encapsulation (GRE) tunnels. Each BS connected to the ASN-GW by using the PPP DS link which has a 1.544 Mbps capacity. The name of the link in OPNET model is 'point\_to\_point\_link\_adv'. The ASN-GW is connected to the VoIP\_Server by using 1000BaseT Ethernet cable which has 1Gbps capacity.

The names of the ASN-GW and servers in OPNET model are 'ethernet4\_slip8\_gtwy' and 'ethernet\_server', respectively. The VoIP node model is 'wimax\_ss\_wkstn\_adv' and the BS (Foreign and Home agent) model is 'wimax\_bs\_router\_adv'. The WiMAX parameters of MNs and BSs are given in Table (2) and Table (3) respectively.

Table 2. Mobile Node Configuration Parameters.

Application	VoIP
Antenna Gain	-1 dBi
Max Power Transfer	0.5 W
PHY Profile	Wireless OFDMA 20 MHz
Modulation and coding	Adaptive
Buffer Size	64 KB
Number of Transmitter	SISO

Table 3. Base Station Configuration Parameters.

Antenna Gain	15 dBi
Max Power Transfer	0.5 W
PHY Profile	Wireless OFDMA 20 MHz
Modulation and coding	Adaptive
Number of Transmitter	SISO

#### • Mobility Configurations

In our simulation model that shown in Figure (1) we assume that MN is carrying voice traffic which is ongoing between Mobile1\_1 and VoIP\_Server. When the simulation time is started, Mobile1\_1 initiates the first session with Base station\_1 (Home agent). After 9.18 min, the MN performs a handover from the current Home\_agent spot where the session was generated to a Base station\_2 (Foreign\_agent\_2) spot. This session will be continued until reaching Base Station\_4 (Foreign\_agent\_4) which is passed through Base Station\_3 (Foreign\_agent\_3). During this movement, the MN will change its Home agent and access address which imply that its IP address also will be changed. According to the handover process, the session at the Home agent will terminate, and the MN has to trigger the handover procedure. The new access router and the foreign BS have not information about the terminated session, so that MN has to register with the foreign BS and to establish IP connectivity with new access address and also reestablish session flow with VoIP\_Server. Figure (2) shows an IP configuration of our simulation.

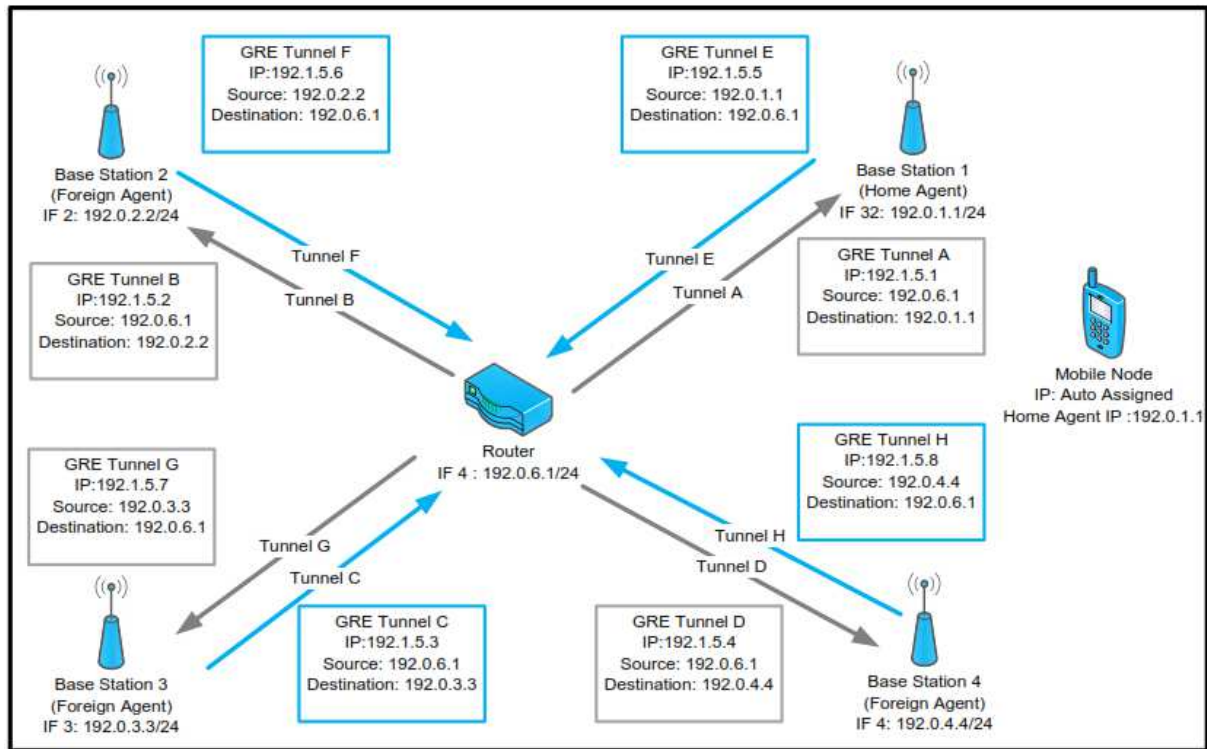


Figure 2. IP configuration for the proposed scenario.

The mobility Configuration defines a mobility profile for the wireless MNs in IEEE802.16e network. The mobility parameters of the simulation scenarios are given in Table (4) and Table (5) respectively.

Table 4. Handover Parameters.

<b>MS Handover Retransmission Timer (ms)</b>	<b>30</b>
Maximum Handover Request Retransmissions	6
Handover Threshold Hysteresis (dB)	0.4
Multi target Handover Threshold Hysteresis (dB)	0
Maximum Handover Attempts per BS	3

Table 5. Scanning Parameters.

<b>Scanning Threshold (dB)</b>	<b>35</b>
Scan Duration (N) (Frames)	3
Interleaving Interval (P) (Frames)	255
Scan Iteration (T)	5
Maximum Scan Request Retransmissions	8
Ground Speed km/h	10

The application Definitions define traffic generation on the network. The VoIP application is set up as the application of the whole network.

## 6. Results of the Simulation

In this section, the results are obtained after implementing the IEEE802.16e network simulation and the statistics are collected by using OPNET Modeler under four different scenarios. The collected statistics from the simulation include Throughput (packet/sec), Mean Opinion Score (MOS), Voice packet end to end delay (sec), Voice traffic send and received

(packet/sec). The results of the simulation potentially indicated to the degree of performance of the audio codecs over mobile IP handover environment based on voice communication.

The curves in Figure (3) correspond to the throughput of Mobile\_1\_1 node versus simulation time with the four audio codecs: G.711, G.723.1, G.729A and GSM-FR. According to the gained results, it is observed that the maximum rate of throughput is recorded for the codec G.729A with the rate of 195 packets /sec. On the other hand, G.711 and GSM-FR codecs have the lower throughput rates. The throughput in G.711 codec is ranged from 120 packets/sec to approximately 10 packets/sec, while GSM-FR codec is ranged from 20 packets/sec to roughly 65 packets/sec. The lowest rate belongs to codec G.723.1 which is oscillated from 5 packets/sec to about 23 packets/sec.

Additionally, it can be clearly seen that throughput of mobile\_1\_1 node is reduced momentarily after 540 sec, 1320 sec and 2100 sec respectively; because when the MN moves outside the home agent coverage area and passes through boarder of foreign agent cell 1, 2 and 3 handover will occur. This reduction will continue until handover is completed. During the handover, the Mobile\_1\_1 node will receive signals from two base stations. When the setup of the new connection is completed, the transmission from/to the serving (Home Agent) BS will be ended, i.e. when the MN crosses a boarder of cells between the serving (Home Agent) BS and target (Foreign Agent) BS, the connection with the serving (Home Agent) BS will be finished. After that, a new connection with the target (Foreign Agent) BS will be



established.

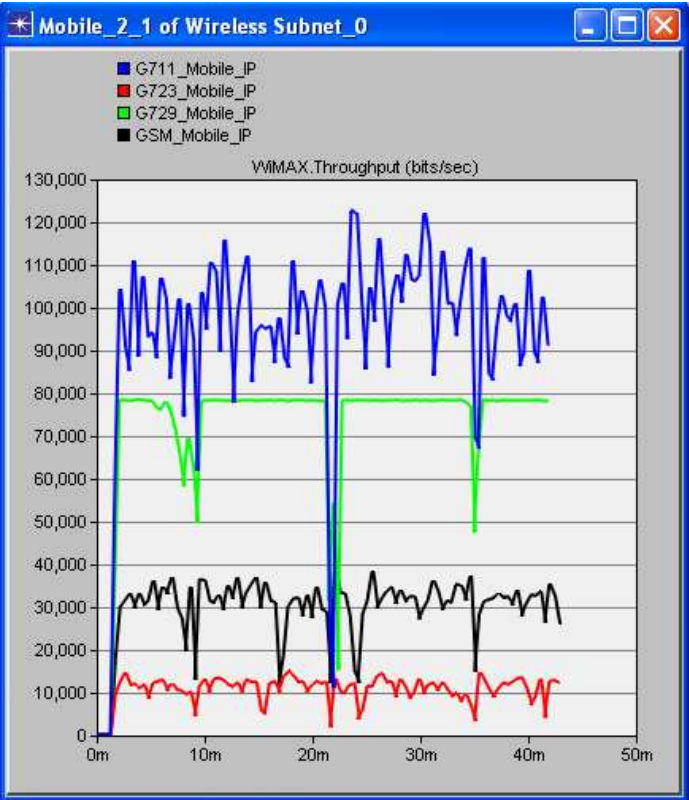


Figure 3. Throughput of various audio codecs.

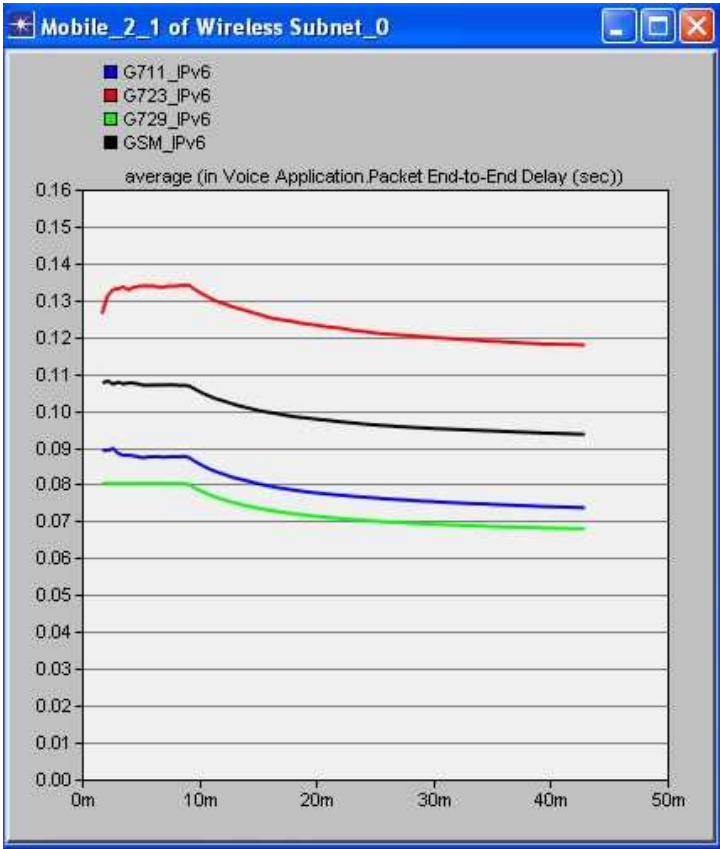


Figure 4. packet End to End delay (sec) under various audio codecs.

Figure (4) illustrates the obtained results between average end to end delay metric and simulation time. It can be seen that the average end-to-end delays of G.723.1 are larger than 0.11 sec which give a lower performance with respect to the other codecs. This result is due to small transfer rate 5.3 Kbps for G.723.1. On the other hand, the average end-to-end delays of GSM-FR and G.711 are larger than 0.09 second and

0.075 second, respectively, the reasons for that are the high bits per frame which are required more time to process them. While, G.729A has small delay with only 0.08 sec due to the high transfer rate and low packet size. These reasons make the G.729A provide the best performance of VoIP applications over IEEE802.16e network.

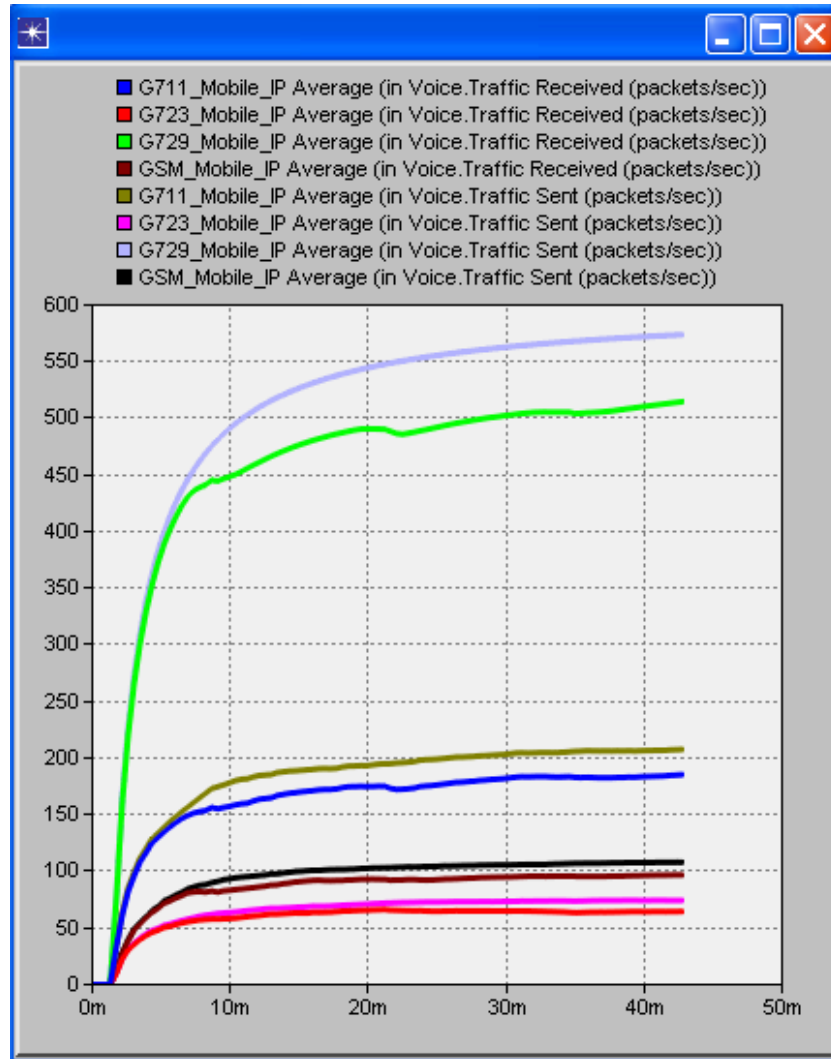


Figure 5. Traffic send and received.

Figure (5) represents the relationship between traffic received and traffic sent versus simulation time with different audio codecs. In order to examine network efficiency, the loss between two traffics (sent and received) must be less enough to make them identical to each other [13]. During the simulation time, the deviation of the traffic received to the traffic sent in G.711 and 729.A codecs is large comparing with GSM-FR and G723.1 codecs that have small deviation. The results indicate that codecs GSM-FR and G723.1 in IEEE802.16e network provide better efficiency since the path loss between traffic sent and received is less compared with

other codecs.

Figure (6) shows the average MOS value for the four codecs. The MOS value gives a description of the perceived quality of receiving voice after being transmitted and compressed using codecs. According to [14], the acceptable MOS value in our results is recorded for codecs G.711 and GSM-FR with a value of 3.1 and 3 respectively. In the contrast, the MOS value of codecs G.723.1 and G.729A is less than 3 which mean that these values indicate that the quality of service is poor and cannot be acceptable if these codecs are used.

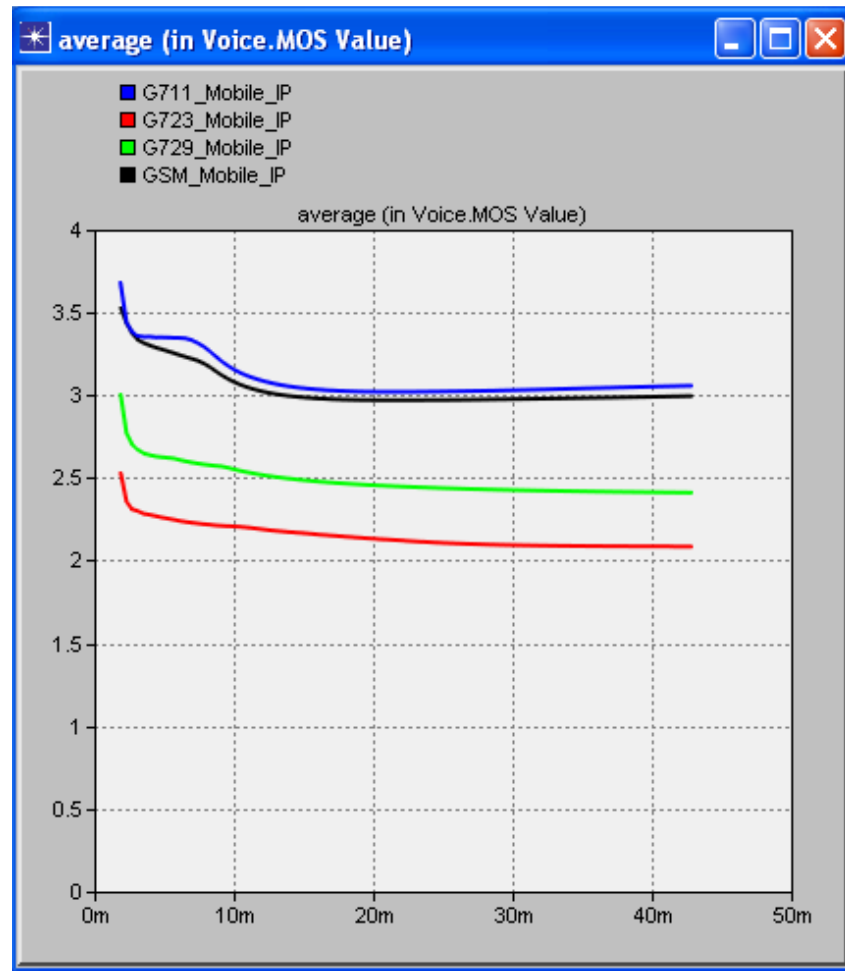


Figure 6. Average voice MOS under various codecs.

## 7. Conclusion

In this study, OPNET Modeler was occupied as a simulation tool for testing audio codecs impact on quality of VoIP based on IEEE802.16e. The proposed handover for IEEE802.16e was validated in various simulation scenarios using four audio codecs. The obtained results show that the acceptable MOS value was recorded for audio codecs G.711 with a value of 3.1, followed by GSM-FR codecs with a value of 3. Audio codecs G.723.1 and GSM-FR recorded lower effect on handover, i.e. the best performance. Additionally, in traffics sent and received, they have small deviation. In terms of end to end delay, G.729A codec recorded a small delay comparing with other audio codecs.

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