The Effect of Mobility on the Performance of VoIP Application in WiMAX Networks

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***Abstract* – in wireless networks, mobility is an important issue because Internet connectivity can only be effective if it is available during the movement of a node. To enhance mobility, the IEEE802.16e standard has set procedures to design wireless access systems to operate on the move without any disruption of services. The major difference between mobile WiMAX and fixed WiMAX is mobility support. However, speed and trajectory of nodes are unpredictable and can vary even in identical circumstances.**

In this paper mobile node trajectory is studied under different codecs schemes in order to evaluate the effect of node movement, towards or away from the base-station, on end to end delay, jitter, mean opinion score and throughput of a VoIP application.

The results showed that trajectory inward improves network performance since mobile node gets closer to the base-station and hence signal is improved and less power is required.

***Index Terms*—trajectory, jitter, throughput, VoIP, Codecs, mobility, WIMAX.**

I. Introduction

WiMAX stands for Worldwide Interoperability for Microwave Access. It embodies the IEEE802.16 family of standards that provide wireless broadband access to residential and commercial Internet subscribers. It provides many benefits to the wireless technologies like high data rate, larger coverage area and offers several classes of Quality of service (QoS) to customers [1].

Due to the growing demand for newer applications that require mobility features as well as the very rapid advancement in technology, the IEEE802.16 workgroup presented the standard version based on the IEEE802.16-2004 standard (fixed version). In order to support such mobile wireless broadband access service which in turn provides high-speed information transmission through, this proposed wireless broadband solution as well as supporting high-speed mobility [2].

Mobile WiMAX performance is influenced by many external factors such as traffic type, network size, traffic load and node mobility (trajectory). It does not provide a great improvement in speed, throughput or capacity. But it provides stable mobile services to portable end user devices, such as laptops and smart phones [3]. The WiMAX system relies on a new radio physical (PHY) layer and appropriate MAC layer to support all demands driven by the target applications.

OFDMA is the modulation adopted on the PHY layer. An optimized resource allocation and support of QoS are provided through a combination with a centralized MAC layer for different classes of services (UGS, e-real-time, real-time, non real-time services and best effort) [2].

Mach and Bestak [4] focused on the importance of node trajectory. Their experimental results showed that throughput is increased when Mobile Stations (MSs) moved towards the BS and it decreased when MSs moved away from the BS.

The performance of VoIP (voice over Internet protocol) calls which are mapped to the BE service class are analyzed and evaluated to investigate the effect and functions of QoS mapping in VoIP applications. Different trajectories were applied to the MS with the G.711 and G.729 encoders in order to identify which encoder gives the best performance to the VoIP application. It was concluded that the G.729 is better than G.711 in terms of QoS parameters and can support up to 80 mobile users [5].

In this paper, the performance of VoIP was evaluated using OPNET simulation package. It has been mapped to UGS service class when different trajectories were applied to different mobile nodes under different codec schemes.

II. VoIP Technology

VoIP is a way to utilize a data network IP to carry voice calls. VoIP carries voice signals as digital packetized signal by converting the original analog voice signal into digitized packets. This process is called encoding and the reverse of this process is called decoding. Both processes are implemented by voice codecs.

A VoIP codec is an algorithm used to encode and decode the voice stream. Different algorithms are used for different codecs to compress and decompress the voice stream. The main difference between the various codecs is the type of modulation and demodulation scheme being adopted [6].

Voice over Internet protocol brings new challenges along with the benefits. Since VoIP has an extreme sensitivity to delay and packet loss in comparison to other network applications such as web and e-mail services, a basic understanding of VoIP traffic and of the quality metrics provided by VoIP monitoring tools will help to keep VoIP network running smoothly [7].

Voice codecs convert an analog voice signal into digitally encoded version. The codecs may vary in sound quality, required bandwidth and computational requirements, etc. [8]. Table (1) shows the commonly used voice codecs with their algorithms, codec delay, bit rates, packet per second and IP packet size.

TABLE 1. VOICE CODECS

IUT-T Codec	Algorithm	Codec Delay (ms)	Bit-Rate (kbps)	Packet Per Second	IP Packet Size (bytes)
G.711	PCM	0.375	64	100	120
G.729	ACELP	35	8	100	50
G.723	CS-ACELP	97.5	5.3	33	60

III. QUALITY OF SERVICE

The term Quality of service refers to the ability to communicate a type of traffic in good conditions, in terms of availability, throughput, transmission delay, jitter, packet loss, and rate etc. It becomes an important factor to support variety of applications that require certain network resources. These applications include multimedia services, voice over IP, etc. [9].

In the field of packet-switched networks and computer networking it is used informally to refer to the probability of a packet succeeding in passing between two points in the network [10].

In VoIP, quality simply means being able to listen and speak in a clear and continuous voice without unwanted noise. The techniques used to measure the voice quality of a VoIP call is the Mean Opinion Score (MOS).

In the realization of VoIP applications. The recommended one way delay and jitter for voice applications is 150 ms and 50 ms respectively [11, 12].

The IEEE802.16 standard divides all services in five different classes. Each group corresponds to a single service class, which is associated with a set of QoS parameters for quantifying the aspects of its behavior. The five service flows are:

1. **Unsolicited Grant Service (UGS):** supports constant bit rate (CBR) or fixed throughput connections at periodic intervals, which guarantees the data throughput and the latency. This type of service is used for T1/E1 and voice over IP, which needs to grant a constant bandwidth without any request.
2. **Extended real-time Polling Service (ertPS):** has a variable bit rate, which is used for real-time applications that have data rate and delay requirements. An example is VoIP with silence suppression.
3. **Real time Polling Service (rtPS):** is a real time data stream comprising variable bit rate (VBR) data packets which are issued at periodic intervals, such as the Moving Pictures Experts Group (MPEG) video
4. **Non-real time Polling Service (nrtPS):** is a delay tolerant data stream consisting of variable sized data packets, suitable for applications such as File Transfer Protocol (FTP).
5. **Best Effort (BE):** does not provide any QoS guarantee, mainly used by applications like email or the short-length FTP [13].

IV. NODE MOBILITY

In wireless networks, mobility is an important issue because Internet connectivity can only be effective if it is available during the movement of a node. Therefore, to enhance node mobility, the IEEE802.16e standard has set procedures to design wireless access systems to operate on the move without any disruption of services.

While in simple mobility, the subscriber may move at speed up to 60 km/h with brief interruptions during handover, of less than 1 sec. Full mobility supports up to 120 km/h speed and seamless handover having less than 50 ms latency and < 1% packet loss [14]. The major difference between mobile WiMAX and fixed WiMAX is mobility support.

However, speed and trajectory of a node are unpredictable and can vary even in identical circumstances. The speed of mobile nodes has a significant effect on network performance as well as network size, traffic types and traffic loads. Movement of a mobile station in a mobile wireless network, can take different forms. Therefore, mobility models are designed to describe the movement pattern of mobile users. Hence, the protocol performance is determined by these mobility patterns [7].

V. NETWORK SIMULATION MODEL

Testing of the network will be done through simulation using the OPNET_14.5 platform in a number of scenarios that explore the parameters affecting the QoS of the VoIP application.

The scenarios were designed to test the effect of node trajectory on QoS under different codec schemes where VoIP is the application under testing. The topology used is shown in Fig. 1, it consists of 5 fixed Subscriber Stations (SS), five mobile stations, one Base Station (BS), one server and a cloud.

To simulate network in real live five different trajectory types were used in this study and these are:

- Wimax_mobility_scenario_1, in which the MS moves at speed of 29 m/s in irregular circular shape with total accumulated time of 11min and 35s.
- Wimax-mg_ms_crowding, in which MS moves in triangular shape at an average speed of 150m/s with total accumulated time of 3 min and 40s.
- Wimax-light_dense_scanning, in which MS moves in irregular circular shape with an average speed of 33 m/s in total accumulated time of 14 min and 36s.
- Move_out_of_range_further, in which MS moves further away from BS at an average speed of 5 m/s in total accumulated time of 37 min and 42s.
- Wimax-mg_connectivity_loss, in which the MS moves at an average speed of 38 m/s at the edge of the coverage area in a total accumulated time of 7 min and 17s.

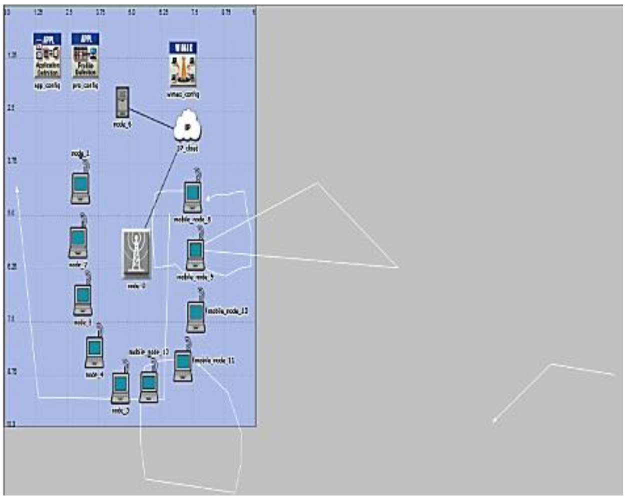
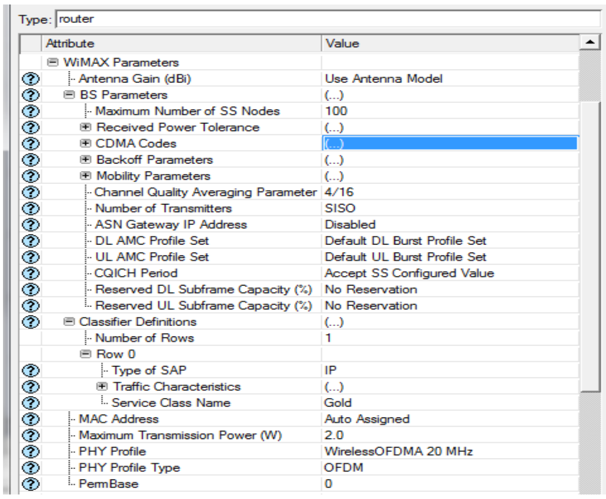


Figure 1. Network topology with trajectories.

In order to run the simulation, parameters for each node (BS, SS and server) will be set, these settings could be seen in Fig. 2, Fig. 3 and Fig. 4 respectively.

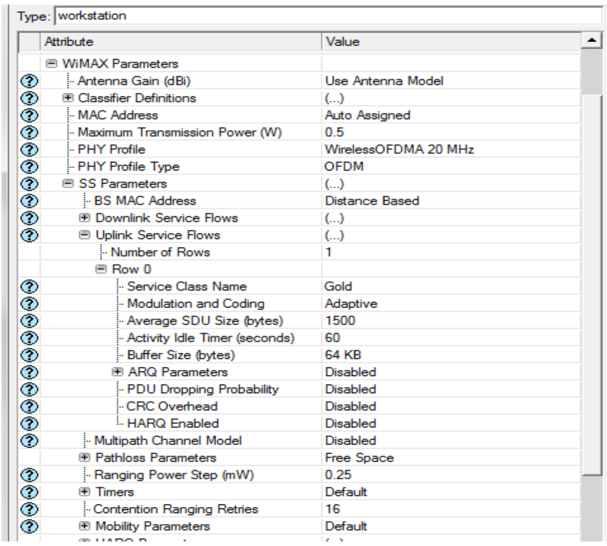


Type: router

Attribute	Value
WiMAX Parameters	
Antenna Gain (dBi)	Use Antenna Model
BS Parameters	(...)
Maximum Number of SS Nodes	100
Received Power Tolerance	(...)
CDMA Codes	(...)
Backoff Parameters	(...)
Mobility Parameters	(...)
Channel Quality Averaging Parameter	4/16
Number of Transmitters	SISO
ASN Gateway IP Address	Disabled
DL AMC Profile Set	Default DL Burst Profile Set
UL AMC Profile Set	Default UL Burst Profile Set
CQICH Period	Accept SS Configured Value
Reserved DL Subframe Capacity (%)	No Reservation
Reserved UL Subframe Capacity (%)	No Reservation
Classifier Definitions	(...)
Number of Rows	1
Row 0	
Type of SAP	IP
Traffic Characteristics	(...)
Service Class Name	Gold
MAC Address	Auto Assigned
Maximum Transmission Power (W)	2.0
PHY Profile	WirelessOFDMA 20 MHz
PHY Profile Type	OFDM
PermBase	0

Figure 2. Base Station parameters.

The "Application Config" node is used to specify the applications or to create new applications that will run during the simulation. Where every new application is given a name and possibly a description.

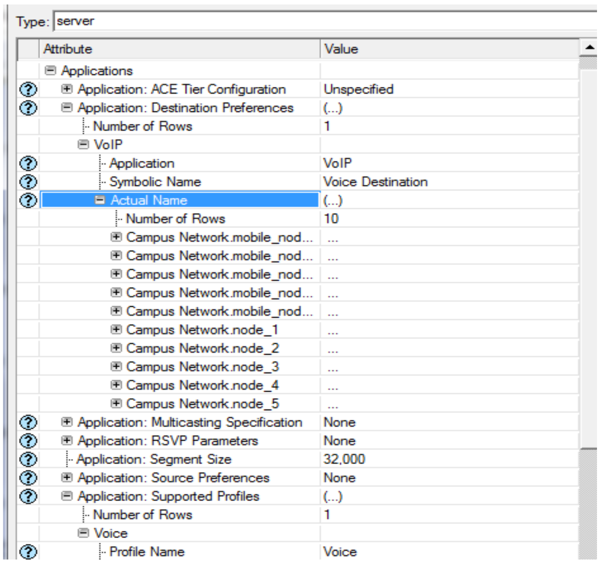


Type: workstation

Attribute	Value
WiMAX Parameters	
Antenna Gain (dBi)	Use Antenna Model
Classifier Definitions	(...)
MAC Address	Auto Assigned
Maximum Transmission Power (W)	0.5
PHY Profile	WirelessOFDMA 20 MHz
PHY Profile Type	OFDM
SS Parameters	(...)
BS MAC Address	Distance Based
Downlink Service Flows	(...)
Uplink Service Flows	(...)
Number of Rows	1
Row 0	
Service Class Name	Gold
Modulation and Coding	Adaptive
Average SDU Size (bytes)	1500
Activity Idle Timer (seconds)	60
Buffer Size (bytes)	64 KB
ARQ Parameters	Disabled
PDU Dropping Probability	Disabled
CRC Overhead	Disabled
HARQ Enabled	Disabled
Multipath Channel Model	Disabled
Pathloss Parameters	Free Space
Ranging Power Step (mW)	0.25
Timers	Default
Contention Ranging Retries	16
Mobility Parameters	Default

Figure 3. Subscriber station parameters.

The "Profile Config" node is used to create user profiles each can define a user(s) behavior. These profiles can then be applied to different nodes in the network to generate application layer traffic. As the applications defined in the "Application Config" objects are used by this object, therefore applications must be created using the "Application Config" object before creating and using this object.



Type: server

Attribute	Value
Applications	
Application: ACE Tier Configuration	Unspecified
Application: Destination Preferences	(...)
Number of Rows	1
VoIP	
Application	VoIP
Symbolic Name	Voice Destination
Actual Name	(...)
Number of Rows	10
Campus Network.mobile_nod...	...
Campus Network.mobile_nod...	...
Campus Network.mobile_nod...	...
Campus Network.mobile_nod...	...
Campus Network.mobile_nod...	...
Campus Network.node_1	...
Campus Network.node_2	...
Campus Network.node_3	...
Campus Network.node_4	...
Campus Network.node_5	...
Application: Multicasting Specification	None
Application: RSVP Parameters	None
Application: Segment Size	32,000
Application: Source Preferences	None
Application: Supported Profiles	(...)
Number of Rows	1
Voice	
Profile Name	Voice

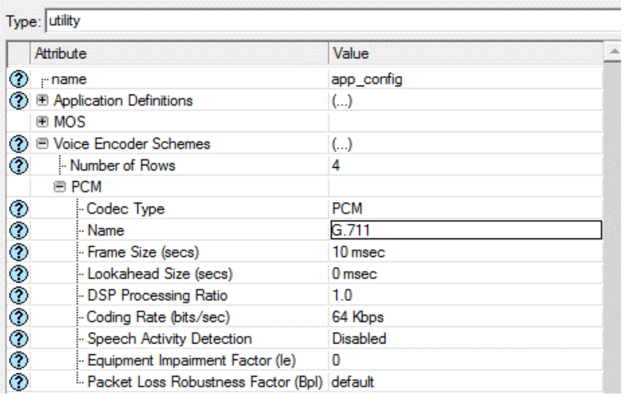
Figure 4. Server parameters

The service class used for all scenarios is the unsolicited grant service since it is the recommended class for VoIP by the IEEE802.16 standard. Six scenarios were implemented in this study, all used the codecs G711, G729A and G723. Three of them without applying trajectories to the mobile nodes and the other three scenarios used the same codecs but different trajectories were applied to the mobile nodes. That is to check the effect of mobility on different QoS parameters. Namely, the jitter, end to end delay, MOS, packet delay variation and throughput. Table (2) below details the parameters of the WiMAX that have been used in this test.

TABLE 2. WIMAX CONFIGURATION PARAMETERS

Parameter	Value
Scheduling class	UGS
Codecs	G711, G729A, G723
Max. latency	30 ms
Efficiency mode	Mobility and Ranging
Profile	OFDMA 20 MHz
Frame duration	5 ms
Symbol duration	102.86 ms
Number of subcarriers	2048
Duplexing mode	TDD

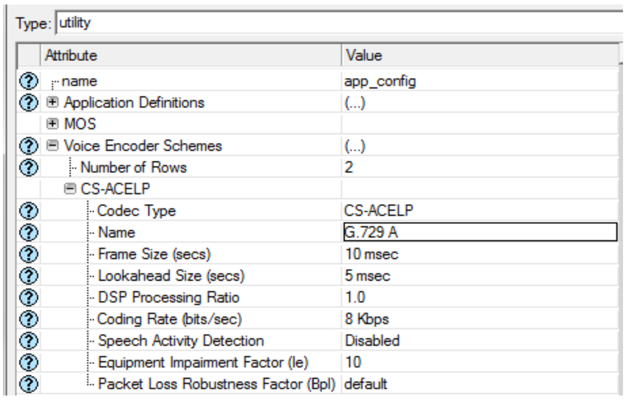
Experiment 1: Voice quality is important for VoIP systems because of the high demands of users' for good quality voice services. In these scenarios, we considered the use of various voice codecs in the same WiMAX network in order to investigate their performance. The codecs parameters set in this experiment for the codec G.711 is shown in Fig. 5, for G.729 in Fig 6 and for G.723 in Fig. 7.



Type: utility

Attribute	Value
name	app_config
Application Definitions	(...)
MOS	
Voice Encoder Schemes	(...)
Number of Rows	4
PCM	
Codec Type	PCM
Name	G.711
Frame Size (secs)	10 msec
Lookahead Size (secs)	0 msec
DSP Processing Ratio	1.0
Coding Rate (bits/sec)	64 Kbps
Speech Activity Detection	Disabled
Equipment Impairment Factor (Ie)	0
Packet Loss Robustness Factor (Bpl)	default

Figure 5. G.711 codec parameters.

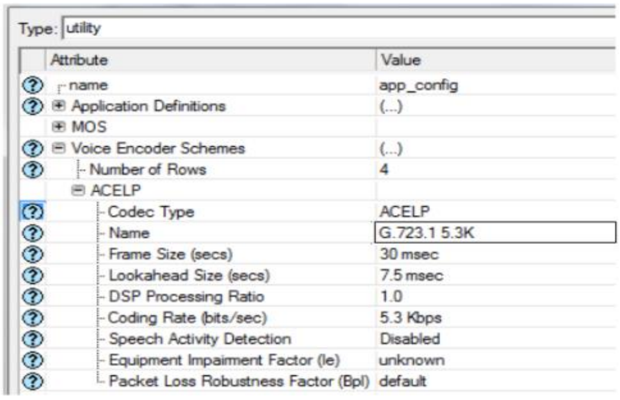


Type: utility

Attribute	Value
name	app_config
Application Definitions	(...)
MOS	
Voice Encoder Schemes	(...)
Number of Rows	2
CS-ACELP	
Codec Type	CS-ACELP
Name	G.729 A
Frame Size (secs)	10 msec
Lookahead Size (secs)	5 msec
DSP Processing Ratio	1.0
Coding Rate (bits/sec)	8 Kbps
Speech Activity Detection	Disabled
Equipment Impairment Factor (Ie)	10
Packet Loss Robustness Factor (Bpl)	default

Figure 6. G.729 codec parameters

Experiment 2: In this experiment, different scenarios were used with simulation setup similar to experiment 1 except that the nodes are now mobile. That is to investigate the impact of trajectory on VoIP performance under the three codecs schemes identified previously.



Type: utility

Attribute	Value
name	app_config
Application Definitions	(...)
MOS	
Voice Encoder Schemes	(...)
Number of Rows	4
ACELP	
Codec Type	ACELP
Name	G.723.1 5.3K
Frame Size (secs)	30 msec
Lookahead Size (secs)	7.5 msec
DSP Processing Ratio	1.0
Coding Rate (bits/sec)	5.3 Kbps
Speech Activity Detection	Disabled
Equipment Impairment Factor (Ie)	unknown
Packet Loss Robustness Factor (Bpl)	default

Figure 7. G.723 codec parameters.

VI. SIMULATION RESULTS AND ANALYSIS

In this section, the chosen performance parameters of VoIP in WiMAX will be presented. And the following subdivisions detailed the results obtained through extensive simulations. The graphs below show the effect of the movement of mobile nodes in a predefined trajectory on the QoS of the test WiMAX network topology under different codecs schemes. The parameters that have been chosen for comparison are those likely to affect the QoS of the running application (VoIP) on the test network.

A. *End to End Delay*

From Fig. 8 it can be seen that the node trajectory has a minimal effect on the end-to-end delay for the VoIP application under the three implemented codecs schemes. While for both G711 and G723 there is no effect on delay at all. The delay for the codec G729A has improved slightly as a result of the node trajectory.

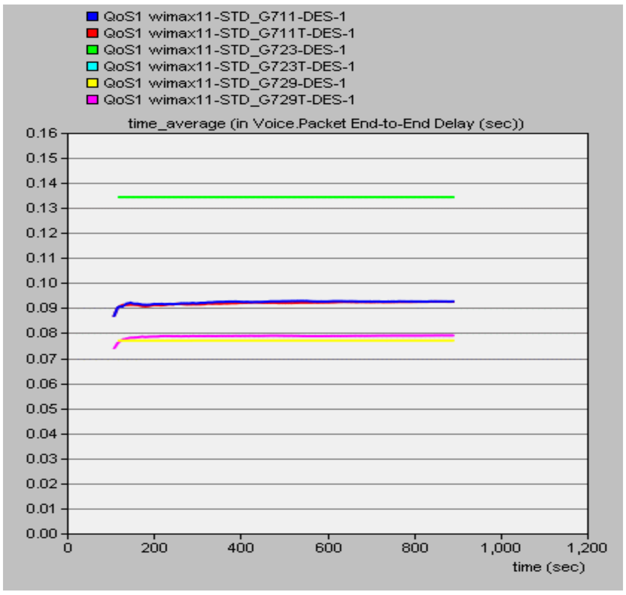


Figure 8. End to end delay.

B. *Voice Jitter*

It is quite clear from Fig. 9, as far as jitter is concerned, that the effect of node trajectory is not significant except for the G729A codec where it reduces jitter marginally.

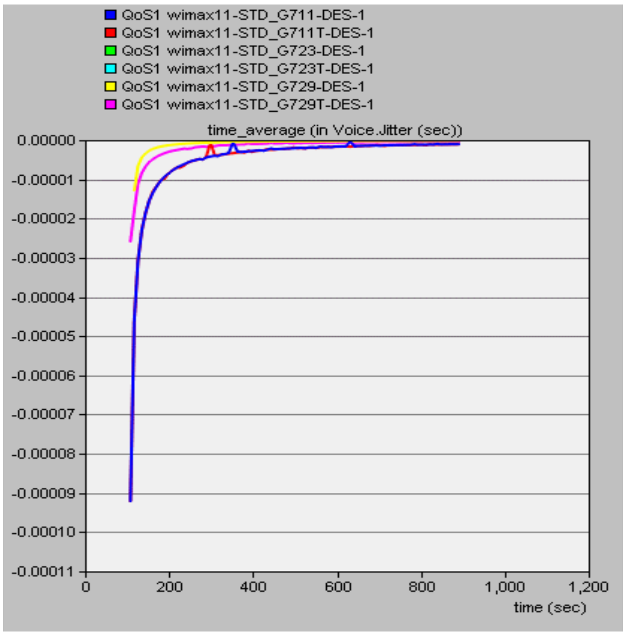


Figure 9. Voice jitter

C. *Mean Opinion Score*

For this parameter, it can be seen from Fig. 10 that there is no change for the G711 codec. But for the G729 there is a little improvement in the MOS value due to nodes trajectory but for the G723 codec the nodes trajectory made the MOS value worse.

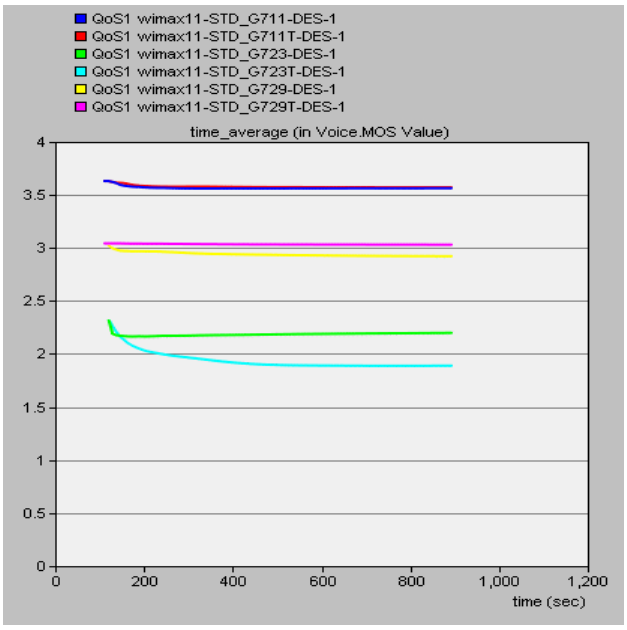


Figure 10. MOS Value

D. *Packet Delay Variation*

Fig. 11 shows that the G711 trajectory case has a lower delay variation value whereas the G729A has the reverse effect where the trajectory case has increased the packet delay variation notably. On the other hand, it has no effect on the G723 codec.

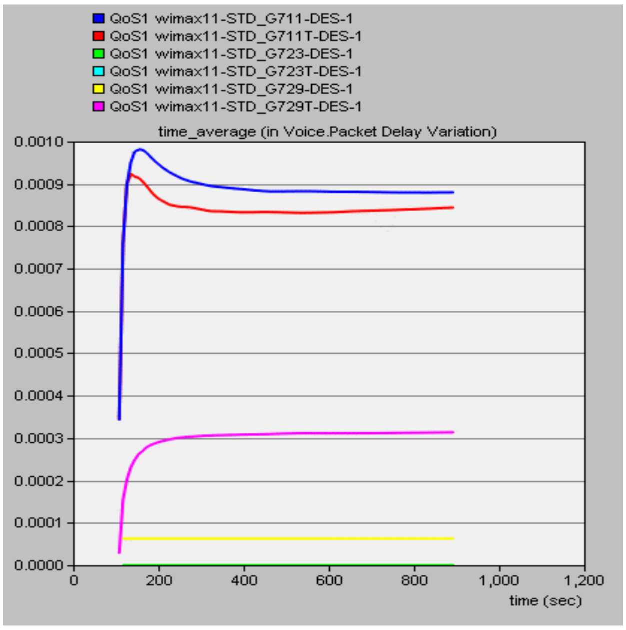


Figure 11. Packet Delay Variation.

E. *Throughput*

Fig. 12 confirms that the nodes trajectory did not induce any effect network throughput in the case of G711 and G723 codecs. But, it improves the throughput significantly for the G729A codec.

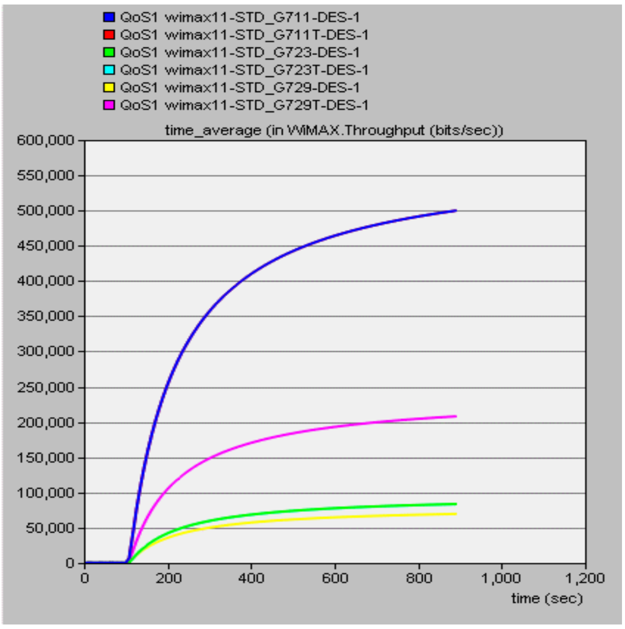


Figure 12. Throughput (packets/sec)

VII. CONCLUSION AND FUTURE WORK

In this paper, OPNET, the simulation tool was used to test the effect of nodes movement in predefined trajectories on the quality of VoIP. A WiMAX network was used to deploy the voice application and test its quality for three different voice codecs schemes. Six scenarios were implemented. Results were collected to inspect the effect of node trajectory on the following QoS parameters: end-to-end delay, jitter, MOS, packet delay variation and throughput.

The results suggested that G729A codec is the one that mostly affected by the node trajectory. As most of the parameters been tested for G729A improved in general. Based on the obtained results, it is clear that the effect of node trajectory is mainly influenced by the type of codec being applied.

So, to deteriorate the effect of node movement on the performance of the application, we recommend to reduce the size and load of the network i.e. smaller coverage areas. And adapt QoS priority as well as different modulation and coding schemes.

REFERENCES

[1] Bassam F.Gumaidah, Hasan H.Soliman and M.Obayya, "Study the Effect of Base Frequency on the Performance of WiMAX Network Carrying Voice", Vol.4, No.4, July 2012.

[2] Nongjun Li, "Overview of WiMax Technical and Application Analysis",2011.

[3] James Kangwook, " A Study of the Impact of Traffic Type and Node Mobility on the Performance of an IEEE 802.16 WiMAX", 2011.

[4] P Mach, R Bestak," WiMAX performance evaluation ",Networking, 2007. ICN'07. Sixth International Conference on, 17-17

[5] Hazri Raziff Othman, Darmawaty M. Ali, Nurul Anis Mohd Yusof, Ku Siti Syahidah Ku Mohd Noh, Azlina Idris, " Performance Analysis of VoIP over Mobile WiMAX IEEE 802.16e Best Effort Class" , IEEE 5th Control and System Graduate Research Colloquium, Aug. 11 - 12, 2014, UiTM, Shah Alam, Malaysia

[6] Priyanka Grover, Meenakshi Chawla,"Performance Analysis Of VOIP Codec's With QoS Parameters",; Volume 2, Issue 5, May - 2015.

[7] Antonio Vilei, Gabriella Convertino, Fabrizio Crudo – "A New UPnP Architecture for Distributed Video Voice over IP,Proceedings of the 5th international conference on Mobile and ubiquitous multimedia, Stanford", California, 2006.

[8] M. Tariq, M. Azad, and S. Rizvi, ''Effect of Mobility Patterns on VoIP QoS in Mobile WiMAX'', International Journal of Computer Science and Telecommunications, pp. 1-7, Volume 4, Issue 1, January, 2013.

[9] P. Rengaraju, C.H. Lung, A. Srinivasan, R.H.M. Hafez, "Qos Improvements in Mobile WiMAX Networks", AHU J. of Engineering & Applied Sciences, Vol. 3, Issue 1, 2010.

[10] Rohit Talwalkar. Rohit Talwalkar,"Analysis of Quality of Service (QoS) in WiMAX Networks", May, 2008

[11] Karan tej Singh, Mohinder Singh and Shanu Malhotra;"A Survey on QoS classes and encoding schemes over WiMAX ";; Volume 3, Issue 5, May 2014

[12] K. Deergha Rao, "Channel Coding Techniques for Wireless Communications", Springer New Delhi, 2015.

[13] Elmabruk M. Laias; "Design, analysis and evaluation of 802.16 Point-to-Multipoint (PMP) Quality of Service Framework based on uplink scheduler and call admission control Analysis"; 2009.

[14] Kranti Bala, Kiran Ahuja ,"Impact of Mobility On QoS of Mobile WiMax Network With CBR Application", Vol 2, No 3 July 2011

BIOGRAPHY

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