



Master's Thesis :

**AUDIO CODECS IMPACT ON QUALITY OF
VOIP OVER MOBILE WIMAX NETWORKS IN
HANDOVER PROCESS**

BY :

SALEH SAAD GARASH

Libyan Academy

Tripoli – Libya



**AUDIO CODECS IMPACT ON QUALITY OF VOIP OVER MOBILE WIMAX
NETWORKS IN HANDOVER PROCESS**

By

SALEH SAAD GARASH

B.Sc. in computer Engineering, Zawia University, 2014

Supervisor

Dr. Adel ali Eluheshi

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DECLARATION

I declare that this thesis, which is being provided to Libyan Academy, is carried out by me in partial achievement of the degree of Master of engineering. In addition, I certify that this thesis is resultant of my private work and it was never submitted before by anyone partially or completely to obtain a degree or any other qualification for any higher educational organization.

The preparation of this thesis is according to Libyan academy regulations for postgraduate studies: computer and electrical engineering Department.

Eng. Salih saad garash
salih.garash@academy.edu.ly
December 2017, Tripoli, Libya

DEDICATION

I dedicate this work to my father and my mother who educated me that love and mercy is the secret of the happy life .

To my wife **REMA** who was always beside me pushing towards innovation.

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Foremost, I fully thank the Allah the most gracious and the most merciful for the grace of faith and for helping me on the hardship of searching and for giving me the confidence and patience to finish this thesis, Today is the day of writing gratitude and appreciation memorandum that are the last touches in my thesis, I would like to thank the people who supported and helped me too much over this period.

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المـلـخـص

تكنولوجيا ويماكس هي واحدة من التقنيات اللاسلكية الناشئة, التي توفر لنا البيانات المتنقلة وخدمات الاتصالات السلكية واللاسلكية عالية السرعة, والاعتبار الرئيسي من ويماكس المتنقلة هو تحقيق تسليم سلس, بحيث لا يوجد فقدان للبيانات, الصوت عبر بروتوكول الإنترنت (VoIP) هو تكنولوجيا لنقل الصوت والوسائط المتعددة عبر بروتوكول الإنترنت (IP), كان يستخدم على نطاق واسع كبروتوكول اتصالات ليحل محل تكنولوجيا الهاتف التقليدية, شبكة الهاتف العمومية المحولة (PSTN).

جودة الصوت مهمة لنظام الاتصالات عبر بروتوكول الإنترنت (VoIP), بسبب زيادة طلب المستخدمين لخدمات الصوت ذات جودة جيدة, احتياجات مستخدمي الإنترنت اليوم في زيادة لإجراء المكالمات عبر (VoIP) بواسطة تطبيقات برمجية متخصصة مثل Skype, Viber, Google Hangouts و WhatsApp لأنها رخيصة, وذات جودة مقبولة.

يشير التسليم Handover إلى عملية نقل المكالمات من المحطة الأساسية (BS) إلى محطة اخرى, وأثناء تسليم المكالمات يحدث فقدان للبيانات.

في هذه الاطروحة, تم التحقيق في أداء برامج الترميز عبر بروتوكول الإنترنت الأكثر شيوعا, وهي G.711, G.723.1, G.729 و GSM-FR عبر شبكة واي ماكس المتنقلة أثناء عملية التسليم Handover والازدحام Congestion, وذلك باستخدام معلمات مختلفة تشير إلى جودة الصوت عبر بروتوكول الإنترنت مثل: التأخير, مؤشر قياس نوعية الكلام, تغير التأخير و الإنتاجية. وقد أجريت هذه المكالمات في حين كانت سرعة العقدة ثابتة أو متغيرة بسرعات مختلفة. تم بناء بيئة الاختبار باستخدام نموذج أونبيت, من نتائج المحاكاة التي تم الحصول عليها. تظهر نتيجة المحاكاة أن أداء الترميز G.711 أفضل من أربعة برامج الترميز الأخرى أثناء تغيير السرعات والازدحام, يتم تسجيل قيمة مقبولة لمؤشر قياس نوعية الكلام والإنتاجية في الترميز G.711 مع ملاحظة زيادة سرعة العقد المتنقلة تؤدي إلى انخفاض معدلات الإنتاجية وزيادة قيمة مؤشر قياس نوعية الكلام (MOS).

ABSTRACT

WiMAX Technology is one of the emerging wireless technology that provide us high speed mobile data and telecommunication services ,The main consideration of Mobile WiMAX is to achieve seamless handover such that there is no loss of data, Voice over Internet Protocol (VoIP) is technology for the transmission of voice and multimedia over the Internet Protocol (IP), has been widely in use as a communication protocol to replace traditional telephone technology, Public Switched Telephone Network (PSTN).

Voice quality is important for VoIP system because of users' high demands for good quality voice services,).The needs of Internet users today in increase for Procedure VoIP calls via specialized software applications such as skype, Viber, Google Hangouts and WhatsApp. Since it's a cheap and has accepted QoS,

Handover refers to the process of call transfer from one base station to another BS, during handover a call is loss of data.

This thesis investigates the performance of the most common VoIP codecs, which are G.711, G.723.1, G.729 and GSM FR over a Mobile WiMAX network during handover process and Congestion, using Different parameters that indicate the quality of VoIP such as like Jitter, MOS, end-to-end delay and Throughput. These Calls had taken place while the node speed was Fixed or Variable with different speeds. The testing environment were built using OPNET modeler, the simulation results obtained, show that the performance of the G.711 codec is better than other codecs during the change of speeds and congestion, acceptable value for MOS and throughput is recorded for codecs G.711, noting that increased mobile nodes speed give rise to lower throughput rates and Increasing value of MOS.

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LIST OF SYMBOLS

Acronym	Definition
WiMAX	Worldwide Interoperability for Microwave Access
4G	Fourth Generation
ASN	Access Service Network
ASN-GW	Access Service Network Gateway
BS	Base Station
DL	Downlink
FDD	Frequency Division Duplexing
GSM	Global System for Mobile communications
MIMO	Multiple Input Multiple Output
PHY	Physical Layer
MS	Mobile Station
IP	Internet Protocol
OFDMA	Orthogonal Frequency Division Multiple Access
VoIP	Voice over IP
QoS	Quality of Service
NRM	network reference model
MAC	Media Access Control
MOS	Mean Opinon Score
TDD	Time Division Duplex
RSVP Protocol	Resource Reservation
DiffServ	Differentiated Services
TDD	Time Division Duplex
PCM	Pulse code modulation
PSTN	public switched telephone network
UGS	Unsolicited Grant service
RTPS	Real-time Polling Service
BE	Best effort

CHAPTER 1
INTRODUCTION

1.1 INTRODUCTION

In 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. In recent researches more attention was paid to increase the handover schemes efficiency. Mobile WiMAX allows the user to move freely during data transmission. The main consideration of mobile WiMAX is that there should be no data loss when the moving user switches from one base station to another i.e. during handover. Handover is procedure when a mobile station changes the serving base station. The reason for handover could be relatively low signal strength or work load of base station. [2]

Large demand refers to large decrease into costs of installation (set up) and operation of telecommunication via internet protocol compared to traditional telephone networks (PSTN) [3]. In addition, VoIP introduces huge and various group of services, such as Voice Call with low cost and the integration of data with possibility to combine technologies of voice, image and data spontaneously, also flexibility, graduation and simplicity of usage , These advantages make telecommunication via internet protocol more important into business field. Many VoIP applications are available on the internet such as Skype, Viber, Tango, and Yahoo messenger [4]. All of these applications provide good quality, and free calls. In VoIP, the analogue voice signal of the transmitter is converted into digital format before compression and encoding it into a stream of IP packets for transmission to the receiver over IP network .At the receiving end, Digital to Analogue Converter (DAC)

works on regenerating the original analogue voice signal after reassemble received IP packets in order and processing it [5].

Communication applications via internet protocol face some challenges. These challenges are loss of packets and delay. This might lower the speech quality; the objective of the present thesis is to investigate Handover effect and Congestion on Quality of VoIP Based on Mobile WiMAX among various audio codec schemes.

1.2 OBJECTIVES OF THE RESEARCH

The main objective of the research is to improve the Quality of VoIP over mobile WiMAX Networks in Handover process; the following are the objectives of this project:

- ❖ To provide a detailed study about WIMAX networks.
- ❖ Study the effect of handover on Quality VIOP.
- ❖ investigate effect of speed change on mobile node over VIOP.
- ❖ Study the effect Congestion on audio codecs.
- ❖ choosing the best Audio Codec is he behind this study and simulation.

1.3 AIM OF THE STUDY

Investigate the impact of audio codecs on quality of VoIP over mobile WIMAX Networks in handover process and Congestion.

1.4 LITERATURE REVIEW

There have been recent studies focusing on the analyzing and evaluating the performance of audio codecs in wireless networks, concentrating on WiMAX and handover process

1. Title:” *Performance Analysis Of VoIP Traffic Over Integrating wireless LAN And WAN Using Different Codecs* ”, by Ali M. Alsahlany, International Journal of

Wireless & Mobile Networks (IJWMN) Vol. 6, No. 3, June 2014 .

This paper studied analysis and evaluation of VoIP performance based on integrated wireless LAN / WAN in frame of voice codecs systems (G.711,G.723,G.729) the different parameters which indicate service quality were estimated and the network model was simulated by using OPNET modeler. The overall results indicated that codec G.729 offered great result into VoIP performance since G.729 has an accepted MOS value and is less deviation related to transmitted packets to received packets also E.T.E delay and voice jitter was lesser into codec G.729 compared to other two codecs indicated to it here.

Relevance to this research: The above mentioned paper studied three audio codecs (G.7.11, G723, G.729), Two of audio codecs (G711, G729) that are the same in this research, but there is a new codec will be tested and studied named GSM, Also in The above-mentioned paper the transmissions taken place while the nodes are stationary. In this research, the transmissions will take place while the nodes are stationary or moving with different speeds .and in this research. Is done test impact audio codecs on quality of VoIP during Congestion and handover process.

2. Title:” *Audio Codecs Impact on Quality of VoIP Based on IEEE802.16e Considering Mobile IP Handover*”, by Hayder S. Rashid Ali Mohammed, American Journal of Networks and Communications, Vol. 4, No. 3, 2015, pp. 59-66.

This paper 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.

Relevance to this research: The above mentioned paper studied four audio codecs (G.7.11, G723, G.729, GSM), that are the same in this research. also in the above-mentioned paper the transmissions taken place while the nodes are stationary or moving with Fixed speeds .and in this research the transmissions will take place while the nodes are moving with different speeds during Congestion.

3. Title:” *QoS of VoIP over WiMAX Access Networks*”, by S. Alshomrani, S. Qamar, S. Jan, I. Khan and I. A. Shah, International Journal of Computer Science and Telecommunications, Volume 3, Issue 4, April 2012.

This paper simulated VoIP performance evaluation Via WiMAX network. since it used different parameters such as jitter, MOS, packet sent and received and packet delay for measuring of voice transmission performance Via WiMAX technology. then it simulated three voice systems G.729, G.723 and G.711 in order to find out the appropriate codec .To transmit voice Via WiMAX network the results showed the simulation into this paper that communication Via internet protocol performs better under code G.711 compared to G. 723 and G.729 . research results also showed that VoIP applications might presented in better way under exponential traffic distribution.

Relevance to this research: The above mentioned paper investigates the performance of VoIP traffic studied three audio codecs (G.7.11, G723, G.729), but a new codec, GSM is not tested which will be used in this research for Evaluation its performance in various QoS parameters, QoS parameters in The above mentioned paper are MOS, Jitter, Packet end-to- end delay, Traffic sent and received. in this research, Throughput was added .and study Audio Codecs Impact on Quality of VoIP in handover process and Congestion.

4. Title:” *Analysis Performance VoIP Codecs over WiMAX Access Network*”, by Gysberth M.Wattimena, International Journal of Advanced Research in Computer Science and Electronics Engineering (IJARCSEE) Volume 1, Issue 7, September 2012.

In this paper Simulations were conducted to evaluate performance of VOIP over WIMAX networks, for this purpose OPNET 14.5 have been used as the simulation platform. The parameters in this simulation are views jitter, packet end to end delay and throughput. Three voice codecs G.711, G.723 and G.729 is the parameters of this experiment. Comparative results between the three scenarios in this experiment give a performance VOIP over WIMAX network. Simulation Result shows that the performance of the G.711 codec is better than other codecs if seen from throughput value of which is derived. The greater the number of subscriber station, the greater the value of throughput for the G711 codec

Relevance to this research: The above mentioned paper studied three audio codecs (G.711, G723, G729) , That is the same in this research, but there is a new codec will be tested and studied named GSM. Also in The above-mentioned paper the

transmissions taken place while the nodes are stationary through a WIMAX network. In this research, the transmissions will take place while the nodes are moving with different speeds through handover process and Congestion.

1.5 PROBLEM STATEMENT

The existing WIMAX wireless networks offer flexibility to support real _ time application such as VOIP , while the IEEE 802.16 (WIMAX) provides large coverage area (approximately 50 km) and high data rate (up to 75 mbps) using radio link. VOIP is an application which carries the real time voice data over the internet which is highly delay intolerant and needs a high priority transmission ,the term handover or handoff refers to the process of transferring an ongoing call or data session from one channel connected to the core network to another. In WIMAX both mobile station (MS) and base station (BS) scans the neighboring base stations for selecting the best base station for a potential handover.

Internet communication is constantly evolving and growing, Where the number of users to serve VoIP in increment, It has become a competition for traditional telecommunications service GSM , The user wants to get best quality of communication , Operation handover during a call is Result loss of data, Thus Lack of quality, The study problem can be identified in the following question:

Which voice codecs offer the best QoS of VoIP during Congestion over WIMAX mobile?

- the question is addressed in this thesis by doing measurement analysis of VOIP over WIMAX through Congestion and handover.
- the approach adopted is based on simulation using the well know networking research simulation tool OPNET.

1.6 PROPOSAL SOLUTION

Voice quality is important for VoIP system because of the users high demands for good quality voice services; the main objective of the thesis is to improve the Quality of VoIP over mobile WIMAX Networks in Handover process. in this thesis a

simulation model using OPNET tool is introduced for testing four audio codecs (G.711, G.723.1, G.729 and GSM FR) During handover and Congestion, for each simulation scenario the Qos metrics is measured , using Different parameters such as Jitter, MOS, end-to-end delay, Throughput , The quality of the voice has been measured at different speeds for mobile nodes over WIMAX network in handover process and Congestion, the purpose was to find the best audio codec Provide Qos high in the voice over IP (VOIP).

1.7 THESIS OUTLINE

This thesis can be considered a comprehensive report of this research, it is divided into six Chapters, and the outline of this thesis research is organized as following:

- Chapter 1: an introduction to the present research. It consists of general information of the background, objectives, and problem statement of this thesis.
- Chapter 2: Related Works, This Chapter will cover the overview of WIMAX networks and their architectures, followed by presenting the overview of the handover process in WIMAX.
- Chapter 3: discusses the issues that arise when using VOIP over wireless networks.
- Chapter 4: Simulation Design and Implementation.
- chapter 5: clarifies and discusses the simulation results that obtained from the performance evaluation of different audio codecs that are applied in all scenarios.
- Chapter 6: concludes the thesis and present future work.

CHAPTER 2
RELATED WORK

2.0 INTRODUCTION

Wireless communication technology is a modern alternative to traditional wired networking that relies on cables to connect digital devices together. Wireless technologies are widely used in both home and business computer networks. A large number of technologies have been developed to support wireless networking in different usages. Mainstream wireless technologies include: Wi-Fi, Bluetooth, 3G and 4G cellular Internet. [6]. The acoustic communications today in the frame of wireless environment acquire publicity increasingly as alternative to lines of terrain cables (DSL Line), since it can be installed in places in which the expansion of cables is very difficult. In the last few years most of voice communications and exchange of files and media Via computers and Mobiles and this means communication is in wireless environment in which the user enjoys potentiality of movement and motion into sensitive places.[7]

In this chapter we will provide an overview about concept of wireless networks, Then we will discuss a WIMAX network technical (IEEE 802.16), In terms of structure and characteristics, finally, overview of the handover process in WIMAX networks.

2.1 WIRELESS NETWORKS

One of the most transformative technology trends of the past decade is the availability and growing expectation of ubiquitous connectivity, whether it is for checking email, carrying a voice conversation, web browsing, or myriad other use cases, Wireless networks are at the epicenter of this trend. At its broadest, a wireless network refers to any network not connected by cables, which is what enables the desired convenience and mobility for the user, wireless technologies in use: Wi-Fi, WIMAX, LTE, earlier 3G standards, satellite services, and more. The arrival of wireless technology has reduced the human efforts for accessing data at various locations by replacing wired infrastructure with wireless infrastructure and also providing access to devices having mobility. [8]

2.1.1 Types of Wireless Networks

A wireless network is any kind of computer network that is connected wirelessly, meaning that the nodes are connected to each other or to the telecommunications network (which connected them to the internet or backbone wired network) without the need of wires. Wireless networks use the electromagnetic waves (commonly radio waves) for carrying the signals and data between the nodes and it is implemented at the physical layer meant to replace the wires. partition on the basis of the geographic range to:

- **WLANS: Wireless Local Area Networks:**

Wireless Local Area Network (WLAN) is an alternative way to connect computers and devices in Local Area Network (LAN) by using radio waves, while LAN technology uses Ethernet cable to connect devices and computers for the purpose of communication within a small area such as home, office or a device within one building. The series of 802.11 are referred to as Wireless LAN.

- **WPANS: Wireless Personal Area Networks**

The two current technologies for wireless personal area networks are Infra-Red (IR) and Bluetooth (IEEE 802.15). These will allow the connectivity of personal devices within an area of about 30 feet. However, IR requires a direct line of site and the range is less.

- **WMANS: Wireless Metropolitan Area Networks**

Wireless MAN is used to connect the two or more networks that are a distance away from each other, like in different cities. It is also used to provide the WLAN services, like Internet, to the entire city. WiMAX is usually used as the reference of the Wireless MAN.

- **WWANS: Wireless Wide Area Networks**

Wide area networking (WAN) is linking networks to allow wider file sharing or connectivity. the most common use is to allow connection to the Internet via an internet service provider (ISP). The following table 2.1 shows a comparison of Wireless Network Types.

Table 2.1 comparison of Wireless Network Types[9]

type	coverage	performance	standards	applications
PAN	Within reach of a person	moderate	Wireless PAN Within reach of a person moderate Bluetooth, IEEE 802.15	Cable replacement for peripherals
LAN	Within a building or campus	High	IEEE 802.11 , Wi-Fi and HlperLAN	Mobile extension of wired network
MAN	Within a city	High	Proprietary, IEEE 802.16 ,and WIMAX	Fixed Wireless between home and businesses and the internet
WAN	worldwide	Low	CDPD and cellular 2G,2.5G and 3G	Mobile access to the internet from outdoor areas

2.1.2 Wireless Network Applications

Applications of wireless networks also fall within private or public scenarios. Usually, private applications are made only available for company employees or home occupants. Access to the applications is not made available to the general public. Public applications, on the other hand, provide open access to anyone. A business traveler, for example, can use a public wireless LAN at an airport to access the Internet while waiting for a flight. These public hotspots are becoming widely available in airports and other areas, such as hotels, convention centers, and coffee shops where there are large concentrations of people toting computer devices. The important applications of wireless networks are :Internet Access ,Voice over Wireless ,Inventory Control, Health Care , Education ,Public Networks.

Applications of wireless communication involve security systems, television remote control, Wi-Fi, Cell phones, wireless power transfer, computer interface devices and various wireless communication based projects.

2.2 OVERVIEW OF WIMAX NETWORKS

WIMAX is “World Interoperability for Microwave Access”. It is a standard typically based on global interoperability, including IEEE 802.16d-2004 for fixed, and 802.16e for mobile high-speed data. WiMAX is gaining popularity as a technology which delivers carrier-class, high speed wireless broadband at a much lower cost while covering it is can provide broadband wireless access up to 50 km for fixed stations and (5 - 15 km) for mobile stations. WI-max is expected to provide fixed, nomadic, portable and eventually, mobile wireless broadband connectivity without the need for direct line-of sight (LOS) with a base station. In a typical cell radius deployment of 3 to 10 kilometers, WI-max forum certified systems can be expected to deliver capacity of up to 40 mbps per channel, for fixed and portable access applications .while mobile network deployments are expected to provide up to 15 mbps of capacity within a typical cell radius deployment of up to 3 kilometers. The frequency range for the fixed standard covers 2 to 11 GHz while the mobile standard covers below 6 GHz. Depending on the frequency band, it can be Frequency Division Duplex (FDD) or Time Division Duplex (TDD) configuration. The data rates for the fixed standard will support up to 75 Mbps per subscriber in 20 MHz of spectrum, but typical data rates will be 20 to 30 Mbps. The mobile applications will support 30 Mbps per subscriber, in 10 MHz of spectrum, but typical data rates will be (3 – 5) Mbps. [10]

2.2.1 Background of WIMAX

The broadband wireless access technology developed rapidly in 1990s. the wireless technology, represented by Local Multipoint Distribution System (LMDS) and Multichannel Multipoint Distribution Service (MMDS), found its position in the market oriented to the users of SOHO, medium/small companies, urban commercial centers and so on. However, beyond the expectation of all, this industry did not boom and expand further. The main reason is the lack of globally uniform standard for broadband wireless access.[11]

In 1999, IEEE organized the 802.16 workgroup to specialize in the technical specifications for broadband wireless access, aiming to establish a globally uniform standard for broadband wireless access. At present, IEEE 802.16 mainly comes down to

two standards: the 802.16-2004, i.e., the 802.16d standard for fixed broadband wireless access, and the 802.16e standard supporting mobile broadband wireless access. Issued on October 1st, 2004, the IEEE 802.16d standard specifies the air interface between the user terminal and base station system in fixed access, and mainly defines the physical layer and MAC layer of air interface. The main feature of 802.16e standard is the support to mobility. This standard specifies a system supporting both fixed and mobile broadband wireless access, which works in a permitted frequency range less than 6GHz suitable for mobility and supports the user terminal moving in the speed of vehicle. Meanwhile, the user capacity of fixed wireless access specified by 802.16d will not be affected.

The IEEE 802.16 workgroup established the specification and standard specifically to the physical layer and MAC layer of the wireless MAN. To compose an operational network, IEEE 802.16 technology requires the support of other components. As a result, the WiMax Forum emerges because of demand. The WiMax Forum was established in April, 2001. At the beginning, the organization intended to offer certification service for conformity and interoperability of broadband wireless access products based on the IEEE 14 802.16 standard and the ETSI HiperMAN standard. The product certificated by WiMax is identified by the mark of “WiMax Certified”. Along with the development of 802.16e technology and specification, the purpose of the organization expands gradually. It not only intends to establish a whole set of certification system based on the 802.16 standard and the ETSI HiperMAN standard, but also turns to the research on broadband wireless access system, analysis of demands, exploration of application modes, expansion of market, which greatly promote the development of broadband wireless access market. It is generally agreed that the IEEE 802.16 workgroup is the constitutor of the IEEE 802.16 WiMax air interface specification, while WiMax Forum is the propellant of technology and industry chain. Nowadays, WiMax is nearly another name of IEEE 802.16 WiMax technology, with the air interface specification covering IEEE 802.16d/e standards.[12]

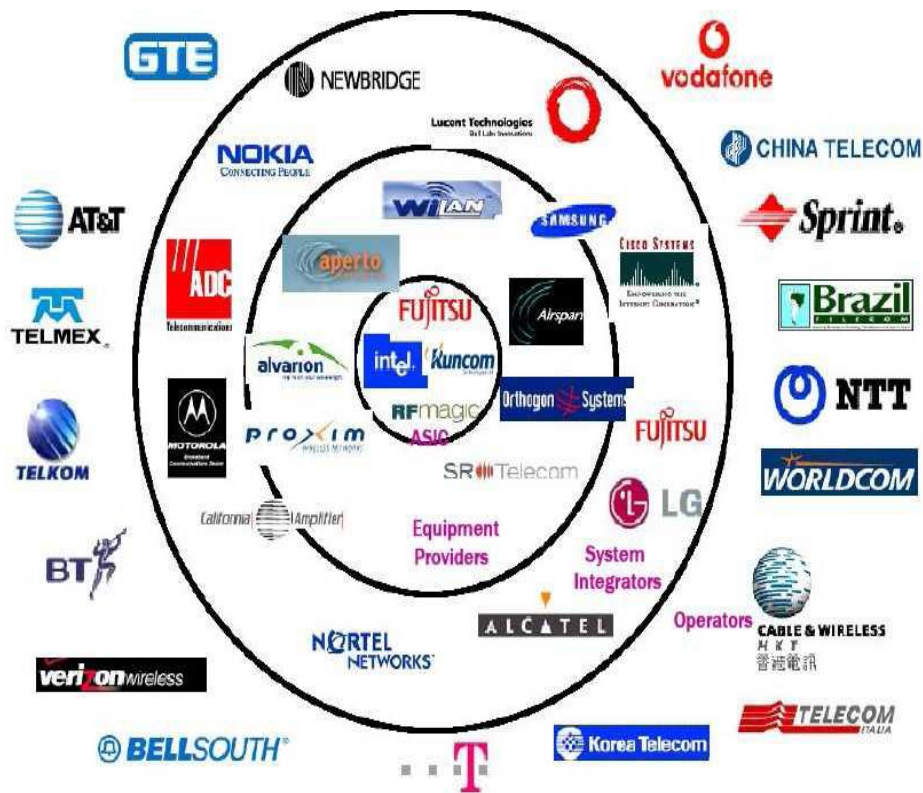


Figure 2.1 WiMax Forum's Members [13]

2.2.2 Introduction of IEEE 802.16 Standards

there was no uniform technical standard for LMDS and MMDS , and the air interfaces provided by different manufacturers were incompatible, which limited the development of the whole communication industry severely. To solve this problem, IEEE organized the IEEE 802.16 workgroup in 1999 to specialize in the development of broadband wireless access technology. IEEE 802.16 is a series of wireless broadband standards written by the Institute of Electrical and Electronics Engineers (IEEE).

The versions of IEEE 802.16 standard will be described below:

2.2.2.1 IEEE 802.16-2001

first WiMAX standard is IEEE 802.16 - 2000, it is fixed broadband wireless access. The standard specifies a set of Medium Access Control (MAC) layer and Physical (PHY) layer to provide fixed broadband wireless access in a point to point (PTP) or point to multipoint (PMP) topology. It provided for basic high data links at frequencies between 11 and 60 GHz. IEEE 802.16-2001 is only suitable for clear areas because the microwave signals in the frequency range of 10–66 GHz have poor penetrability, and the signals are easily affected by rain attenuation

2.2.2.2 IEEE 802.16a

The IEEE 802.16a standard is the extension of IEEE 802.16-2001. It operates in the frequency range of 2–11GHz . The signal coverage reaches up to 50km (generally within radius of 10km) , seldom affected by rain attenuation. Therefore, the requirements on antenna installation are reduced a lot. Additionally, the support of Mesh topology is added to the IEEE 802.16a, and MAC also offers guarantee to QoS, which supports voice and video messages. All of these features make the IEEE 802.16a standard more competitive in the market. IEEE 802.16a2003 standard enhances the MAC layer to support multiple physical layer specifications, and enabled the standard to be used at frequencies below the 11 GHz minimum of the original standard.

2.2.2.3 IEEE 802.16-2004 (IEEE 802.16d)

In 2004, the IEEE 802.16 workgroup integrated the IEEE 802.16-2001 and IEEE 802.16a standards revised them again, and issued the IEEE 802.16-2004 (generally called the IEEE 802.16d) standard. IEEE 802.16-2004 defines details specifically to the physical layer and the MAC layer of the 2-66G frequency range. IEEE 802.16d is based on 802.16a with some minor improvements. This extension supports both time division duplex (TDD) and frequency division duplex (FDD) transmission and also creates system profiles for conformance testing of 802.16a equipment's. The WiMAX Forum profiles supporting IEEE 802.16 2004 are in the 3.5 GHz and 5.8 GHz frequency bands.[12]

2.2.2.4 IEEE 802.16-2005 (IEEE 802.16e)

the IEEE 802.16 workgroup presented the IEEE 802.16e standard version based on the IEEE 802.16-2004 standard, aiming to propose a wireless broadband solution providing high-speed information transmission and supporting high speed moving , it is considered the sole wireless broadband technology of the next generation competitive to 3G. In addition to supporting mobile communication, IEEE 802.16e also defines some functions in close connection with the mobility features, including supporting Handoff, the sleep mode saving energy, call search, and improved safety. Table 2.2 is the comparison between 802.16 standards.[12]

2.2.2.5 IEEE 802.16m

The beyond evolutionary update of the IEEE 802.16 e standard is the IEEE 802.16m (802.16-2011), and it is anticipated it will provide data rates of 100 Mbps for mobile applications and 1 Gbps for fixed applications and IEEE 802.16m systems operate in RF frequencies less than 6 GHz. It will allow cellular, macro and micro cell coverage; with currently there are no restrictions on the Radio frequency (RF) bandwidth although it is expected to be 20 MHz or more . Table 2.2 is the comparison between 802.16 standards.

Table 2.2: Comparison IEEE 802.16 Standards [14]

IEEE 802.16e	IEEE 802.16 Rev'd	IEEE 802.16	Standards
Est.Mid-2005	May 2004	December 2001	Completed
2-6 GHz	2-11 GHz	10-66 GHz	Spectrum
Mobile internet	Wireless DSL&Backhaul	Backhaul	Application
Non-Line of Sight	Non-Line of Sight	Line of Sight only	Channel Conditions
Up to 15 Mbps at 5Mhz Channelization	Up to 75 Mbps at 20-Mhz Channelization	32-134 Mbps at 28-MHz Channelization	Bit Rate
Same as 802.16d, scalable OFDMA	OFDM256,OFDMA 2048QPSK,16QAM, 64QAM	QPSK,16QAM & 64QAM	Modulation
Same as 802.16d	Selectable Channel bandwidths between 1.5 & 20 MHz	20,25 & 28 MHz	Channel bandwidths

2.2.3 PHY and MAC Layer of WiMax

WiMAX PHY is responsible for the transmission of data over the air interface (physical medium). The PHY receives MAC layer data packets through its interface with the lowest MAC sub layer, and transmits them according to the MAC layer QoS scheduling. WiMAX MAC layer comprises of three sublayers, which interact through service access points (SAP) to provide the MAC layer services, as shown in Figure 2.2, The convergence sublayer (CS) interfaces the WiMAX network with other networks by mapping external network data (from ATM, Ethernet, IP, etc.) to the WiMAX system.[15]

2.2.3.1 Characteristics of Physical Layer

The WiMAX physical layer is based on orthogonal frequency division multiplexing. OFDM is the transmission scheme of choice to enable high-speed data, video, and multimedia communications and is used by a variety of commercial broadband systems, including DSL, Wi-Fi, Digital Video Broadcast-Handheld (DVB-H), and MediaFLO, besides WiMAX.

Physical layer set up the connection between the communicating devices and is responsible for transmitting the bit sequence. It also defines the type of modulation and demodulation as well as transmission power. WiMAX 802.16 PHY-layer considers two types of transmission techniques OFDM and OFDMA. Both of these techniques have frequency band below 11 GHz and use TDD and FDD as its duplexing technology. WiMAX physical layer is based on the orthogonal frequency division multiplexing (OFDM). OFDM is a good choice of high speed data transmission, multimedia communication and digital video services. It even can maintain very fast data rate in a non-line of sight condition and multipath environment. In the following subsection we provide a detailed description of the OFDM. The role of the PHY-layer is to encode the binary digits that represent MAC frames into signals and to transmit and receive these signals across the communication media. The WiMAX PHY layer is based on OFDM; which is used to enable high-speed data, video, and multimedia communications and is used by a variety of commercial broadband systems. [16]

2.2.3.2 Characteristics of MAC Layer

The IEEE 802.16 MAC was designed for point-to-multipoint broadband wireless access applications. The primary task of the WiMAX MAC layer is to provide an interface between the higher transport layers and the physical layer. The MAC layer is divided into three sub-layers: Service Specific Convergence Sub-layer (CS), Common Part Sub-layer (CPS), and Privacy Sub-layer (PS).[17]

- i. The main function of CS is to convert and map the external network data received by SAP to the MAC SDU, and then transmit to the SAP of MAC layer.
 - Classifying external network service data units (SDUs).
 - Associating SDUs to the proper MAC service flow identifier (SFID) and connection identifier (CID)
 - Payload header suppression (PHS)
- ii. Common Part Sub-layer CPS is the hardcore of MAC, with the main functions of system access, band width allocation, connection establishment and connection maintenance
 - Provides the core MAC functionality of system access
 - Fragments or combines SDUs to appropriate MAC PDUs
- iii. The main function of PS is to provide authentication, key exchange and encryption/decryption processing.

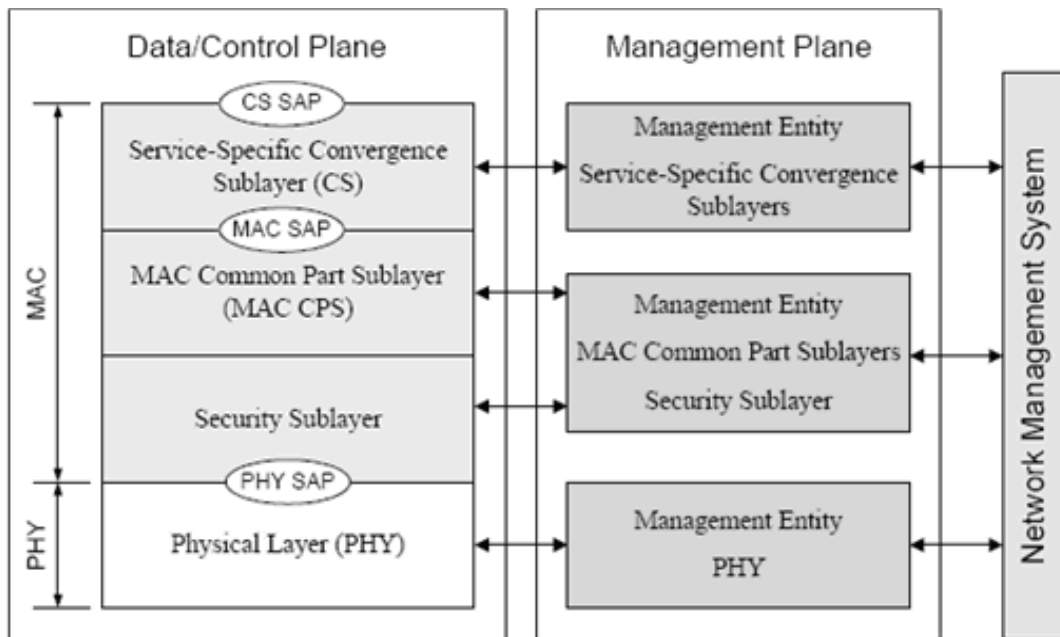


Figure 2.2: MAC Layer and PHY Layer of WiMax

2.2.4 WiMAX Architecture

The WiMAX network architecture is designed to provide an IP friendly framework with scalable data capacity, open access to innovative applications and services, enhanced QoS and mobility. The IEEE 802.16 standards define the structure of the PHY (Physical) and Link Layer operations that occur between mobile stations and base stations (BSs). There are three main components of WiMax network architecture. The first is the mobile stations which are used as a source of network connection for end user. The second is an access service network which is formed of more than two or three base stations. It also contains ASN gateways which build the radio access at the end. The third component is connectivity service network which is responsible for providing IP functions. The base stations provide mobile management functions, triggering and tunnel establishment, radio resource management, dynamic host control protocol proxy, quality of service enforcement and multicast group management.[18]

2.2.4.1 Network Reference Model

The WiMAX Network Reference Model (NRM) logically representing a WiMAX network architecture is shown in the figure below.

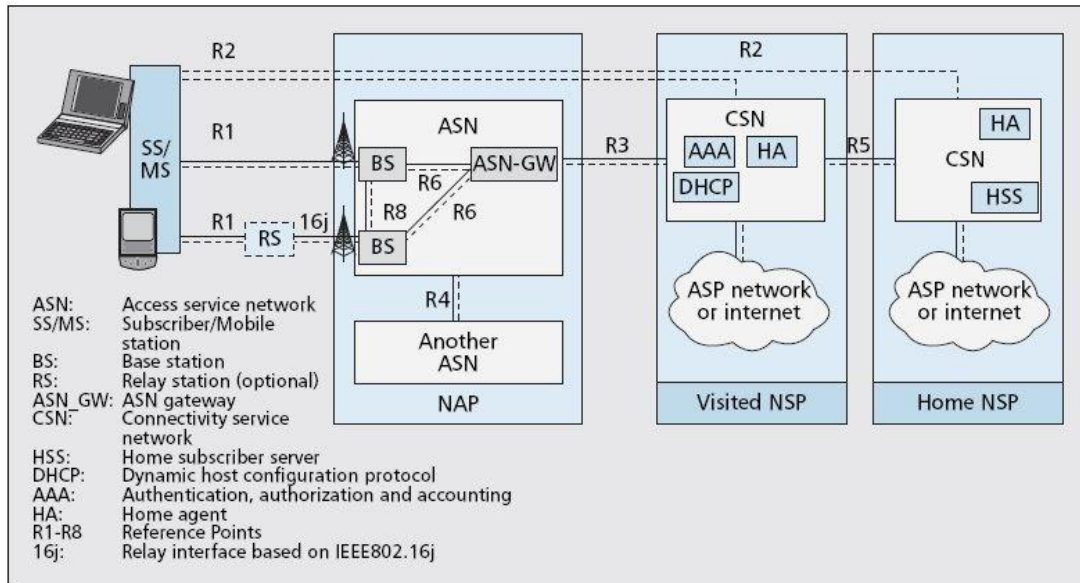


Figure 2.3: WiMAX Network Reference Model [19]

The Network Reference Model identifies the key functional entities and reference points for network interoperability. The network access provider (NAP) is a business entity that provides WiMAX radio access infrastructure while the Network Service Provider (NSP) is the business entity that provides IP connectivity and WiMAX services to the subscribers according to negotiated SLAs (Service Level Agreements) with one or more NAPs. The WiMAX NRM has several logical entities such as MSs (Mobile Stations) /SS (Subscriber Stations), ASN (Access Service Network), CSN (Connectivity Service Network) which represent logical grouping of functions they perform. The interaction between various elements of these entities takes place through the reference points R1-R8.[20]

- 1. Mobile Station (MS):** The Mobile Station (MS, often referred as Subscriber (SS)) is mobile equipment that provides wireless connectivity of one or more hosts to the WiMAX network. In this context the term MS is used more generically to refer to both mobile and fixed device terminals.

2. Access Service Network (ASN): represents a complete set of network functions required to provide radio access to the MS. These include:

- Layer 2 connectivity with the MS
- Transfer of AAA (Authentication, Authorization and Accounting) messages to the H-NSP (Home NSP)
- Preferred NSP discovery and selection
- Relay functionality for establishing layer 3 (L3) connectivity with MS
- Radio Resource Management (RRM)
- ASN and CSN anchored mobility
- Paging and location management
- ASN-CSN tunneling

The ASN may be implemented as an integrated ASN where all functions are collated in a single logical entity, or it may have a decomposed configuration in which the ASN functions are selectively mapped into two separate nodes, a BS (Base Station) and an ASN-GW (ASN Gateway).[20]

- **Base Station (BS)**

a logical network entity that primarily consists of the radio related functions of an ASN interfacing with an MS over-the-air link according to MAC and PHY specifications in IEEE 802.16 specifications subject to applicable interpretations and parameters defined in the WiMAX Forum system profile. Each BS is associated with one sector with one frequency assignment and may incorporate a Down Link/Up Link scheduler.

- **ASN Gateway (ASN-GW)**

ASN-GW is a logical entity that represents an aggregation of centralized functions related to Qos, security, and mobility management for all the data connections served by its association with the BSs through the R6 interface. It also performs IP layer interactions with the CSN through R3 interface and with other ASNs through R4 interface for mobility.

3. Connectivity Service Network (CSN)

a set of network functions that provide IP connectivity services to WiMAX subscriber(s). The CSN may further comprises network elements such as routers, AAA proxy/ servers, home agent, and user databases as well as interworking gateways or enhanced broadcast services and location-based services. A CSN may be deployed as part of a green field WiMAX NSP or part of an incumbent WiMAX NSP. The following are some of the key functions of the CSN:–IP address management–AAA proxy or server–QoS policy and admission control based on user subscription profiles–ASN-CSN tunneling support –Subscriber billing and interoperate settlement–Inter-CSN tunneling for roaming–CSN-anchored inter-ASN mobility–Connectivity to Internet and managed WiMAX services such as IP multimedia services (IMS), location-based services, peer-to-peer services, and broadcast and multicast services – Over-the-air activation and provisioning of WiMAX devices [20]

4. Reference Points

The WiMAX network reference model defines several reference points (RPs) between several entities in the WiMAX network (see Figures 13.1 and 13.3). Those RPs introduce interoperability points between equipments from different vendors. In the scope of the Release 1 WiMAX network architecture, there are six mandatory RPs (R1 to R6) and two informative RPs (R7 and R8).

Reference point R1 refers to the radio interface between the MS and the ASN, and R2 is a logical interface between the MS and the CSN. R3 is the logical interface between the ASN and the CSN. R4 interconnects two ASNs (ASN profile B) or two ASN-GWs (ASN profiles A or C), It conveys both control and data plane messages . R5 is the interface interconnecting two CSNs. and Reference point R6 is normative in WiMAX architecture Release 1 in the context of specific ASN profiles. In the ASN, there are currently two additional interfaces (R7 and R8) defined for further interoperability points in the scope of the next WiMAX architecture releases.[21]

2.2.5 Technologies of WIMAX

2.2.5.1 MIMO (Multiple Input Multiple Output) Technology

Multiple-Input Multiple-Output (MIMO) technology is a wireless technology that uses multiple transmitters and receivers to transfer more data at the same time. MIMO technology takes advantage of a radio-wave phenomenon called multipath where transmitted information bounces off walls, ceilings, and other objects, reaching the receiving antenna multiple times via different angles and at slightly different times. MIMO has become an essential element of wireless communication standards including IEEE 802.11n (Wi-Fi), IEEE 802.11ac (Wi-Fi), HSPA+ (3G), WiMAX (4G), and Long Term Evolution (LTE 4G).

MIMO is one of several forms of smart antenna technology, the others being MISO (multiple input, single output) and SIMO (single input, multiple output). The of MIMO system with N transmitting antennae and M receiving antennae is shown in Figure 2.5.

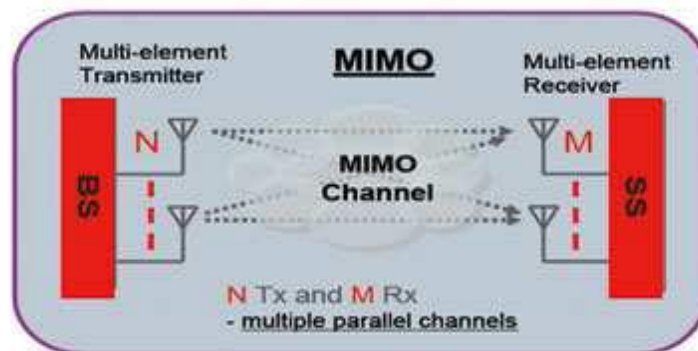


Figure 2.4: MIMO system [22]

Following are the benefits or advantages of MIMO:

- There is lower susceptibility of tapping by unauthorized persons due to multiple antennas and algorithms.
- The wide coverage supported by MIMO system helps in supporting large number of subscribers per cell.
- MIMO offers potential for: larger data rate, larger spectral efficiency, larger number of users, improved range/coverage, better quality of service (QoS), lower bit-error rate (BER).

2.2.5.2 OFDM (Orthogonal Frequency Division Multiplexing)Technology

OFDM is a high-speed transmission technology in wireless environment .The main idea of OFDM is to divide the bonded channel into many orthogonal sub-channels in the frequency range, and use a sub-carrier on each sub-channel for modulation, in which the sub-carriers are transmitted in parallel. Accordingly, the interference between signal waves will be eliminated. Because the carriers on these sub-channels in OFDM are orthogonal to each other, they have overlapped frequency spectrum, which reduces the interference between sub-carriers and improves the utilization rate of frequency spectrum.[12]

OFDM has developed into a popular scheme for wideband digital communication, used in applications such as digital television and audio broadcasting, DSL Internet access, wireless networks, power line networks, and 4G mobile communications. FDM stands for Frequency Division Multiplexing and OFDM stands for Orthogonal Frequency Division Multiplexing. As shown in the figure 2.6 FDM systems carriers are far apart with respect to each other and in OFDM systems carriers are densely packed and are orthogonal to the other carriers. Orthogonal means peak of one carrier occurs at null of the other. Hence OFDM system is bandwidth efficient compare to FDM system. in FDM system carriers are not orthogonal.

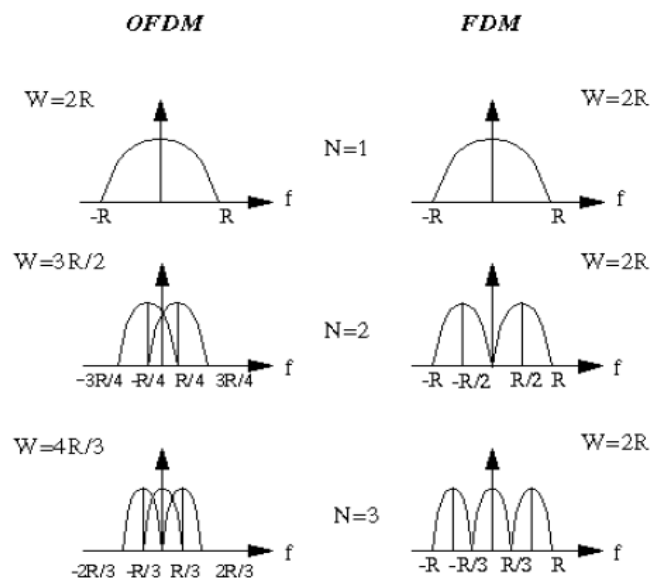


Figure 2.5: OFDM vs. FDM

Advantages and Disadvantages of OFDM

OFDM tends to replace CDMA to be the new generation core technology of wireless communication, mainly because of its advantages:

- Makes efficient use of the spectrum by allowing overlap.
- OFDM makes use of cyclic prefix to eliminate ISI (Inter Symbol Interference) found in the multipath channel environment. Hence it is robust to multipath fading.
- It is possible to allocate bandwidth as per resource requirements. Hence OFDM is bandwidth scalable technique.
- Provides good protection against channel interference and impulsive parasitic noise.
- Flexibly allocation of frequency spectrum resource.

disadvantages :

- The OFDM signal has a noise like amplitude with a very large dynamic range, therefore it requires RF power amplifiers with a high peak to average power ratio.
- It is more sensitive to carrier frequency offset and drift than single carrier systems are due to leakage of the DFT.
- As OFDM spectrum travels through multiple paths which require guard band to avoid ISI errors due to timing offsets.

2.2.5.3 OFDMA

Orthogonal Frequency-Division Multiple Access (OFDMA) is a multi-user version of the OFDM scheme, a digital multi-carrier modulation scheme which uses the principle that information can be transmitted on a radio channel through variations of a carrier signal's frequency, phase or magnitude. The OFDMA system allocates the bandwidth resources available to users in demand, which realizes the optimized utilization of system resources easily. As different users occupy non-overlapped subcarrier sets, there is no interference between users in case of ideal synchronization, i.e., no Multiple Access Interference (MAI). The figure 2.7 shows OFDM and OFDMA subcarrier allocation. The sub-channel OFDMA divides the bandwidth of the entire OFDM system into several sub-channels, and each sub-channel has several sub-carriers allocated to a user and each user may occupy more than one sub-channel.

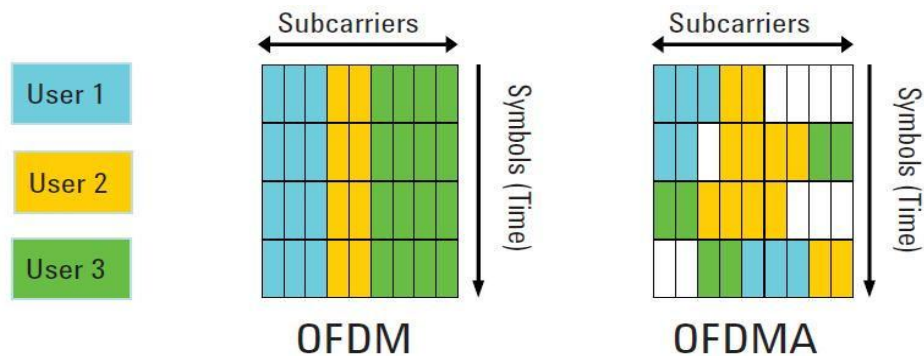


Figure 2.6: OFDM and OFDMA subcarrier allocation [23]

Following are the advantages of OFDMA:

It provides multi-user diversity, A single frequency network can be used to provide excellent coverage and good frequency re-use, When used in a cellular system, it is possible to minimize interference from neighboring cells by using different carrier permutations between the two cells, Offers frequency diversity by spreading the carriers all over the used spectrum.

2.3 WIMAX HANDOVER OVERVIEW

Mobility is one of the most important features in wireless cellular communication. When a mobile node changes its location from one location to another, it must support the minimum requirements are reliable, efficient, and continuous data transmission. Handoff is a process with the intention of changing the network access point of a mobile node without any data loss or disturbance of the current connection while a call is in progress. Handover is a mechanism to maintain uninterrupted user communication session during a user's movement from one location to another . When the Mobile Station (MS) moves to another cell and performs handover, the service packets for the MS will be delayed and the service might be disrupted for some time. This delay is known as Latency time.[24]

Handover (HO) mechanism is one of the critical operations in Mobile WIMAX. The handover procedure is comprised of several stages, MS scanning, cell selection, handover initiation and decision and network re-entry including ranging, authorization, authentication and re-registration. takes place Handover when a mobile station (MS) moves from a serving base station (BS) to another BS.[25]

Handover is considered as a highly important issue in WiMAX and LTE and it provides continuous connection. Handover occurs when a mobile station migrates from one BS to another BS. In terms of the type of handover, it is characterized as hard handover and soft handover [26]. Hard handover is known as break-before- make, which means that the MS disconnects the existing link before establishing communication with new BS. Soft handover comprises macro diversity handover (MDHO) and fast base station switching (FBSS) that improve the QoS performance while adding more BSs .

2.3.1 Types of handover

There are two types of handovers used in cellular network systems: hard handover and soft handover. The hard handover is used when the communication channel is released first and the new channel is acquired later from the neighboring cell. For real-time users it means a short disconnection of communication. Thus, there is a service interruption when the handover occurs reducing the quality of service. Hard handover is used by the systems which use time division multiple access (TDMA) and frequency division multiple access (FDMA) such as GSM and General Packet Radio Service (GPRS) [27]. in the IEEE 802.16e are defined three types of handover : Hard handover,

Macro Diversity Handover (MDHO) and Fast Base Station Switching (FBSS). Hard handover is mandatory in WiMAX systems.

2.3.1.1 Hard handover

During hard handover the MS communicates with only just one BS in each time. Connection with the old BS is broken before the new connection is established. Handover is executed after the signal strength from neighbor's cell is exceeding the signal strength from the current cell. This situation is shown in below figure 2.9 – Red thick line at the boarder of the cells represents the place where the hard handover is realized. in a hard handover the mobile station will interrupt the communication with the serving BS and make a transition to the target BS. The MS have to register with the target BS and can then begin to listen and send to the new BS instead. The hard handover is used in cases where the soft handover is not suitable or cannot be employed, e.g. in handovers between different networks.

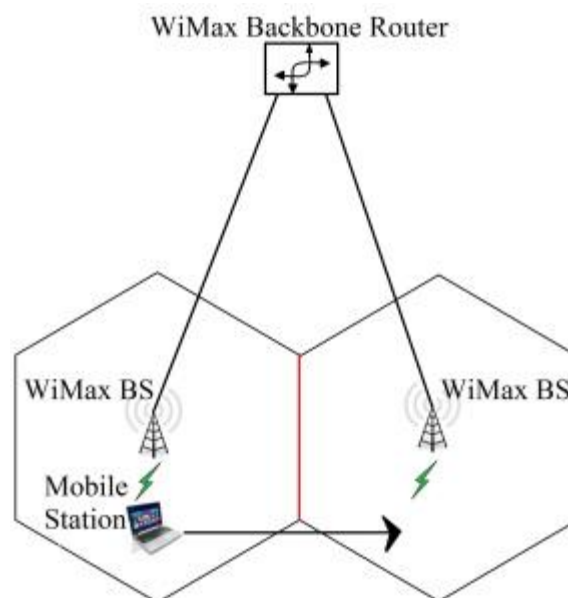


Figure 2.7: Hard Handover (HHO)

2.3.1.2 Soft Handover

The soft handover (SHO), in contrast to hard handover, establishes multiple connections with neighboring cells. SHO is characterized by “make before break” connection in which the MS can be connected to different BS at a given point of time and do not have to disconnect its current connection before establishing a new connection. SHO is divided into two types: Macro-Diversity Handover (MDHO) and Fast Base Station Switching Handover (FBSS).

- **Macro Diversity Handover:** - MDHO is designed to allow full seamless mobility at much higher speeds (up to 120 kmph). It has the advantage of performing handovers within sectors having the same carrier frequency due to their employing the universal reuse concept. In the MDHO approach, the downlink data are transmitted to MS by two or more BSs such that diversity combining can be performed at the MS. For uplink MDHO, the MS transmission is received by multiple BSs such that selection diversity of the received information could be performed.[28]

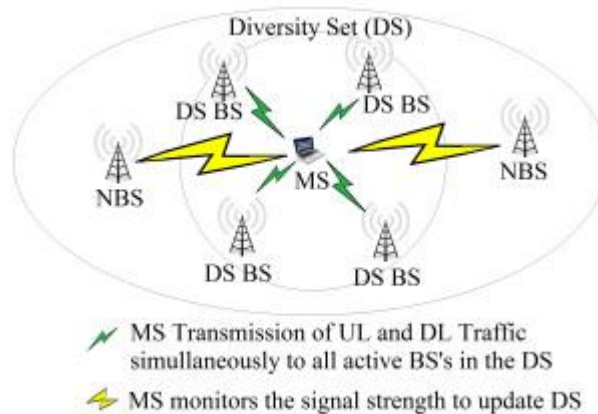


Figure 2.8: Macro-Diversity Handover (MDHO)

- **Fast Base Station Switching:** - . In this method a diversity set is maintained for each mobile station. The serving base station and mobile station monitors the neighboring base stations that can be added in diversity set. Diversity set is maintained by both mobile station and serving base station. Diversity set is collection of base stations that can be chosen as target base station for a handover, In fig. 2.11, the mobile station selects one base

station from the mobile station selects one base station from diversity set as anchor base station sends its current location to it which is sent to base station controller for decision of a handover. Whenever there is a need of handover base station controller sends handover initiation message to mobile station. Handover decision can be taken by mobile station, base station or base station controller depending upon the implementation [29].

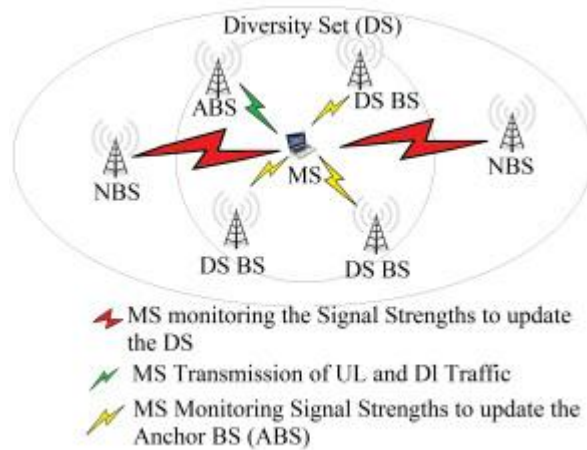


Figure 2.9: Fast Base Station Switching (FBSS)

The important concepts in the MDHO and FBSS are:

- **Diversity Set Updating Diversity set:** Diversity set is the group of BS's selected by an MS such that all these BS's operate on same frequency channel and are synchronized on time and frame level. Diversity set is formed by sharing of MAC context between BS and MSS. Also MSS select only those BS's as the member of diversity set whose CINR (Carrier to interference noise ratio).
- **Active BS:** All the BS's of the diversity set having all the information about the MSS including the MAC context.
- **Neighbor BS:** This is the BS which is not the part of diversity set i.e., no traffic is exchanged with this BS but still MS can perform the signal strength measurement with this BS.

2.3.2 Why handover

- When the MS is moving away from the area covered by one BS and entering into the area covered by another BS, then handover is needed to transfer the connection of MS from current BS to other BS before the MS moves out of the range of first BS, to avoid call termination .
- When the traffic handling capacity of any BS is exhausted then in order to accommodate more or new calls, loaded BS by means of handover transfers the ongoing or newly originated call to the neighboring BS with overlapping coverage area .
- in order to avoid interference on any channel by different MS from different cells (BS) using the same channel (but in different cells) then call is transferred to another channel in same cell or another cell.
- In order to avoid or reduce interference due to “near -far” effect in CDMA networks, soft handover is useful in such scenarios.

CHAPTER 3

VOICE OVER INTERNET

PROTOCOL (VOIP)

3.1 INTRODUCTION

Today's modern society is highly dependent on several communication services. On a daily basis most of us retrieve information online, correspond in different forms of text messages and communicate through speech over the cellular network. The globalization has led to Voice over IP (VoIP) becoming increasingly more popular with Skype as one of the most widely used applications.

The use of VoIP in business applications with high call volume can reduce communication costs by a great magnitude. This is why VoIP is widely used in call centers, help desks, corporate Private Branch Exchanges (PBX) and over-the-phone sales. But the traditional means of using VoIP involve initiating the call using a personal computer or a landline phone, where the user needs to be statically located during the call. However, mobility is the key single-most convenience that people are used to now-a-days in voice conversation. This is why, despite the high service cost associated with cellular networks (GSM or CDMA), mobile phones are more popular than landline or VoIP services due to the convenience of mobility. To attain an even greater consumer acceptance than the current market, VoIP service needs to be offered in mobile devices while maintaining the low cost of service.

In This chapter we will talk about VoIP System Architecture, its mechanism, Audio codes more widely used are disused, then its main protocols, some key issues for VoIP are reviewed.

3.2 VOIP OVERVIEW

VoIP is the most widely used real-time application in the Internet and is becoming increasingly more popular every day. Both corporate and home users have embraced this technology leading to its quick growth. VoIP can offer voice communication at a very low price which has become a lucrative option for private users due to the recent global recession Furthermore, the globalization of businesses and industrial markets in the last decade has heightened competition among companies and led to rigorous cost cutting in firms of all sizes. But the communication requirement is always increasing. Due to the global nature of many businesses, international calls have become an expensive everyday necessity. This is why companies are increasingly adopting VoIP to reduce the cost of local and international voice communication. global demand for VoIP

services market was valued at over USD 83 billion in 2015, is expected to reach above USD 140 billion in 2021 and is anticipated to grow at a CAGR of above 9.1% between 2016 and 2021. [30] As shown below.

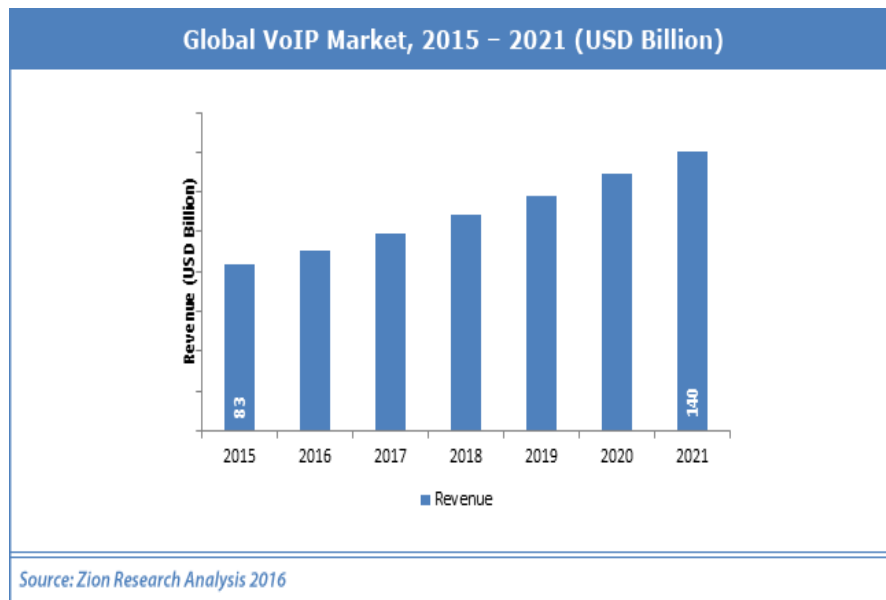


Figure 3.1: Global VoIP Services Market [31]

3.2.1 What is VOIP?

VoIP (voice over IP) is the transmission of voice and multimedia content over Internet Protocol (IP) networks. The terms Internet telephony, broadband telephony, refer to the provisioning of communications services (voice, fax, SMS, voice-messaging) over the public Internet, rather than via the public switched telephone network (PSTN). VoIP refers to a way to carry phone calls over an IP data network, whether on the Internet or your own internal network. A primary attraction of VoIP is its ability to help reduce expenses because telephone calls travel over the data network rather than the phone company's network. Public Internet phone calling uses the Internet for connecting phone calls, especially for consumers.

3.2.2 Importance of VoIP

Voice service cost is primarily associated with fixed deployment and regular maintenance. VoIP does not require any dedicated infrastructure and a major advantage

of using VoIP is the cost savings by hardware reduction and the elimination of the service charges and toll costs associated with regular fixed-line telephony like PBX and PSTN. VoIP can be used over any existing IP network and can share the same network with other applications. Deployment usually only requires installation of VoIP phones, if they are used, although personal computers can be used instead to further reduce the installation cost. Sometimes central directory and call monitoring services are employed in order to improve voice performance and the users' convenience. Such services are usually very cheap and can be co-located with other services and hosted on any existing server. Corporate networks are usually maintained for data services anyway and no separate maintenance is required for voice calls. In this way, VoIP can minimize the cost of deployment, operations and maintenance for intra-company and inter-company (external) communications.

VoIP is a key factor in IP convergence where a single high speed Internet connection can be used for all kinds of business communication services including voice calls, teleconferencing, email, pager, etc. Convergence is another reason behind corporate interest in VoIP. A converged network eliminates the need for duplicate hardware and special vendors to manage separate voice networks and Private Branch Exchanges (PBX). Convergence allows voice, video and data to be used in a combined fashion using a single device and a single network infrastructure. Thus, a high user interactivity can be achieved at a much lower cost through convergence. A recent survey showed that 36% of small business organizations use VoIP over Wi-Fi for their internal communications [32].

Household Internet users have embraced VoIP with equal enthusiasm. With a growth rate of 88%, residential broadband users are expected to reach 121 million in Western Europe only. About 73% of these users will use VoIP giving an estimate of 87.8 million home users who will start using VoIP by 2012. A growth rate of 240% for household VoIP users with 50% household penetration is also reported in. A VoIP penetration of 20% in U.S. household users is expected in. The number of Internet users is a good indication of IP telephony use since most broadband users also use the VoIP services that comes without incurring any additional cost. Latest statistics in [33] shows that 28.7% of world population (6.845 billion) is using the Internet. Although, most users are from Asia (42%) and Europe (24.2%), the user growth rate is highest in North America (77.4%), Oceania (61.3%) and Europe (58.4%).

3.2.3 Applications of VoIP

The ability of cheap verbal communication and the widespread penetration of the broadband Internet are the major attractions of VoIP. A number of potential VoIP applications are listed in this section.

- **Call Center and Help:** desk Many companies offer a help desk to assist their customers. For instance, computer and laptop services, mobile phone operators and Internet Service Providers (ISP) all offer some “Helpline” telephone numbers where customers can call to seek assistance. The volume of the calls to these numbers is very high and the use of VoIP for these services can reduce production/service cost.
- **Sale over phone** Today producers of consumer goods are adopting over the phone sales where operators call potential customers and offer them products such as life insurance policies, utility connections, holiday accommodations or fitness equipment's. This sales method reduces the costs involved in renting an office, paying salary to a high number of employees and the maintenance of office equipment's. Moreover, semi-interested customers who might not visit a shop or are not looking seriously for a product can be reached. The problem with this method is that it requires a large number of phone calls where VoIP can be used to reduce the operational costs and save a fortune
- **Corporate PBX** Most organizations provide phones to their employees to increase interactivity and lessen process delay so that a high productivity can be achieved. But the use of a PSTN phone for every employee is expensive and traditionally a Private Branch Exchange (PBX) is used to minimize the PSTN line requirement. But PBX needs a separate, specifically built infrastructure which incurs an additional cost. VoIP based PBX can utilize a corporate LAN and keep the communication cost to a minimum.
- **Home Phone Applications** The traditional PSTN connections used in home applications are now largely being overtaken by VoIP connections. The overwhelming interest of home users in VoIP based packages is primarily due to the low service cost. Moreover, VoIP based connections also support the simultaneous use of multiple devices, e.g., PDA, laptop, mobile phone and wireless handset, etc., with a single connection. With an appropriate

configuration, all such connected devices can start ringing on an incoming call which offers the convenience of choosing the appropriate device.

- **On-call Paging** On-call staffs and patients at hospitals and nursing homes frequently need to contact each other. Considering the volume of such communications, a low-cost solution would be highly beneficial. VoIP can be used in such scenarios to provide a reliable voice communication system at a low cost of service.

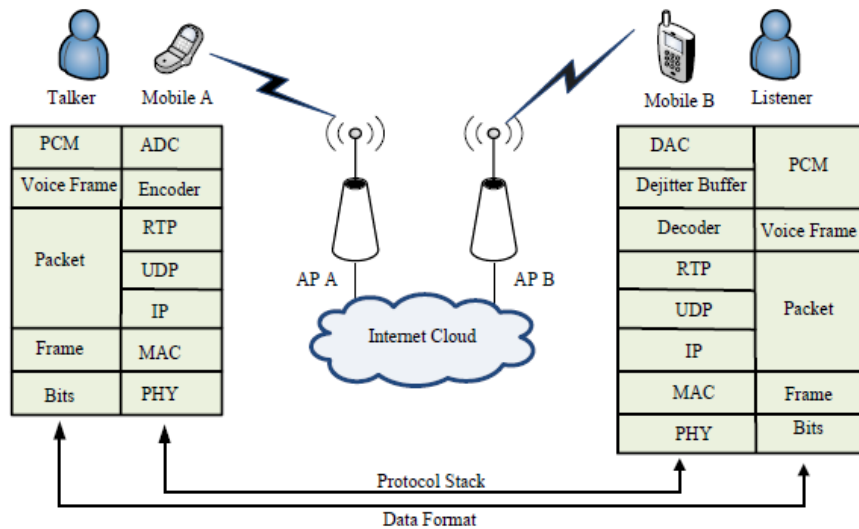


Figure 3.2: typical VoIP systems with different protocol layers and data units.

3.3 VOIP COMPONENTS

VoIP is a mature technology that has been widely deployed across public and private sectors since it uses existing IP data network infrastructure, eliminating expensive traditional dedicated voice circuits. To describe how the VoIP and IP Telephony work we first need to see what components does VoIP network have.

- **IP phone** – In order to send and receive voice calls, IP phones make use of a type of network connection known as Ethernet network connection. An example of IP telephony device is Cisco IP and Cisco 7975G.
- **Gateway** – A gateway is able to promote calls which take place between various networks. With the help of a gateway you can place a call between your IP

phone and your office. You could also place a call to the PSTN to call your home.

- **Call agent** – Most of the characteristics which were formerly a part of PBX have now been replaced by Call Agents. For instance, to approximately conclude how calls are routed, a call agent can be configured with the help of rules. An example of such a call agent is Cisco UCM (Unified Communication Manager)
- **Application server** – Application Servers provide on the top services such as voice mail in a VoIP environment.
- **Videoconference station** - A videoconference station provides access for end-user participation in videoconferencing. The videoconference station contains a video capture device for video input and a microphone for audio input. A user can view video streams and hear audio that originates at a remote user station.
- **Other components** _ such as software voice applications, interactive voice response (IVR) systems, and soft phones, provide additional services to meet the needs of an enterprise site.[34]

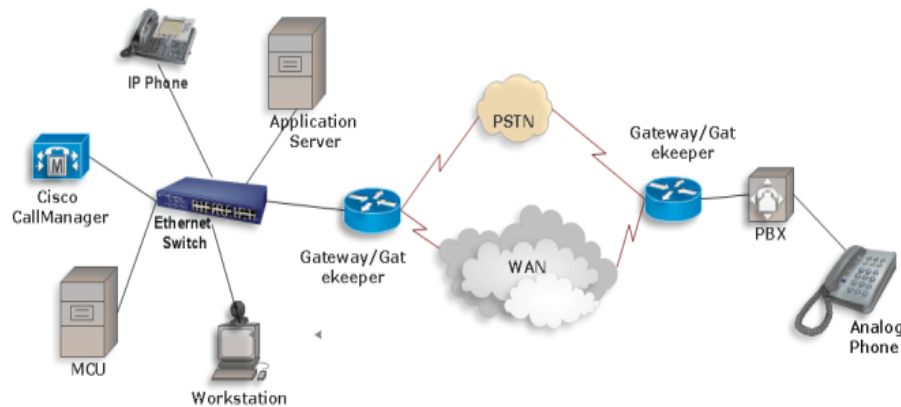


Figure 3.3: VoIP components

3.4 TYPES OF VOIP

The most interesting VoIP feature is that you will be able to place calls in different ways and it will depend on your choice. There are several different types of VoIP service depending on the infrastructure used for the communication.

i. Phone to Phone (Analog Telephone Adapter or ATA)

You will need an analog telephone adapter or ATA that helps you to connect your standard phone to the computer network and use VoIP. This is a standard that converts analog signals into digital signals.[35]

ii. Phone to computer (IP phone)

These are the phones that have been specifically designed to work with VoIP service. These phones can be directly connected to the router and it has all the necessary hardware as well as software to manage IP calls.

iii. Computer-to-computer

This is the most popular and easiest of all VoIP calling as you will be able to make calls between two computers that are on the same network. Also, you won't need to pay for the long distance calls because they use the Internet, There are no additional charges on the calls that you make from a computer because the calls are absolutely free. This type of VoIP services via specialized software applications (softphone software) such as Skype, Messenger, and Viber etc. These services require users to download their software and get them installed on PC, Caller and receiver need to use same VoIP software application.[35]

iv. Computer to Phone

You can place calls from your computer to any phone around the world and charges are very low.

v. Mobile-to-other devices

. You can make and receive VoIP calls the same way as if you are sitting at your desk With an app that you can download for Android or IOS, you cellular phone becomes an end device.[35]

3.5 VOIP PROTOCOL ARCHITECTURE

VoIP consists of a collection of technologies and protocols that together enables communication services over the Internet rather than over the publicly switched telephone network as for conventional voice calls. The relevant protocols are depicted in Figure 3.4 and described in the sections to follow. The key feature to regard is that the IP protocol adds a header of 20 bytes and that the link layer adds additional overhead, 18 bytes for a MAC header if Ethernet is used [36].

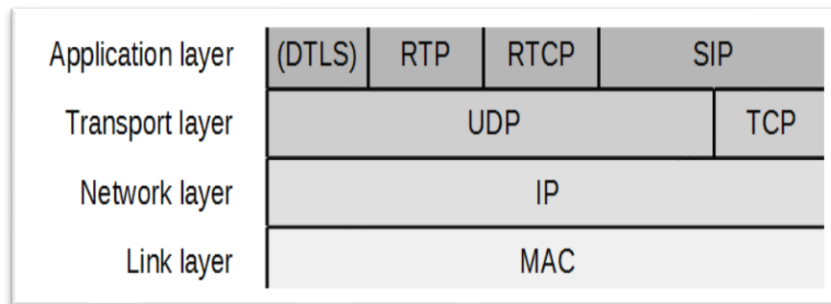


Figure 3.4: Protocols relevant in the VoIP architecture and mapping to their respective layers in the Internet protocol suite.

3.5.1 Transport Control Protocol

The Transport Control Protocol (TCP) [37] was originally specified in 1981 and is one of the core protocols upon which Internet communication relies. It is connection-oriented and designed to provide reliability and flow-control resulting in that the data is accurately delivered and in-order, assuming the two entities have a functioning network connection between them. The reliability comes at the price of additional overhead and increased latency since a connection must be established and maintained.

3.5.2 User Datagram Protocol

The User Datagram Protocol (UDP) and TCP are the two main protocols for the transport layer in the Internet protocol suite. Being connectionless, UDP differs fundamentally from TCP in what it provides. Data may be delivered outof- order, duplicated or even lost but since no connection is required to be established or maintained the latency is greatly reduced compared to TCP. Hence UDP is suitable for real-time data where low latency is of greater importance than complete delivery. Voice communication is such an example since minimal delay is more desirable than completely accurate recreation of the audio signal, as long as it is comprehensible.[38]

3.5.3 Datagram Transport Layer Security

The Datagram Transport Layer Security (DTLS) protocol is specified by the Internet Engineering Task Force (IETF) in . It is designed to provide integrity, authentication and confidentiality for the UDP protocol. The use of the DTLS protocol is optional for VoIP traffic. The purpose of it is to create a Virtual Private Network (VPN) between the two clients. Using a VPN connection prevents the traffic from being classified as VoIP traffic en route and thereby potentially sniffed or blocked by various parties. On the other hand, the prioritization of VoIP traffic that some network equipment provides cannot be utilized if VPN is enabled. Note that a VPN can be created using other protocols as well, such as IPsec [39] or TLS . The main point is that it is possible, and sometimes necessary, to hide the VoIP traffic and doing so adds an overhead.

3.5.4 Real-time Transport Protocol

When transmitting real-time data over a network, the Real-time Transport Protocol (RTP) [40] is typically used. The protocol provides end-to-end network functions for streaming media and can run both on TCP and UDP but is in practice only used with UDP since it is more suitable for real-time data. RTP itself does not provide any mechanisms for QoS, that is instead handled by its sister protocol RTCP,

3.5.5 RTP Control Protocol

Real-Time Transport Control Protocol (RTCP) is a protocol that works with Real-Time Protocol (RTP) to monitor data delivery on large multicast networks. RTCP carries statistical and control data, while RTP delivers the data. RTCP statistics typically include the number of bytes sent, packets sent, lost packets and round trip delay between endpoints .[41]

3.5.6 Session Initiation Protocol

This is the IETF's standard for establishing VOIP connections. It is an application layer control protocol for creating, modifying and terminating sessions with one or more

participants. The architecture of SIP is similar to that of HTTP (client-server protocol). Requests are generated by the client and sent to the server. The server processes the requests and then sends a response to the client. A request and the responses for that request make a transaction. SIP has INVITE and ACK messages which define the process of opening a reliable channel over which call control messages may be passed. SIP makes minimal assumptions about the underlying transport protocol. This protocol itself provides reliability and does not depend on TCP for reliability. SIP depends on the Session Description Protocol (SDP) for carrying out the negotiation for codec identification. SIP supports session descriptions that allow participants to agree on a set of compatible media types. It also supports user mobility by proxying and redirecting requests to the user's current location. The services that SIP provide include. [42]

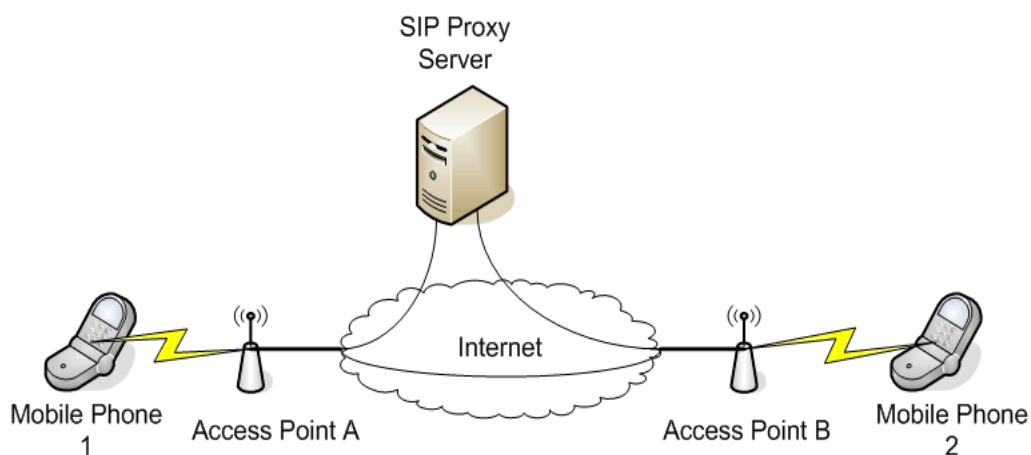


Figure 3.5: Example of SIP being used for Voice over wireless network

3.6 VOIP SYSTEM STRUCTURE

At a high level a VoIP system consists of three main parts - a sender, an IP network and a receiver, all depicted in Figure 3.6. A voice-codec in the sender digitalizes and compresses the received voice stream into speech frames,. To not congest the network, several speech frames are then packetized to form the payload of a packet (e.g. RTP packet) and headers required by the network are added. The network may then introduce different impairments such as packet loss, delay and jitter before the packet is delivered to the receiver. The packets are there stripped of their headers and the speech frames are extracted by the depacketizer. A buffer is used to counteract the jitter introduced by the network at the cost of additional delay. Finally the speech frames are decoded and outputted, potentially with the use of packet loss concealment to compensate for lost packets. [43]

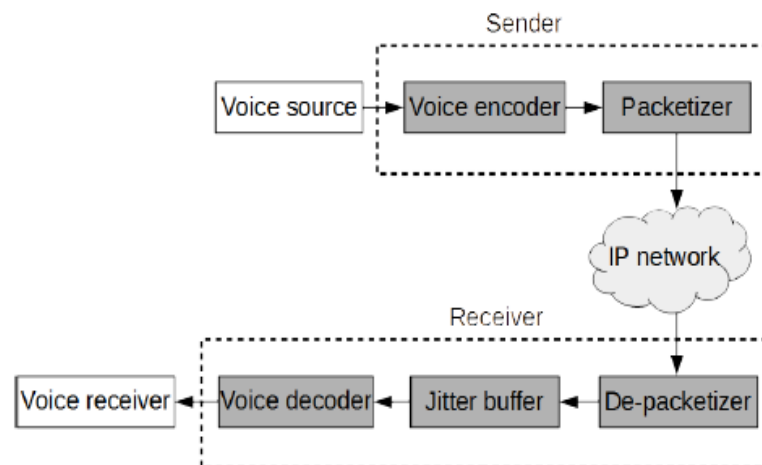


Figure 3.6: Conceptual diagram of a VoIP system.

3.7 VOICE COMPRESSION CODECS

The word codec comes from mashing together the functions of compressing (co) and decompressing (dec) analog sound into digital bits for use by computers and networks. A number of codecs are ITU (International Telecommunications Union) standards, formalized for international use and incorporation into devices. If a codec name starts with a G and a period, such as G.711 or G.722, it's an ITU standard.. There are different codec types based on the selected sampling rate, data rate, and implemented compression. Codecs have various functions and have certain

requirements, for example a codec which requires high bandwidth will perform best with high bandwidth and will also have better voice quality, it means Internet connection must be fast .Every codec has a capability to produce a specific quality of speech. Choose the codec depends on network conditions and its requirements. We will discuss in this section the most popular and commonly used codecs on a large scale in in voice communications Networks. [44]

3.7.1 G.711 Codec

G.711 is a codec that was introduced by ITU in 1972 for use in digital telephony, i.e. in ISDN, T.1 and E.1 links. The codec has two variants: A-Law is being used in Europe and in international telephone links, u-Law is used in the U.S.A. and Japan. G.711 uses a logarithmic compression. It squeezes each 16-bit sample to 8 bits, thus it achieves a compression ratio of 1:2. The resulting bitrate is 64 Kbit/s for one direction, so a call consumes 128 Kbit/s (plus some overhead for packet headers). This is quite a lot when compared with other codecs. This codec can be used freely in VoIP applications as there are no licensing fees. More details about the G.711 codec are shown in the table (3.1).

3.7.2 G.729 Codec

G.729 is a codec that has low bandwidth requirements but provides good audio quality (MOS = 4.0). The codec encodes audio in frames, each frame is 10 milliseconds long. Given the sampling frequency of 8 kHz, the 10 MS frame contains 80 audio samples. The codec algorithm encodes each frame to 10 bytes, so the resulting bitrate is 8 Kbit/s for one direction ,more details about the G.729 codec are shown in the table (3.1).

3.7.3 G.723.1 Codec

G.723.1 was originally designed for delivering video and speech over regular PSTN phone lines and allows very low bit rate coding of a voice signal. It is a dual-rate speech codec designed for the H.323 and H.324 audio and video conferencing standards for compressing voice at toll quality. G.723.1 uses 30 MS long voice frames and encodes them at either 5.3 Kbps using Multi pulse LPC with Maximum Likelihood Quantization (MPC-MLQ) algorithm or 6.3 Kbps using Algebraic Code-Excited Linear Prediction (ACELP) algorithm. Including the look ahead delay of 7.5 ms, the minimum algorithm delay is 37.5 MS . A silence suppression technique is incorporated in G.723.1

Annex A which uses 4 bytes of silence insertion descriptor (compared to the 2-byte SID of G.729). Due to the high level of compression, the voice quality of G.723.1 is not high and the Dual-Tone Multi-Frequency (DTMF) or Fax tones cannot be reliably transmitted using it.[45]

3.7.4 GSM-FR Codec

The ITU-T GSM 06.10 Full Rate (FR) was introduced in 1987 and was the first speech codec used in GSM cellular networks. It uses Regular Pulse Excitation Long Term Prediction Linear Predictive (RPE-LTP) algorithm. The encoder uses a 13 bit PCM signal and generates voice streams at 13 Kbps. Although the voice quality is not very high, the computational complexity is very low which incited its popularity in the early GSM networks.

Basic characteristic of standard codecs are illustrated in Table 3.1 explains compare all above CODECs with each other to get insight knowledge.

Table 3.1: Comparison of voice compression codecs

Codec	Algorithm	standard	Bit Rate (Kbps)	Memory (word)	Frame length (ms)	MOS
G.711	PCM	ITU	64	1K	10	4.4
G.729A	CS-ACELP	ITU	8	2K	10	3.7
G.723.1	ACELP	ITU	5.3	3K	30	3.6
GSM-FR	RPE-LTP	ETSI	13	1K	20	3.6

CHAPTER 4

DESIGN AND IMPLEMENTATION

4.1 INTRODUCTION

This chapter discusses the network modeling, implementation and configuration, three models have been implemented for simulate, each network model has four scenario, for different audio codecs (G711, G723, G729, GSM), to test impact handover and Congestion on quality of VoIP based on mobile WiMAX , a network model would be designed to study the performance parameters (jitter, end-to-end delay, throughput and MOS) that affect the quality of voice transmission in mobile WiMAX networks, we have used student version of OPNET Modeler 14.5. To implement different scenario, this tool has well defined user interface and a rich set of modules where users can efficiently create suitable simulation environments.

4.2 SOFTWARE ENVIRONMENT

We are using the Optimized Network Engineering Tool (OPNET v14.5) software for our simulations, OPNET is a network simulator, it provides multiple solutions for managing networks and applications e.g. network operation, planning, research and development, network engineering and performance management. OPNET 14.5 is designed for modeling communication devices, technologies, protocols and to simulate the performance of these technologies.

4.2.1 OPNET Modelers

OPNET “Optimized Network Engineering Tools”, it is a computer software to simulate communication networks. Although there is large variety of simulation packages available in the market, OPNET Modeler has been chosen for simulation environment because of the following attractive features:

1. Includes hundreds of per-built models and used to study performance changes of network: organizational scaling, technology changes and application deployment, also set different parameters to see the desired output.
2. Provides variety of development environment that support the modeling of distributed systems and communication networks.
3. Provide huge number of documentation for the user to develop the network models.
4. Offer flexible and easy graphical interface for viewing the results.

5. Results from OPNET are easily interpreted. With comprehensive tools to display, plot and analyze time series, histograms.
6. Provides graphical editors to configure our own networks, edit our own devices, define our own packet formats, and design our own protocols.[46]

OPNET Technologies provides solutions for the academic research, for example assessment and improvement of wireless network technologies such as WiMAX (Worldwide Interoperability for Microwave Access), Wi-Fi, UMTS (Universal Mobile Telecommunications System) and seamless communication. Design and assessment of MANET protocols, analysis of optical network, and enhancement in the core network technologies such as IPv6, MPLS, and power management schemes in wireless sensor network [41]. Now a day OPNET is very useful software in research fields. The OPNET usability can be divided into four main steps. The OPNET first step is the modeling, it means to create network model. The second step is to choose and select statistics. Third step is to simulate the network. Fourth and last step is to view and analyze results. All these steps are shown schematically in the below figure 4-1

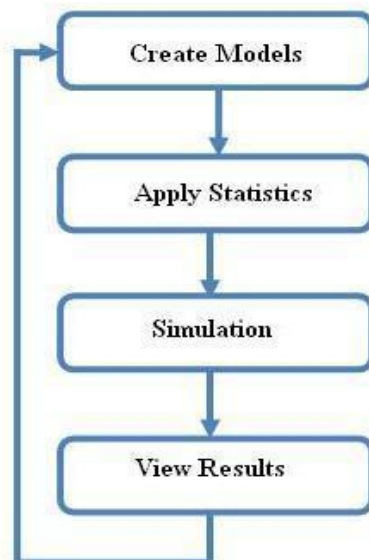


Figure 4.1: Flow chart of OPNET operator

Each object used in OPNET has certain characteristics. These characteristics are represented the form of data items called attributes. Attributes are useful for two primary purposes; to inform users about selected characteristics of an object; and to allow these characteristics to be modified for specific applications. The type of information that can be specified with attributes is general; it includes characteristics, object behavior, and internal structure.

4.2.2 Building Model

Run the OPNET modeler 14.5 to make a network model, the first step is to create a blank scenario by start-up wizard and the project editor workspace will be opened, the network design is done through two methods, one is automatically and the other is manually. The first method is automatically generating different topologies using rapid configuration. The second method is by dragging different kind of objects from the object palette to the project editor workspace, a user can also import some predefined scenarios from the hard drive. But however wireless network cannot be designed by importing scenarios, when the network is designed then the nodes need to be configured. This configuration can also be performed manually or by using pre-defined parameters in the workspace.

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 ASNGW supports the mobility in IEEE802.16e network and the VoIP_Server handles the VoIP traffic. The MNs moved along with the trajectories indicated by a white color line around the cells. 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 ASNGW is connected to the VoIP_Server by using 1000BaseT Ethernet cable which has 1Gbps capacity.

The simulation is run for 40 minutes, the statistics that we are going to concentrate on are MOS, Throughput, and Packet End to End Delay, jitter.

Steps to develop the scenario are as follows:

- ✓ Start OPNET Modeler by double-clicking on the OPNET Modeler 14.5 icon.

- ✓ Select File > open.
- ✓ A new window appears. Select Project from the menu and click OK.
- ✓ Another window appears and asks for project name and scenario name.
- ✓ Enter project name and scenario name.
- ✓ Uncheck the check box to not use the startup wizard and click OK.
- ✓ Select Topology > Deploy Wireless Network. This launches the Wireless Network Deployment Wizard as shown in Figure (4.2).

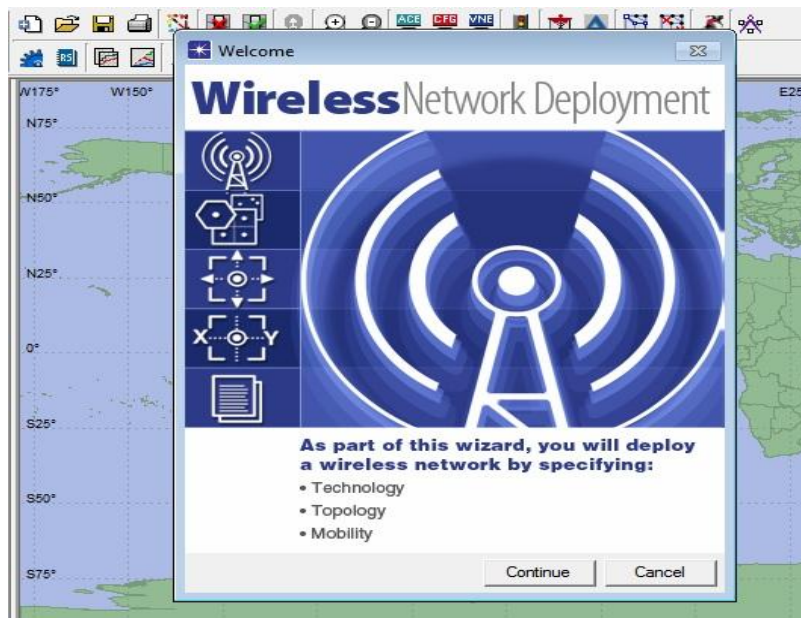


Figure 4.2: Deploy Wireless WiMAX Network

- ✓ Click Continue and next until the Technology tab is reached. Select the technology as WiMAX and set the Subscriber node transmission power as 0.005 W and base station transmission power. as shown in Figure (4.3)

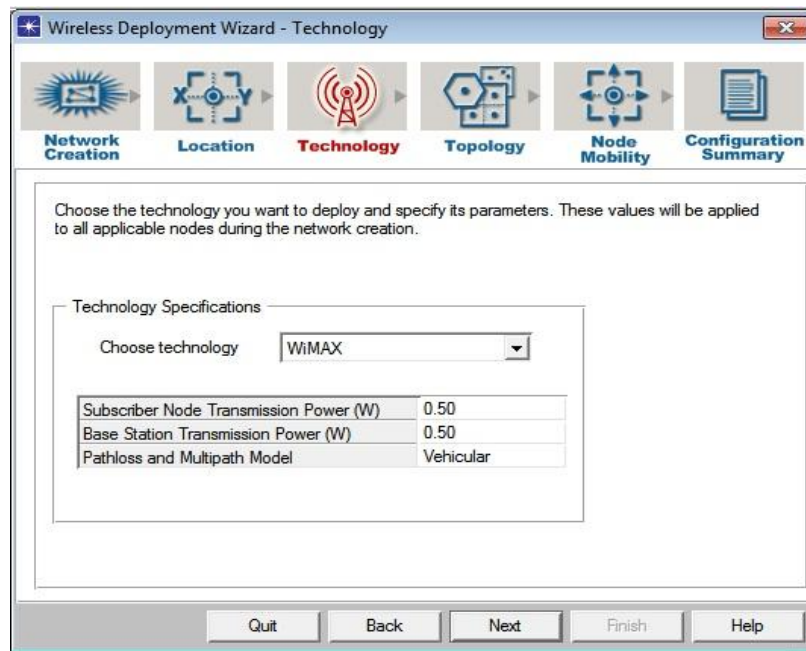


Figure 4.3: WiMAX Technology Specification

- ✓ In the topology tab choose the geographical overlay Cell (Hexagon), the number of cells to be 4 and the cell radius to be 1 Km and from the drop down menu select the node placement to be Random. The settings are shown in figure (4.4).

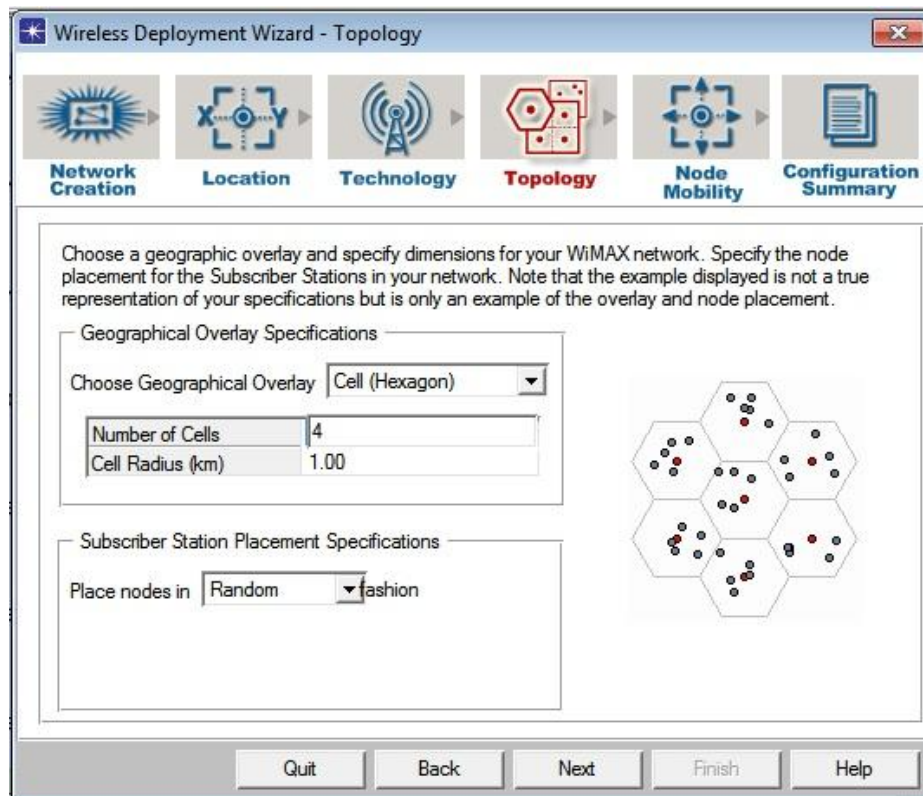


Figure 4.4: WiMAX Technology

- ✓ On clicking next the second tab of topology appears. Select the Access point as *wlan_ethernet4_slip4-adv* from the dropdown list with a number of 1, changes the mobile node to *wimax_ss_wkstn_adv* with a number of 1. Click Next.
- ✓ The node mobility tab appears. Leave unchanged. Click Next.
- ✓ The configuration summary appears in figure (4.5).

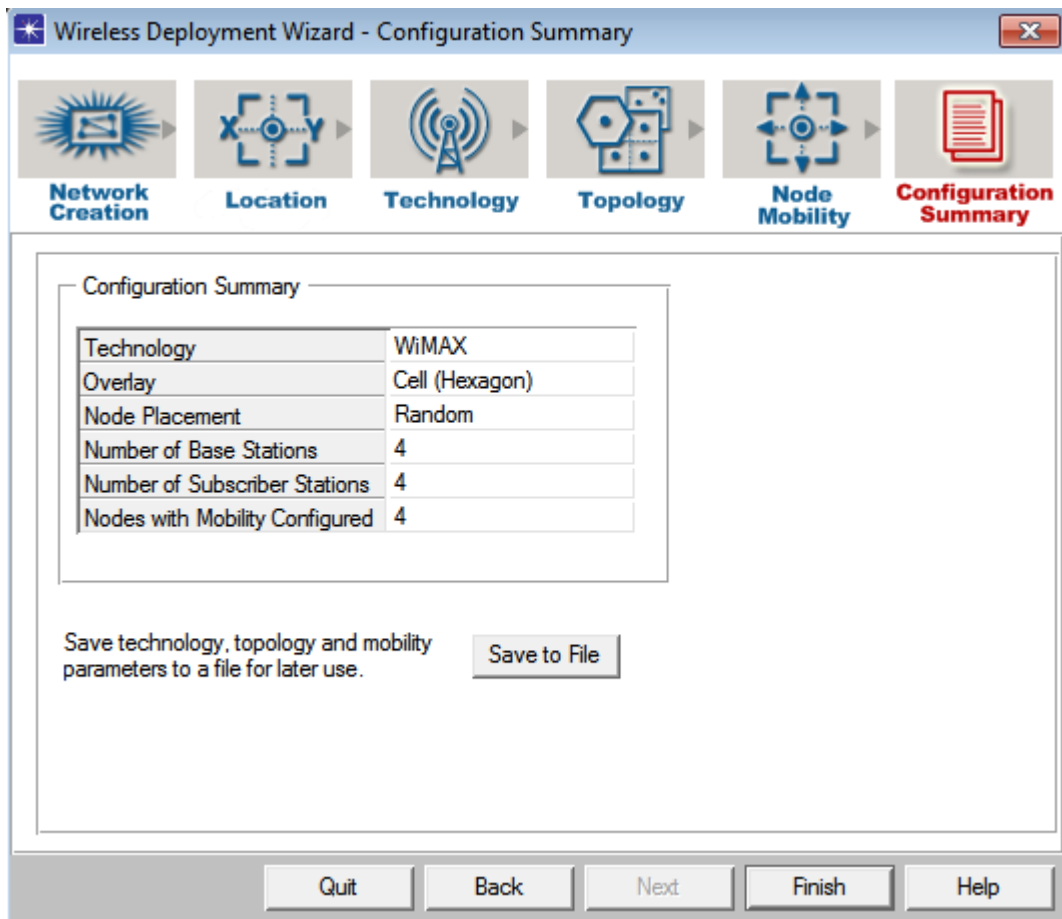


Figure 4.5: configuration summary (WiMAX network)

- ✓ To begin with mobile configuration, Right-click on the mobile node, and select Edit Attributes.
- ✓ The deployed scenario contains a WiMAX Configuration tool and Mobility Tool, both of these tools are outside the logical WiMAX Network. Use the menu Edit > Find Node/Link to locate both tools then cut and paste inside the WiMAX logical subnet.
- ✓ To select server and router, Click on the palette icon (), scroll to the internet toolbox select *ethernet_server* and drag and drop it on the work space. Similarly, drag an *ethernet4_slip8_gateway* and drop it on the workspace.
- ✓ Now scroll up to the Links and drag a 1000 BaseT link to connect the server and the Router.
- ✓ From the object palette bring an Application Config and a Profile Config node to configure the application.
- ✓ Right click on the Application Config and select Edit Attributes

- ✓ Under Application Definition add a row.
- ✓ Enter Application Name as voice.
- ✓ Expand the Description menu select the value of voice menu to edit.
- ✓ The Application Configuration is configured as shown in figure (4.6).

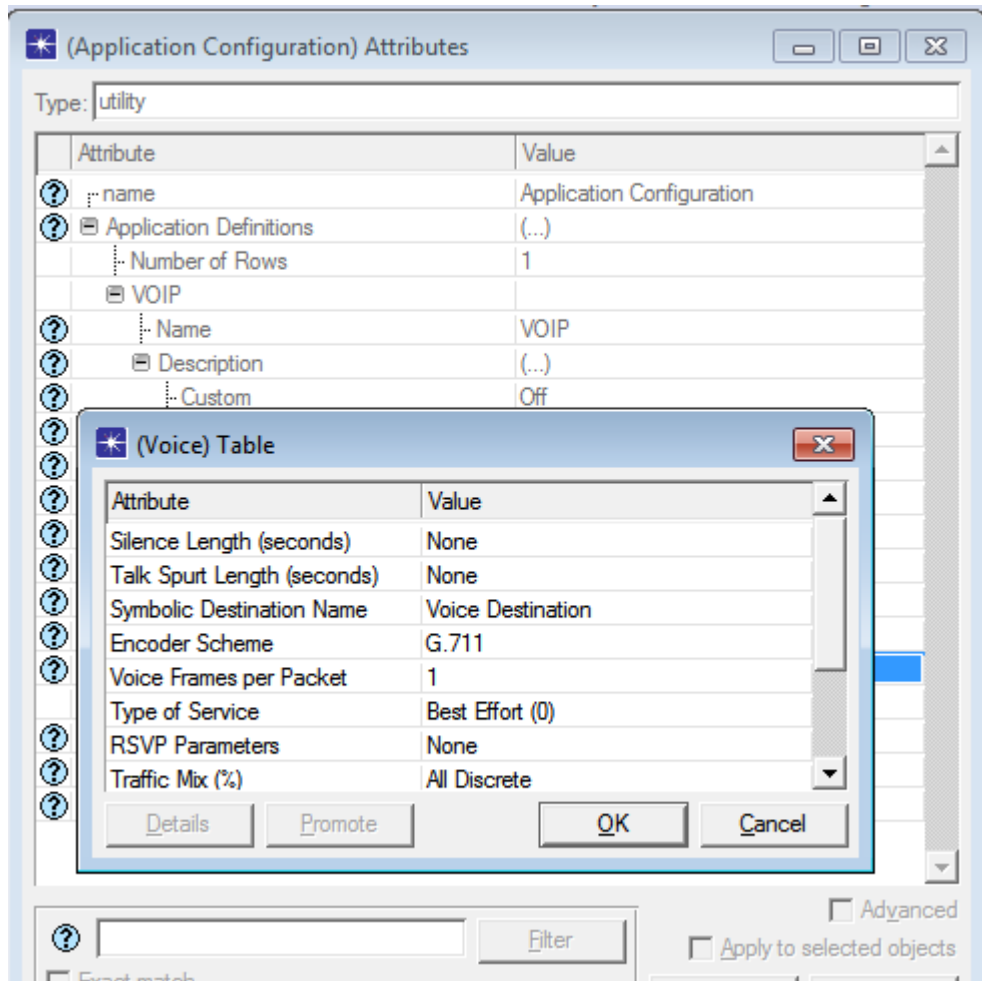


Figure 4.6: Application Configuration

- The Profile Configuration is configured as shown in figure (4.7).

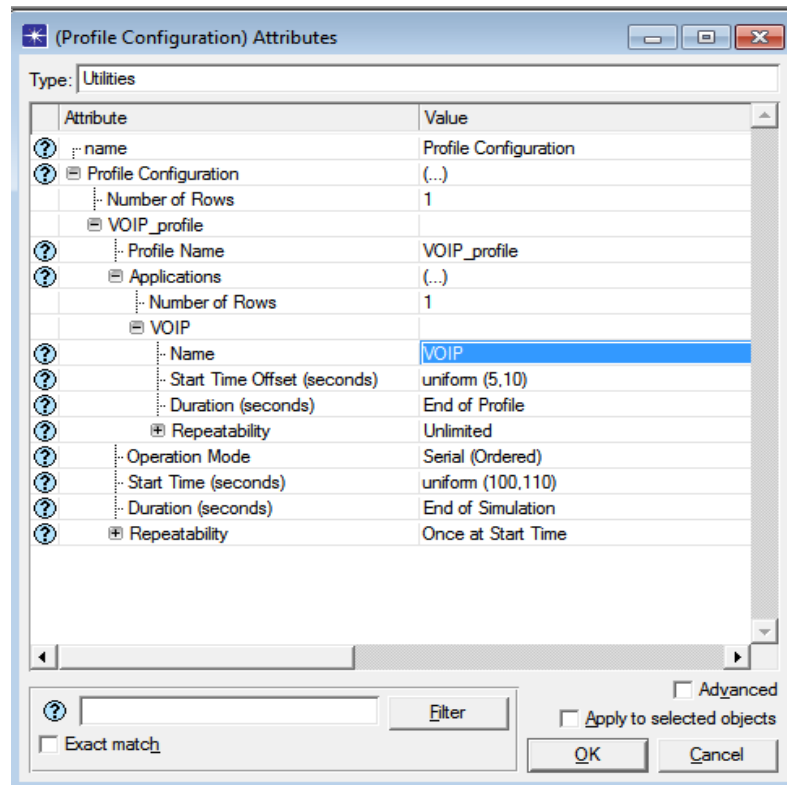


Figure 4.7: Profile Configuration

- ✓ Edit Attributes on the WiMAX_Config node. This node contains the global parameters of the WiMAX network and is deployed automatically when a WiMAX network is deployed. Settings are set as Figure 4.8.

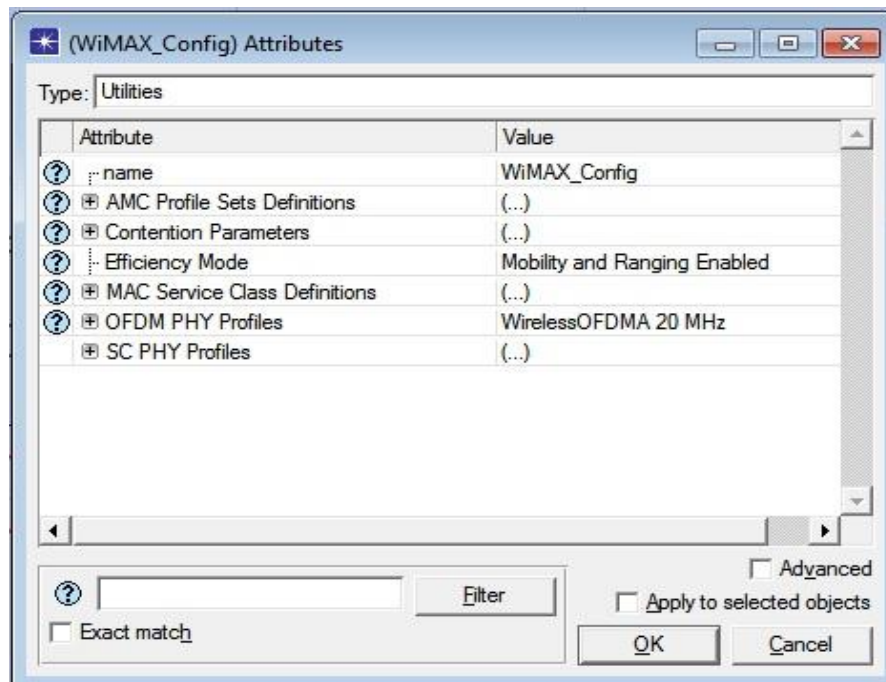


Figure 4.8: WiMAX Configuration

4.2.3 Simulation results and Statistics

In OPNET there are two kinds of statistics, one is Object statistics and the other is Global statistics. Object statistics can be defined as the statistics that can be collected from the individual nodes, on the other hand Global statistics can be collected from the entire network, when someone choose the desired statistics then run the simulation to record the statistics. These collected results are viewed and analyzed, to view the results right click in the project editor workspace and choose view results or click on DES, results then view results. Then a browser pops up as shown in this figure 4-2.

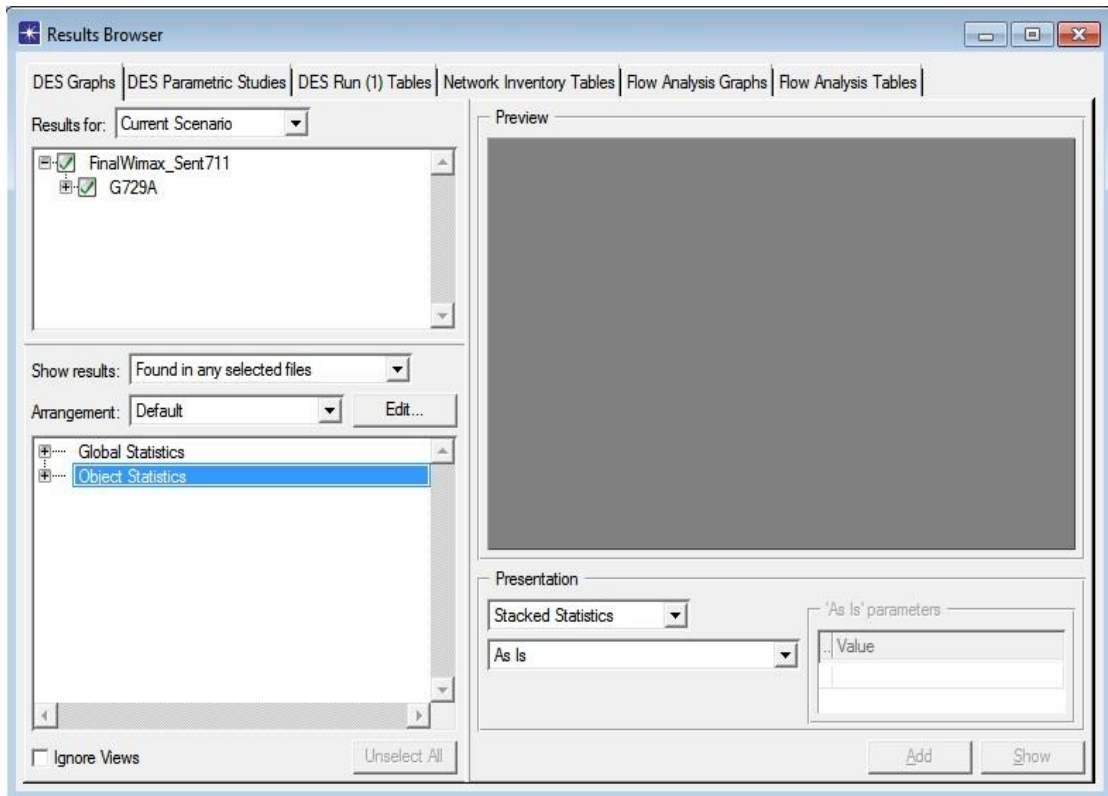


Figure 4.9: OPNET results browser

4.3 NETWORK SIMULATOR

4.3.1 Network Model

The configuration of the suggested Mobile WiMAX network model would be done using OPNET environment. OPNET simulator is chosen to its global recognition in terms different scenarios to evaluate the best voice encoding schemes of utilizing VoIP over Mobile WiMAX network, it is taken into account the impact of handover mechanism that occur during the movement of MN across the cells, in the proposed simulation, all scenarios have the same structure and topology. Divided network model to three models are model for fixed speed and model for variable speed and model for congestion.

- **Network Model for Fixed speed**

In our simulation model that show in Figure (4.10) ,we assume that MN is carrying voice traffic which is ongoing between mobile nodes and VoIP_Server .when the simulation time is started, mobile node (MN) initiates the first session with Base station_1 (Home agent) speed is 10 km/h. After 10 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; Shows Figure (4.10) network Model for Fixed speed.

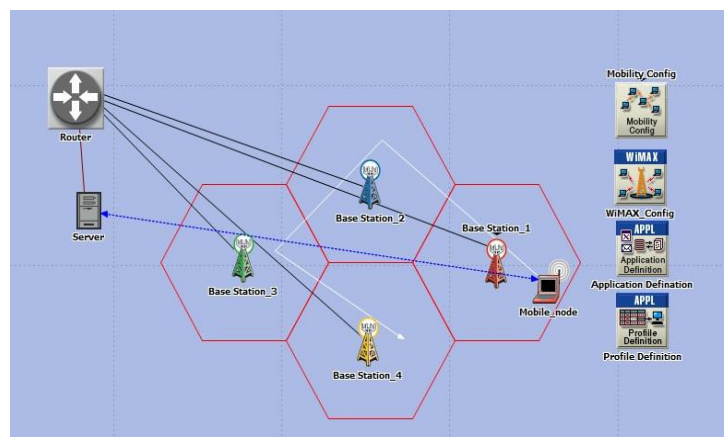


Figure 4.10: Network Model for Fixed speed.

- **Network Model for Variable speed**

The second network model consists of four hexagonal cells of WIMAX mobile networks, each cell consists of one BS(Base station) and four MS (Mobile station) Moving at different speeds are 10,20,30,40 km/h as show in Fig 4.11, white lines represent the trajectories set for mobile nodes to move around the network while simulation, network was modeled in area of 10000m x 6000m, we simulate four scenarios. Each scenario was run for 2500 sec (simulation time), all the simulations show the required results, under each simulation we check of Quality VOIP during handover process, multiple graphs from simulations like first for jitter, second is for the MOS, and third one is for the throughput, and fourth is for end to end delay, we configured the profile with VOIP application, The Table 4.1 show simulation parameters.

Table 4.1: network simulation parameters

Description	Value
Simulation Time	40 min
Simulation Area (m x m)	10000 x 6000
Number of MS	4
Type of Nodes	Mobile
Number of BS	4
MS speed	10-20-30-40 km/h

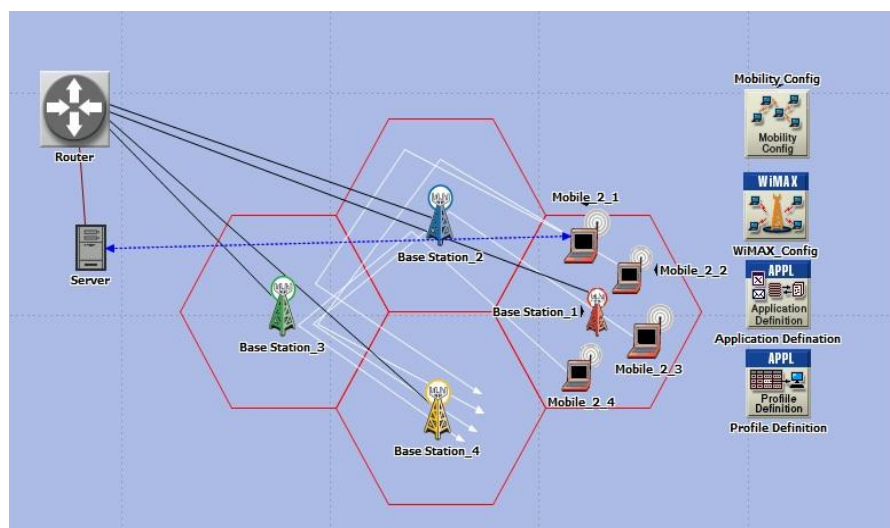


Figure 4.11: Network Model for Variable speed.

- **Network Model for Congestion**

In this model comparison was made between four types of audio codecs most commonly used, G711 and G723 and G729A and GSM-FR, During calls congestion in the WiMAX network, A set of voice calls is conducted by eleven mobile stations at the same time and at different speeds respectively (10-15-20-25-30- 35-40-45-50-55-60) km / h , The figure 4.12 show the simulation model for congestion scenario of different encodings.

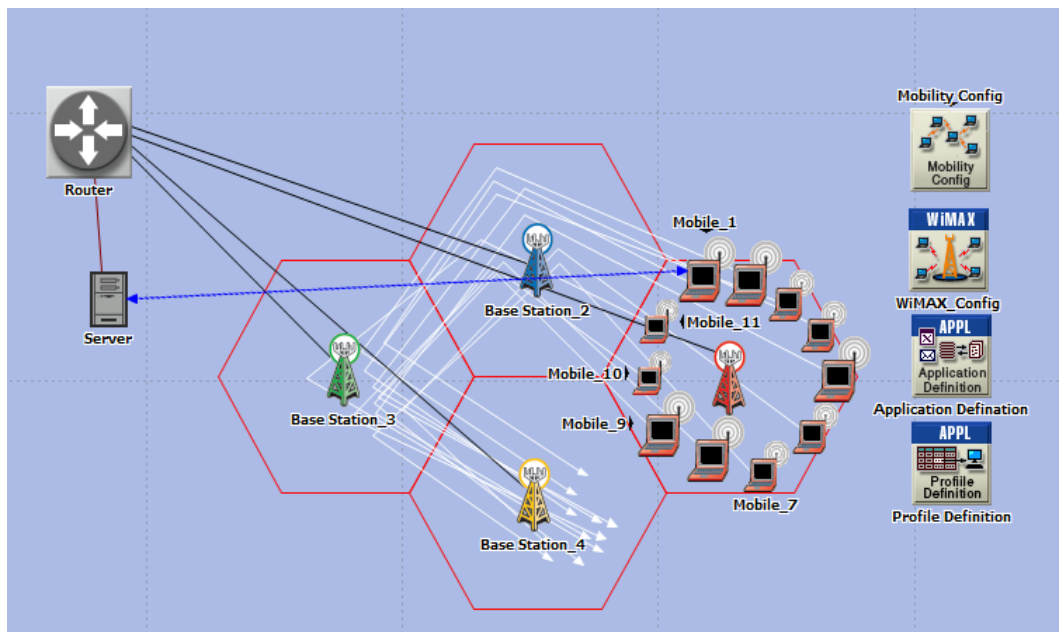


Figure 4.12: Network Model for Congestion.

4.3.2 ASN Gateway

Each BS connected to the ASN-GW by using the PPP DS link which has a 1.544 Mbps, show in Figure (4.10, 4.11, and 4.12) The ASN gateway typically acts as a layer 2 traffic aggregation points within an ASN. The ASNGW is connected to the VoIP Server by using 1000 Base T 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*'.

4.3.3 WiMAX parameters of Handover and BSs

The WiMAX parameters of Handover and BSs are given in Table (4.2) and Table (4.3) respectively.

Table 4.2: Handover Parameters.

Description	Value
MS Handover Retransmission Timer (ms)	30
Maximum Handover Request Retransmissions	6
Handover Threshold Hysteresis	0.4
Multi target Handover Threshold Hysteresis	0
Maximum Handover Attempts per BS	3

Table 4.3: Base Station Configuration Parameters.

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

CHAPTER 5

RESULTS AND DISCUSSION

5.1 INTRODUCTION

The results of the simulations are analyzed and discussed in this chapter, The results obtained in the form of graphs, all the graphs are displayed as average, the results are obtained after implementing the mobile WIMAX network simulation, and the statistics are collected by using OPNET Modeler 14.5 under four different scenarios for Voice codecs (G.711- G.723.1- G.729A- GSM FR) during handover And congestion. The collected statistics from the simulation include, Throughput (packet/sec), Mean Opinion Score (MOS), jitter (sec) and Voice packet end to end delay (sec).

5.2 ANALYZING SIMULATION

The four scenarios were run in the OPNET Modeler 14.5. We run the simulation for 40 min i.e. 2400 sec and save the graphs for analysis and calculation. These graphs were found very helpful in the statistical analysis of the audio codecs. The required graphs were saved as the bitmap image for the statistical analysis. we will discuss handover for fixed and Variable speed and discuss Effect of congestion on Voice codecs.

5.2.1 Handover for Fixed speed

The results are obtained after implementing the IEEE802.16e network simulation, the node (MS) moved with speed 10 Km/h through a trajectory as show in Figure 5.1, the statistics are collected by using OPNET Modeler under four different scenarios. The collected statistics from the simulation include the following parameters:

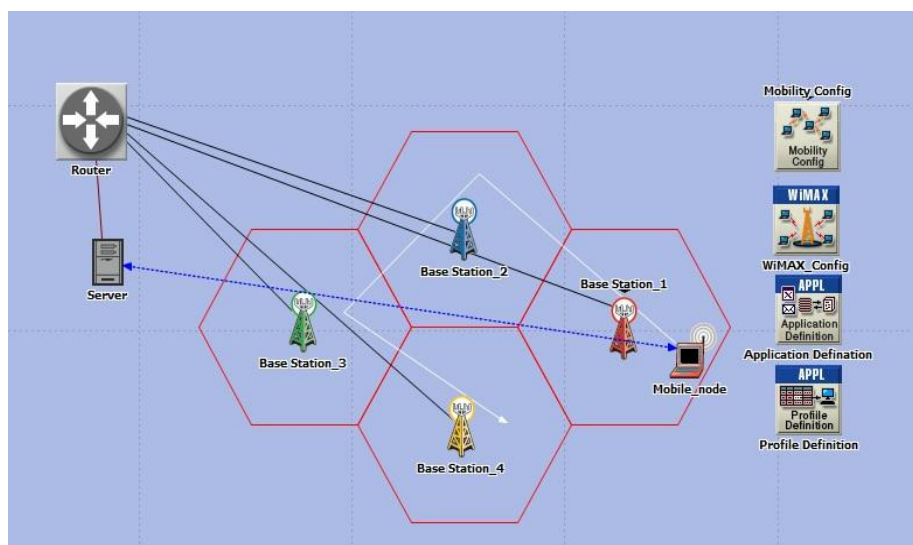


Figure 5.1: network topology

A. Throughput

Throughput is a measure of how many units of information a system can process in a given amount of time. Throughput is usually measured in bits per second (bit/s or bps) [47]. Fig 5.2 shows comparative result of Throughput for codecs that are used in this experiment. According to the gained results, it is observed that the maximum rate of throughput is recorded for the codec G.711 with the rate of 150,000 bit /sec, On the other hand, G.729 and GSM-FR codecs have the lower throughput rates. The throughput in G.729 codec is ranged from 10,000 bit/sec to approximately 60,000 bit/sec, while GSM-FR codec is ranged from 10,000 bit/sec to roughly 43,000 bit/sec. The lowest rate belongs to codec G.723.1 which is oscillated from 10,000 bit/sec to about 23,000 bit/sec. Table 5.1 shows a value Throughput for codecs various.

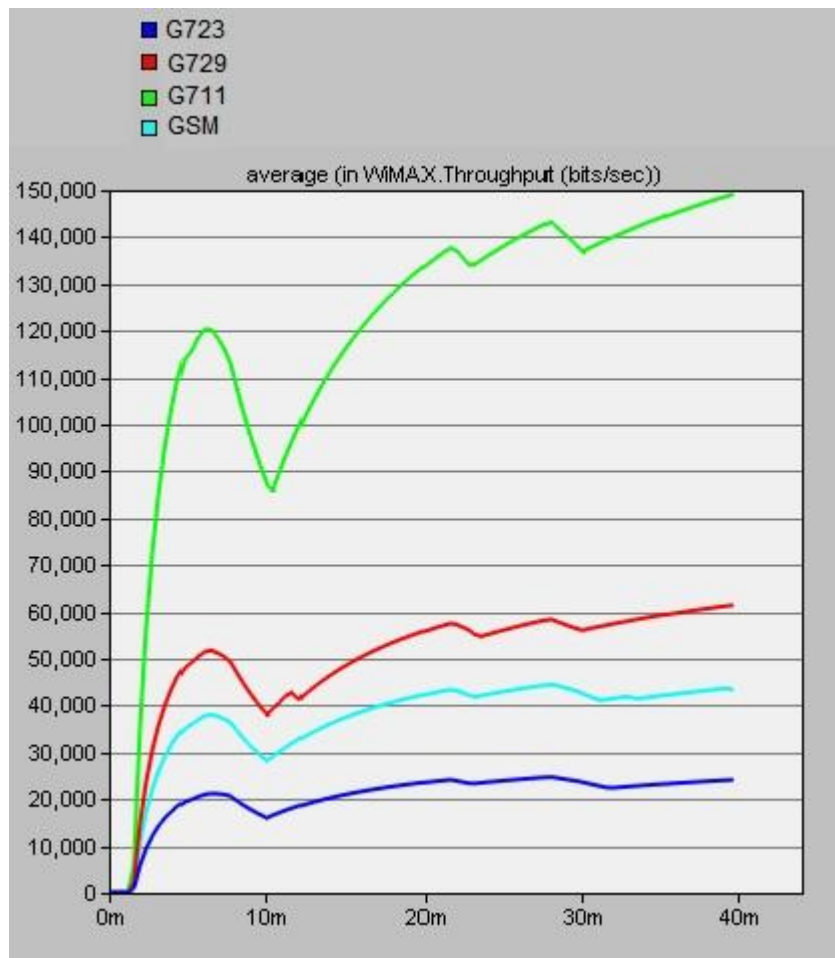


Figure 5. 2: Throughput of various audio codecs.

Table 5.1: Throughput for codecs various

Codec	Throughput (bit/sec)	
G711	10,000	150,000
G723	10,000	22,000
G729	40,000	61,000
GSM	20,000	43,000

Additionally, it can be clearly seen that throughput of mobile1_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 node will receive signals from two base stations when the setup of the new connection is complete 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

B. Packet End to End Delay

End-to-end delay or one-way delay (OWD) refers to the time taken for a packet to be transmitted across a network from source to destination [48], it is a common term in IP network monitoring. Figure 5.3 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 is 0.094 sec , the reasons for that are the high bits per frame which are required more time to process them.

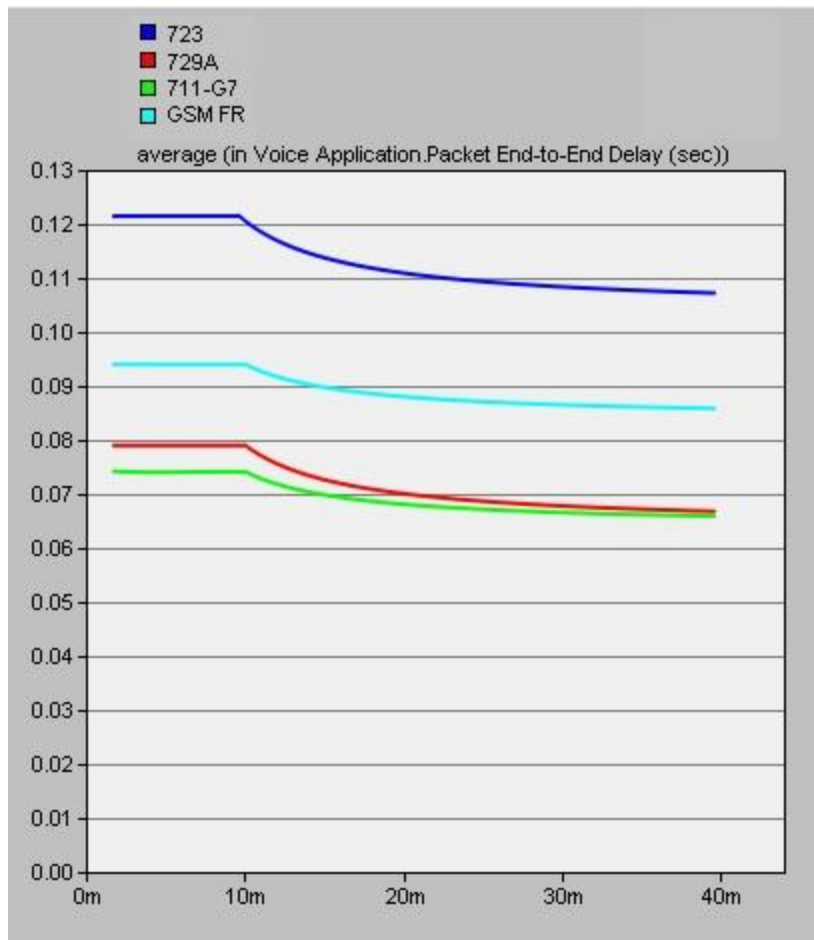


Figure 5.3: packet End to End delay (sec) under various audio codecs.

While, G.711 has small delay with only 0.075 sec due to the high transfer rate. This reason make the G.711 provide the best performance of VoIP applications over IEEE802.16e network; the results of the codecs G711, G729A are almost similar.

C. MOS

The Mean Opinion Score (MOS), recommended by ITU-T in 1996, is the most widely used subjective measure of voice quality. A MOS value is normally obtained as an average opinion of quality based on asking people to grade the quality of speech signals on the five point scale (Excellent =5; Good=4; Fair=3; Poor=2; Bad=1) under controlled conditions as set out in the ITU-T standard p.8009 [49]. Figure 5.4 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 The curves in Figure 5.4, the acceptable MOS value in our results is recorded for codecs G.711 and GSM-FR with a value of 3.2 and 3.1 respectively. y. 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.

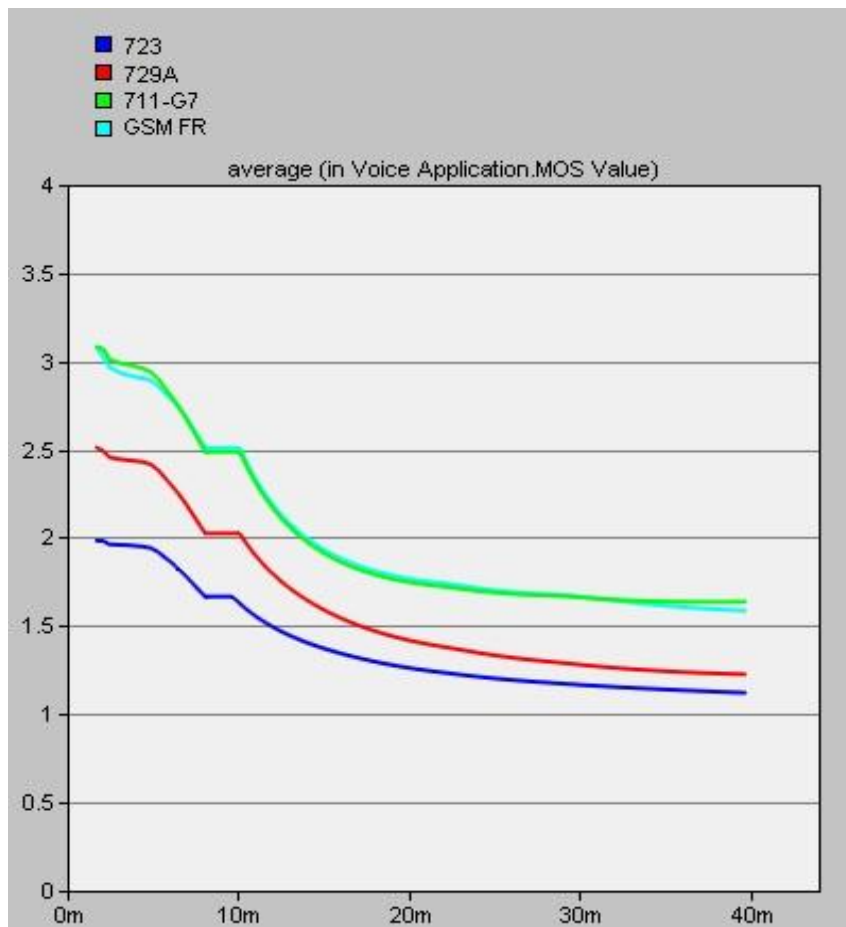


Figure 5.4: Average voice MOS under various codecs.

D. Jitter

Jitter is calculated as the signed maximum difference in one way delay of the packets over a particular time interval [50]. Generally, jitter is defined as the absolute value of delay difference between selected packets. Fig. 5.5 shows the comparative results of voice jitter for the codecs that are used in this experiment. It can be seen from the figure that the G.723 codec scheme has large value of jitter variation of (-0.00012), 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 0ms which shows that there is no jitter or delay variation between the VoIP packets. The voice jitter for G.723 is almost identical to G.711; Table 7 Show jitter values for different codecs.

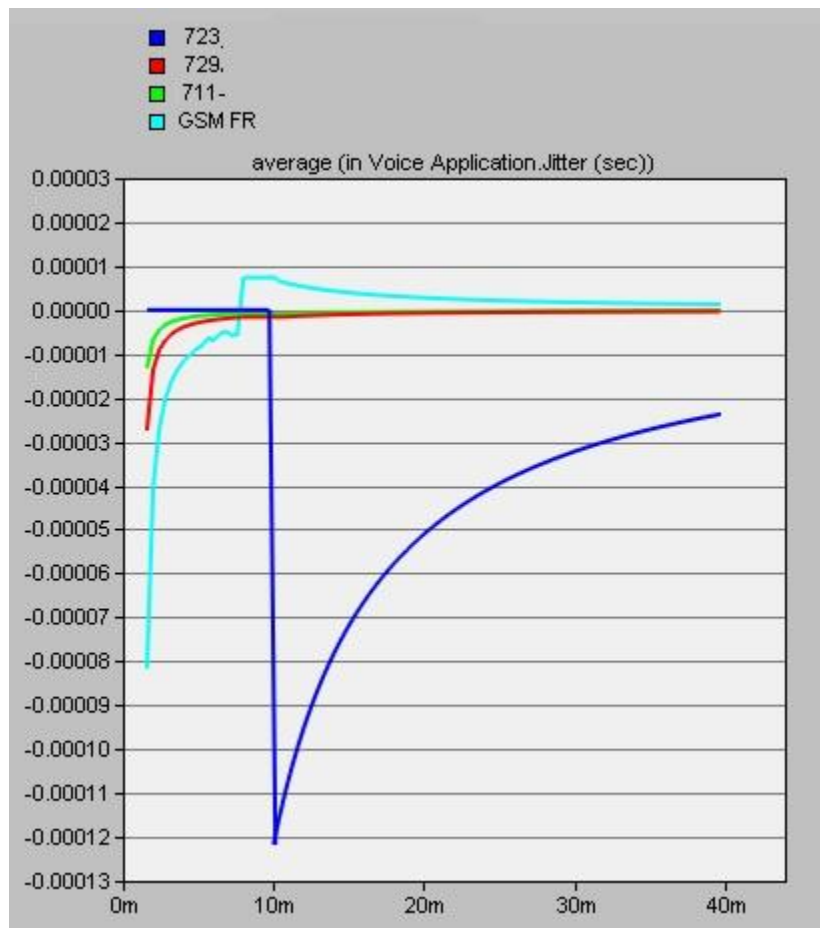


Figure 5.5: Average voice Jitter under various codecs.

Table 5.2: jitter values for different code

codec	value	
G711	-0.00001	0
G723	-0.00012	0
G729A	-0.00003	0
GSM-FR	-0.00008	0.00001

5.2.2 Handover for Variable speed

The results are obtained after implementing the mobile WIMAX network simulation, which contains on four MS (Mobile station) , Mobile 2_1, Mobile 2_2, Mobile 2_3, Mobile 2_4 , moving at different speeds are 10 ,20 ,30 ,40 km ,respectively; white lines represent the trajectories set for mobile nodes to move inside the network, all the simulations show the required results, under each simulation we check of Quality of VOIP during handover process.

A. Throughput

Fig 5.6 shows comparative result of Throughput for codecs that are used in this experiment. It is observed that the maximum rate of throughput is recorded for the codec G.711 with the rate of 350,000 bit /sec as Figure: 5.6 a, G729A codec have throughput rates 160,000 bit /sec as Figure: 5.6 c, the lowest rate belongs to codec G.723.1 which is oscillated from 40,000 bit/sec to about 45,000 bit/sec as Figure: 5.6b, additionally, increased speed mobile nodes give rise to lower throughput rates.

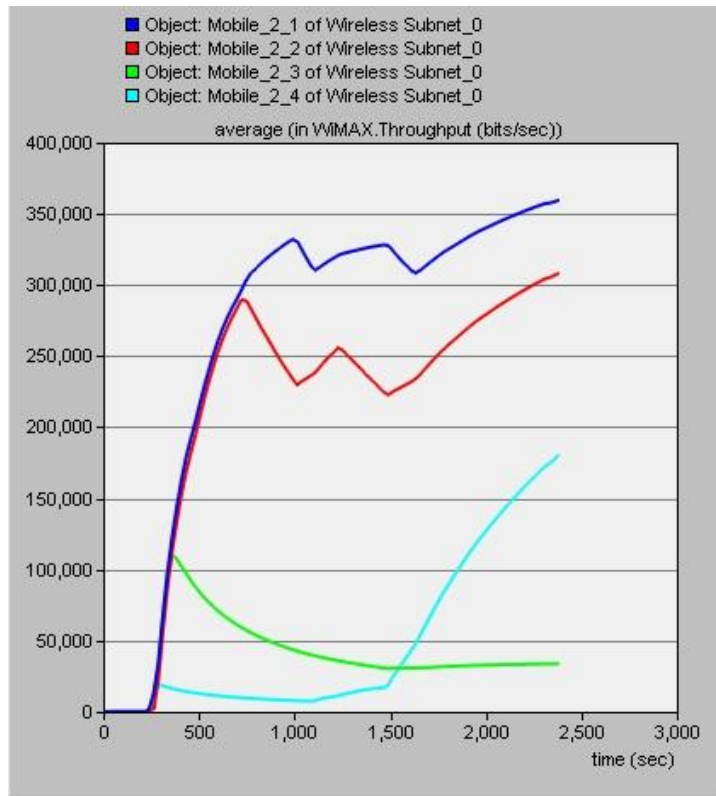


Figure 5.6a: Throughput for G711 audio codec

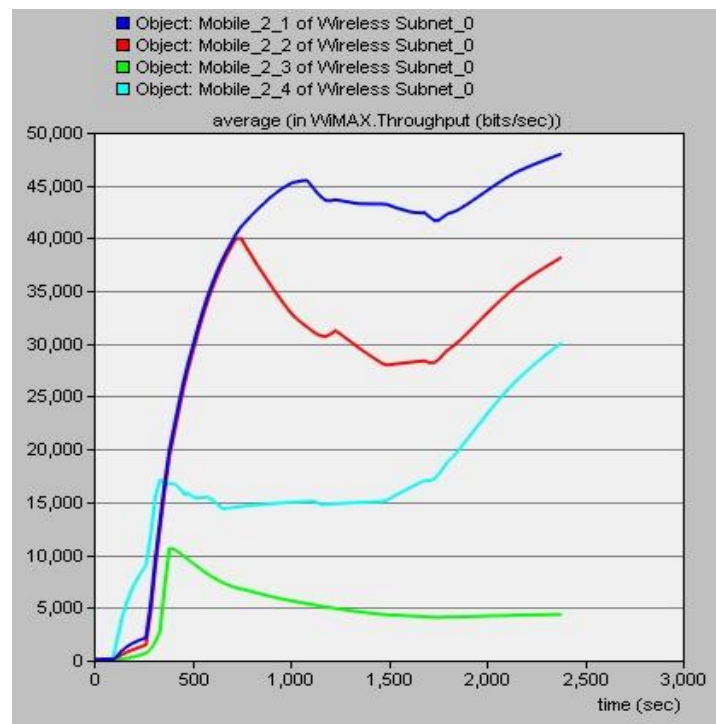


Figure 5.6 b: Throughput for G723 audio

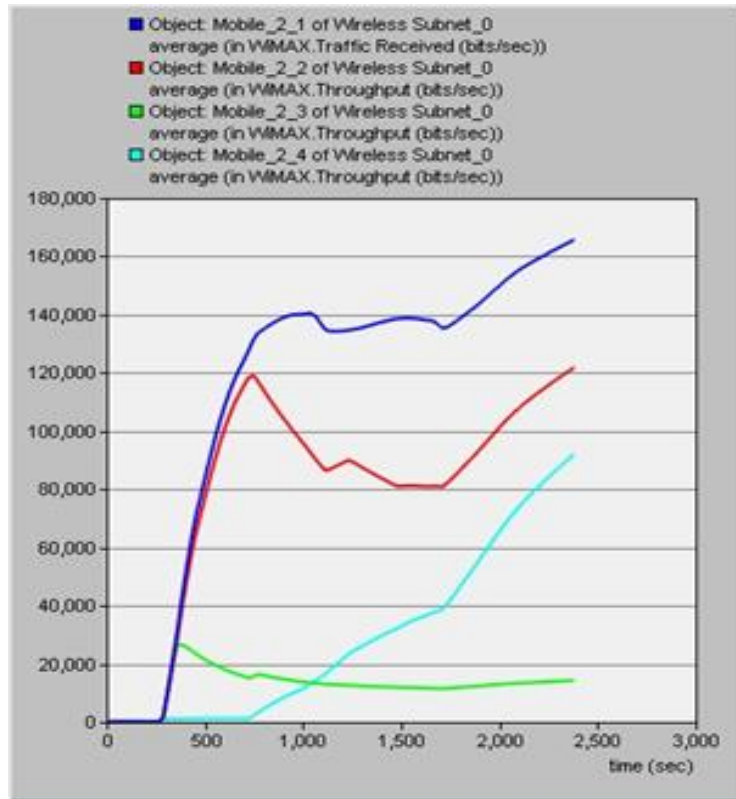


Figure 5.6 c: Throughput for G729A audio codec

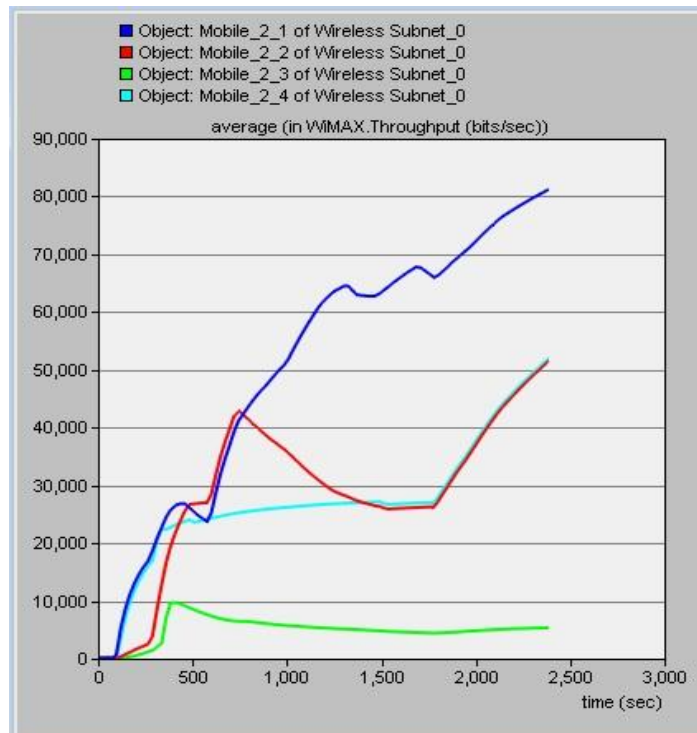


Figure 5.6 d: Throughput for GSM-FR audio codec.

B. Packet End to End Delay

End-to-end delay refers to the time taken for a packet to be transmitted across a network from source to destination. The end-to-end delay for all four scenarios is demonstrated in Figure 5.7 (a, b, c, d). The results show that G.723 codec scheme yields the highest voice packet end-to-end delay averaging around 0.13 sec, which is still an acceptable value. Other two codec schemes i.e., G.711 and G.729 yields the lowest voice packet end-to-end delay averaging around 0.85 sec. due to the high transfer rate and low packet size.

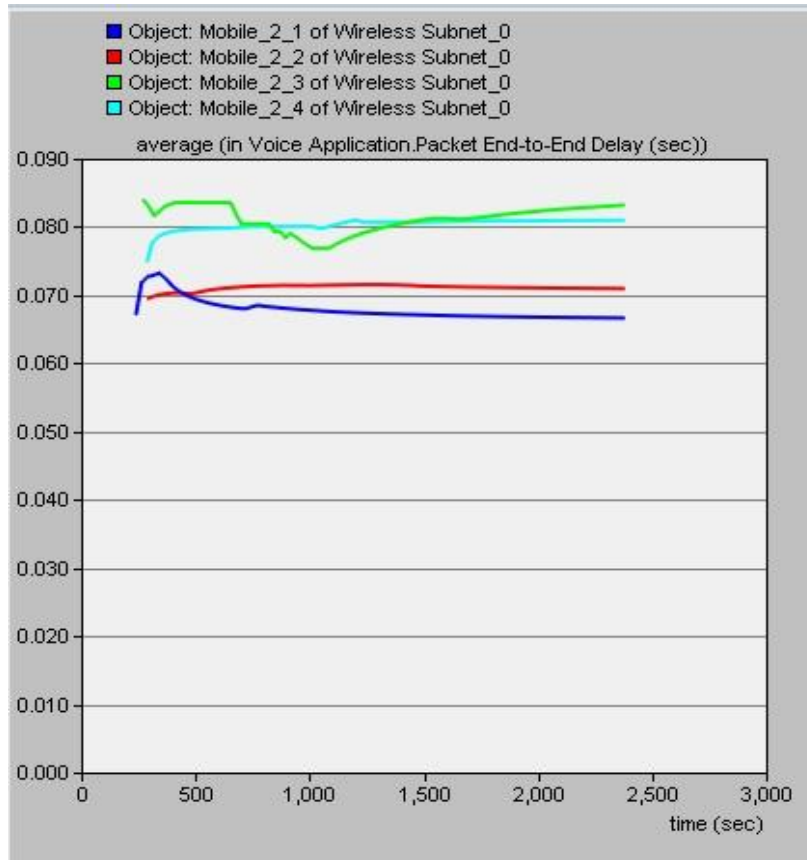


Figure 5.7 a: Packet End to End Delay for G711

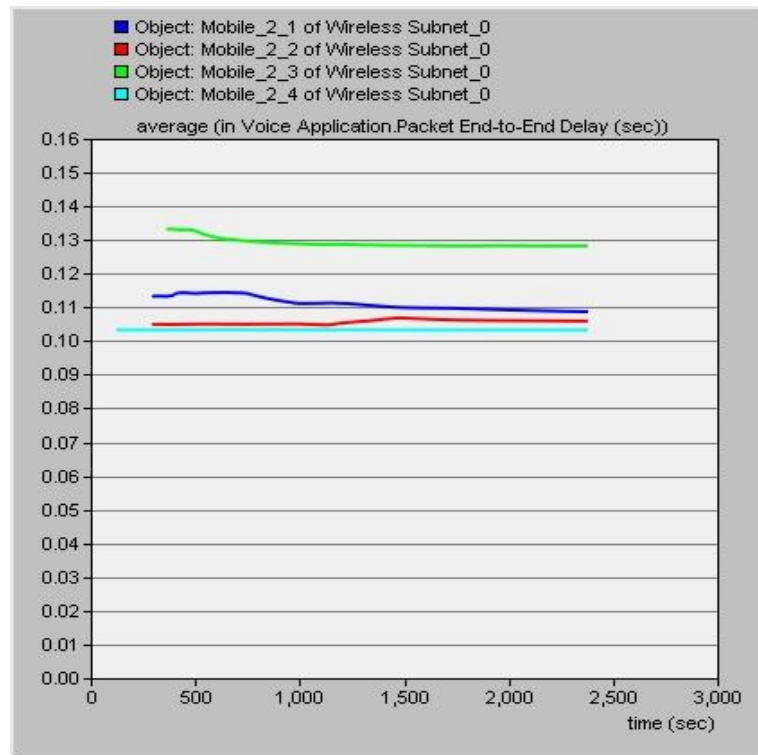


Figure 5.7 b: Packet End to End Delay for G723

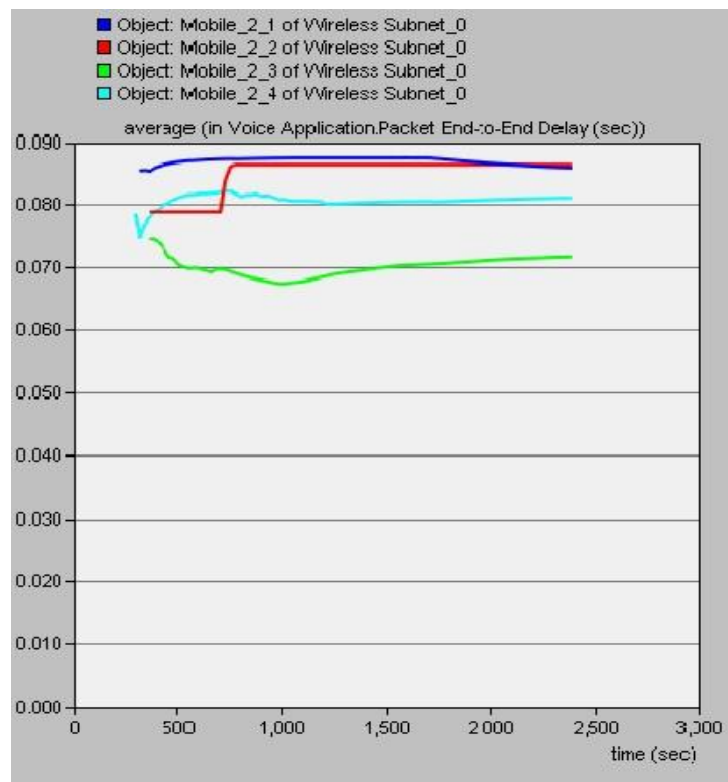


Figure 5.7 c: Packet End to End Delay for G729A

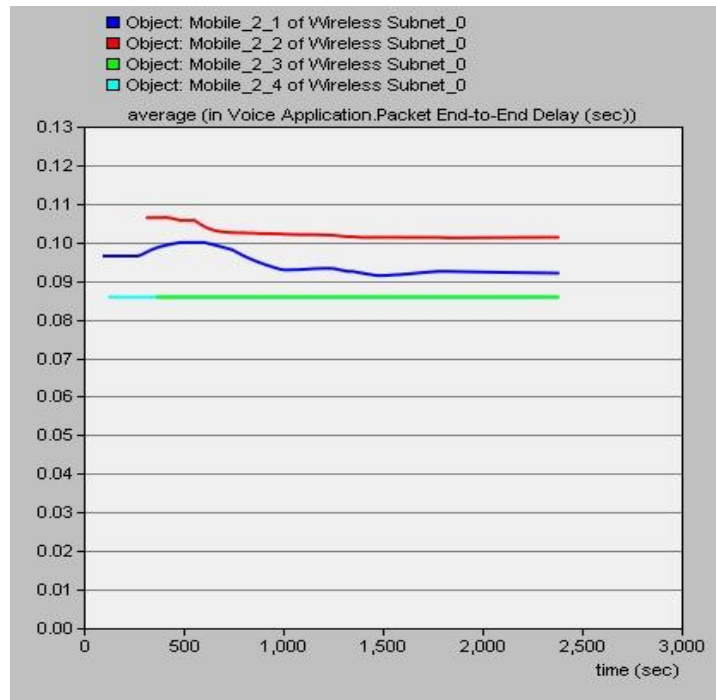


Figure 5.7 d: Packet End to End Delay for GSM-FR

C. Mean Opinion Score (MOS):

The curves in Figure (5.8a) show according to the gained results, the highest MOS value is recorded for codecs G.711 with a value of 3.6, for the mobile node (2-4) speedily 40 km\h. and the lowest MOS value is 1.7 for codec G.729 as shown in Figure (5.8c). Results of average voice MOS under various codecs are shown in Table 6.

Table 5.3: Average voice MOS under various codecs.

audio codecs	mobile nodes	Speed(Km\h)	value
G 711	Mobile 2-4	40	3.6
G723	Mobile 2-4	40	2.1
G729	Mobile 2-1	10	1.8
GSM	Mobile 2-1	10	3.3

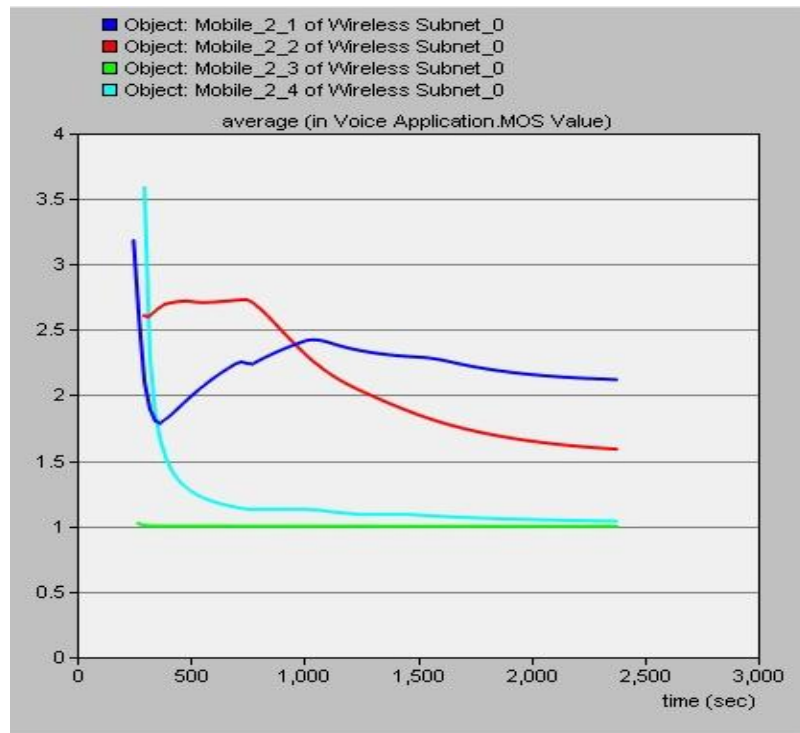


Figure 5.8a: MOS under G711 audio codec

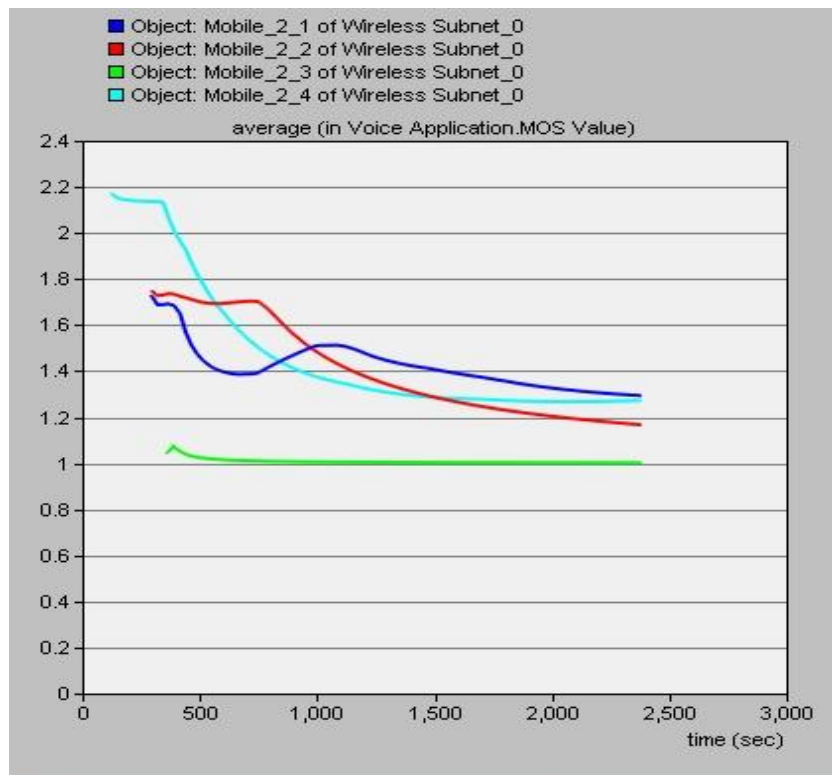


Figure 5.8b: MOS under G723 audio codec

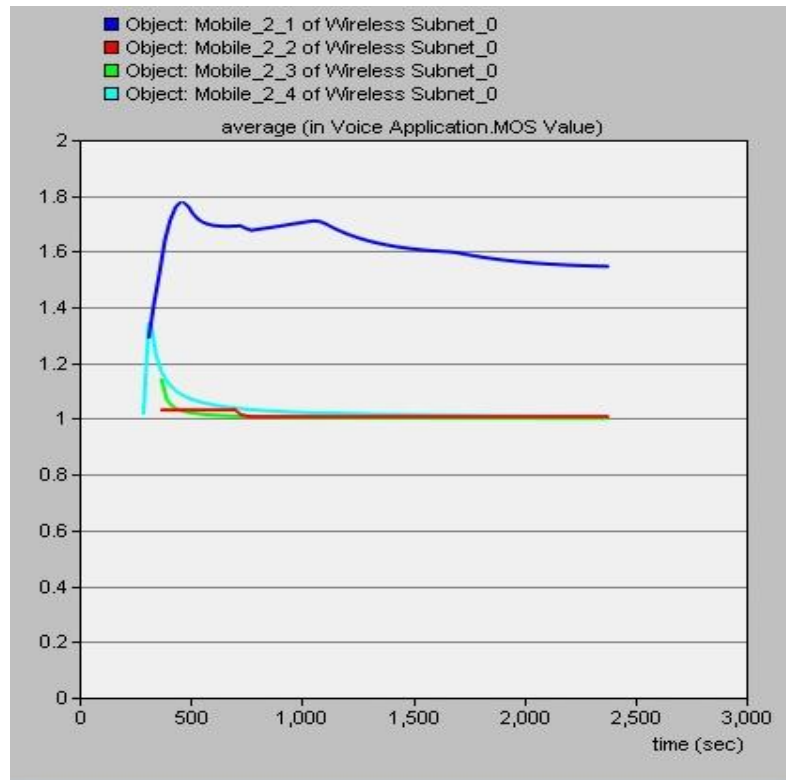


Figure 5.8c: MOS under G729 audio codec

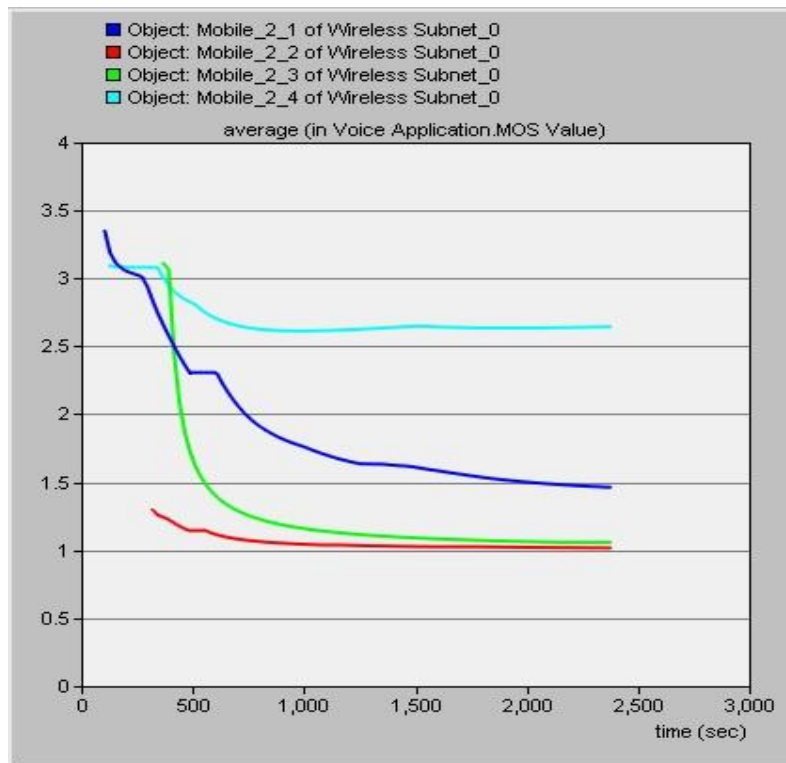


Figure 5.8d: MOS under GSM-FR audio codec

D. Jitter

Jitter it represents delay that resulted from starting transmit voice call until receiving it. Fig.5.9 (a, b, c, d) shows the comparative results of voice jitter for the codecs that are used in this experiment. it can be seen from the figure 5.9d that the GSM codec scheme has large value of jitter variation of (0.00037) sec and therefore yields highest curve among the codec scheme. in Mobile node 2-2. and result G.729 codec has smaller value of jitter variation is (- 0.00045) sec, the negative value of jitter means that the time difference between the packets at destination is less than that source. Table 6.5 Show jitter values for different codecs.

Table 5.4: Readings of Jitter vs Number of nodes.

No of Nodes	Jitter (sec)			
	G711	G723	G729A	GSM
Mobile node 2-1	0	0	0	0
Mobile node 2-2	0	0	- 0.00045	0.00037
Mobile node 2-3	0.00006	- 0.00001	0.00015	0
Mobile node 2-4	0	0	0	0

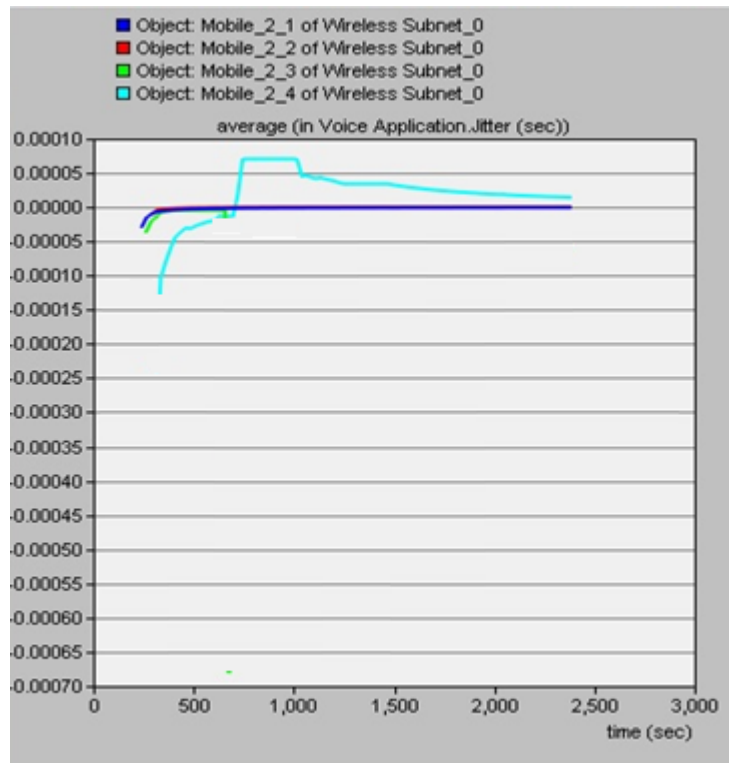


Figure 5.9a: jitter for G711 audio codec

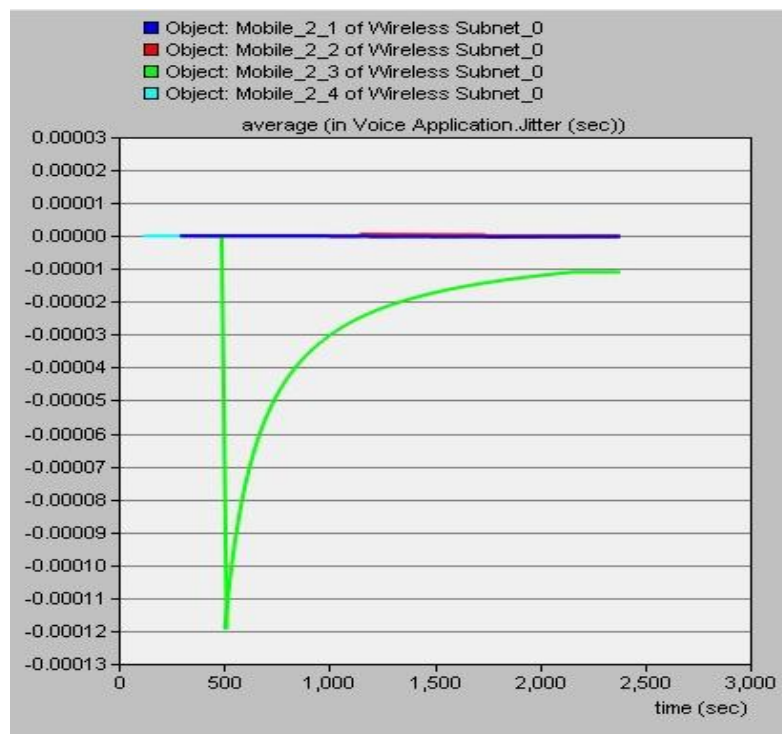


Figure 5.9b: jitter for G723 audio codec

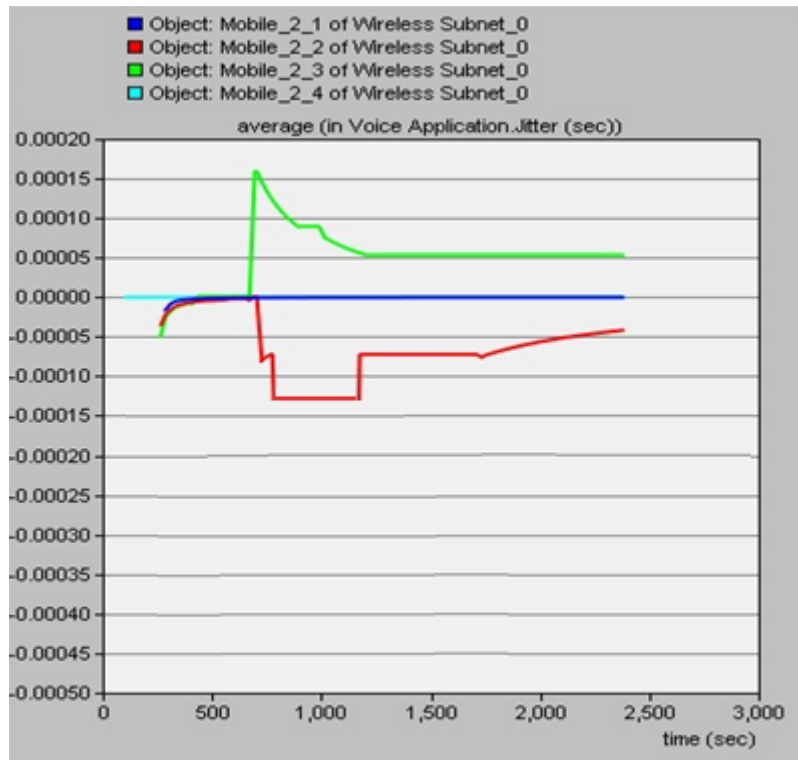


Figure 5.9c: jitter for G729A audio codec

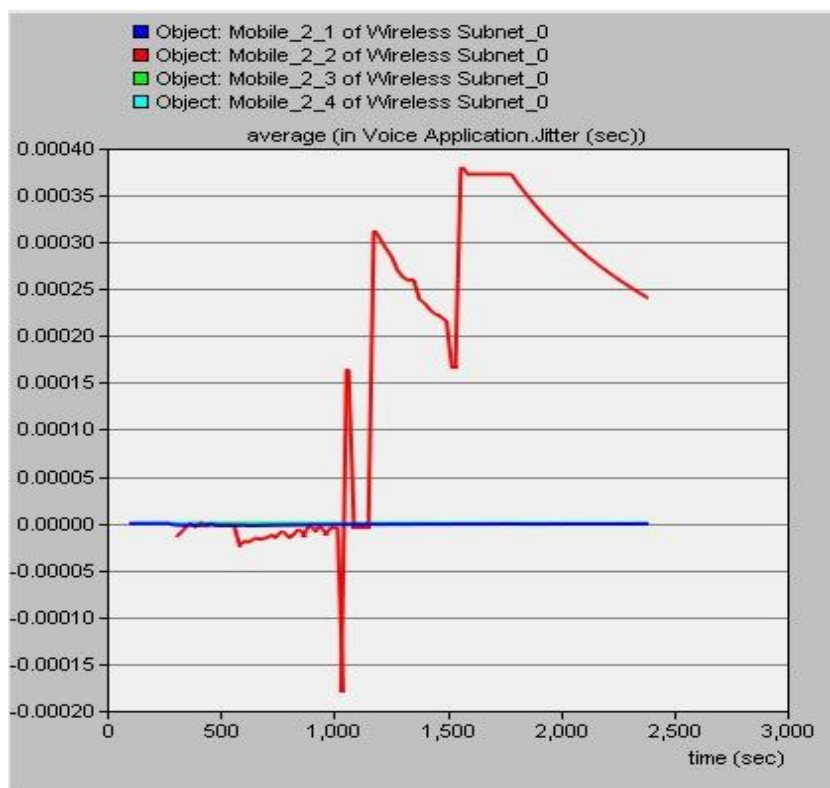


Figure 5.9d: jitter for GSM audio codec

5.2.3 Effect Handover Congestion on voice codec

In this scenario comparison was made between four types of audio codecs most commonly used, G711 and G723 and G729A and GSM-FR, during calls congestion over the WiMAX network, a set of voice calls is by eleven mobile stations at the same time and at different speeds respectively. the collected statistics from the simulation include:

A. Mean Opinion Score (MOS)

It is noted from Figure 5.10 that the Codec G711 has the highest value for MOS was 1.4, is Best quality, note decrease in MOS values when is congestion calls Compared with results MOS in the case MS Procedure one connection, Table 5.5 shows the MOS values for each audio codec in calls congestion and One call, codec G729A is the most affected by congestion, Has the lowest value.

Figure 5.10 shows graph curves for each code during call duration with estimated 40 minutes.

Table 5.5: Comparison value MOS in congestion and non-congestion

Audio codec	congestion	Non-congestion
G711	1.4	3.2
G723	1.25	2
G729A	1	2.5
GSM-FR	1.15	3.2

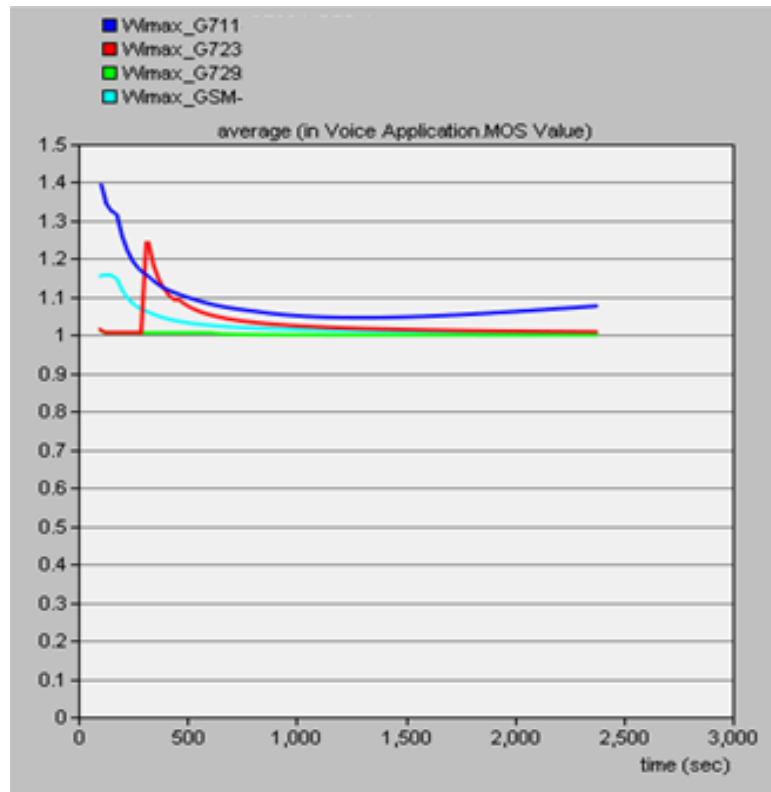


Figure 5.10: MOS for audio codec in congestion

B. Throughput

It is observed that the maximum rate of throughput is recorded for the codec G.711 with the rate of 550,000 bit /sec as Figure 5.11. while the G729A and GSM-FR encoders have roughly the same productivity rate of 150,000 bits per second. the lowest data transfer rate was 100,000 bits per second for the G723 encoding.

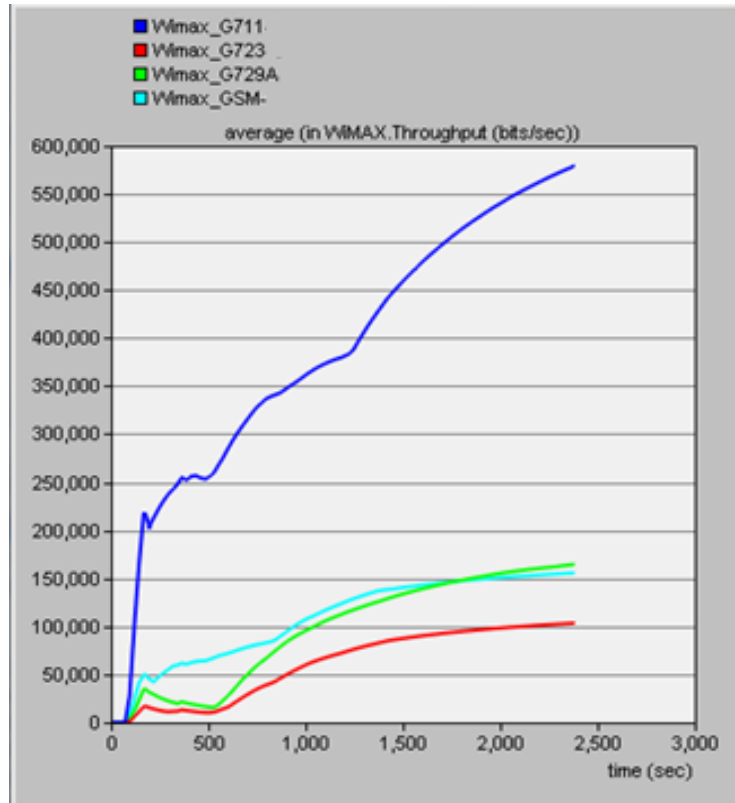


Figure 5.11: Throughput for audio codec in congestion

Throughput is the number of packets transmitted effectively in the network, Table 5.6 shows the difference between the productivity values in the case of congestion and in the case of a single call.

Table 5.6: Throughput in congestion and non-congestion

Audio codec	congestion	Non-congestion
G711	550.000	150.000
G723	100.000	25.000
G729A	151.000	60.000
GSM-FR	150.000	42.000

C. End to End Delay

The one-to-one delay refers to the time required to transfer packets over a network from one source to another, Figure 5.12 shows the results obtained between the average delay time from end to end and simulation measurement, it can be seen that the average end-to-end delay in G.723.1 is greater than 0.11 seconds, which gives lower performance compared to other codecs, The G711 and G729A codecs have no delay and their value is approximately zero 0.085.

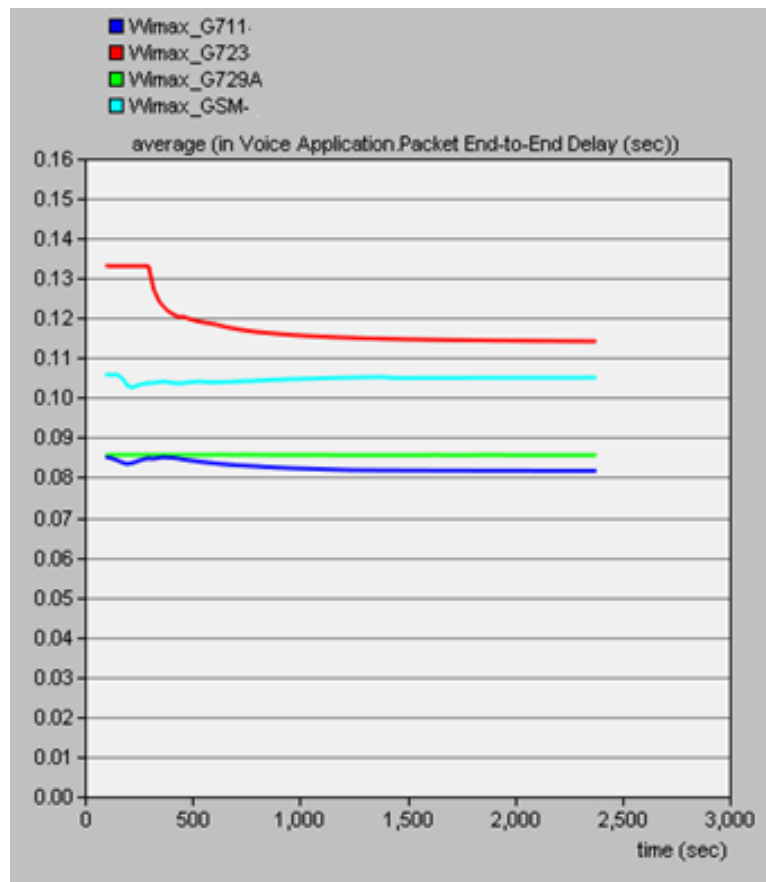


Figure 5.12: End to End Delay for audio codec in congestion

D. Jitter

Jitter It detects the differences in response time in the network caused by congestion, It measures the timing difference between packet sending and arrival. it can be seen from the figure 5.13 that the G.723 encryption scheme has great value for change in jitter (0.000005) seconds, the jitter average is 0 mille seconds for the G.711 encoding, indicating that there is no fluctuation in jitter or delays between VOIP packets.

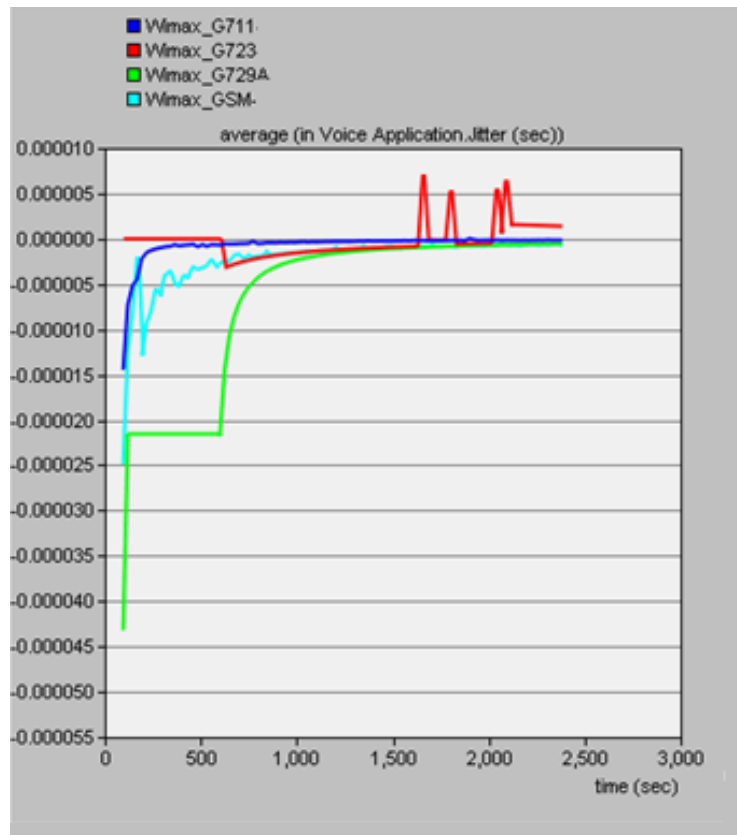


Figure 5.13: Jitter for audio codec in congestion

CHAPTER 6

CONCLUSION AND FUTURE WORK

6.1 CONCLUSIONS

In this thesis, OPNET Modeler was selected as a simulation tool for testing audio codecs impact on quality of VoIP based on WIMAX mobile, investigates an impact of handover on the speech quality in VoIP communication, and investigates in congestion impact on quality VoIP in handover process Simulation was done with the OPNET simulator, using four audio codecs, G711, G723, G729 and GSM, The parameters in this simulation are views jitter, packet end to end delay, MOS and throughput. The findings which have been reached to from simulation experiments results are summarized as following:

The Simulation results of Handover for Fixed speed

The obtained results show the acceptable MOS value in our results is recorded for codecs G.711 and GSM-FR with a value of 3.2 and 3.1 respectively, and 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.

is observed that the maximum rate of throughput is recorded for the codec G.711 with the rate of 150,000 bit /sec, i.e. the best performance; the lowest rate belongs to codec G.723.1 which is oscillated from 10,000 bit/sec to about 23,000 bit/sec.

Comparative result of jitter for codecs, Show the G.723 codec has large value of jitter variation -0.00012 sec.

It can be seen that the average end-to-end delays of G.723.1 are larger than 0.11 sec, This result is due to small transfer rate 5.3 Kbps for G.723.1 and high packet size, On the other hand, the average end-to-end delays of GSM-FR are larger than 0.09 second, the reasons for that are the high bits per frame which are required more time to process them, While, G.711 and G.729A has small delay with only 0.07 sec due to the high transfer rate and low packet size.

The Simulation results of Handover for Variable speed

The results are obtained after implementing the mobile WIMAX network simulation, Which Contains on four MS (Mobile station) are Mobile 2_1, Mobile 2_2, Mobile 2_3, Mobile 2_4 , Moving at different speeds are 10 ,20 ,30 ,40 km ,respectively.

According to the gained results, the highest MOS value is recorded for codecs G.711 with a value of 3.6, for the mobile node (2-4) speedily 40 km\h. and the lowest MOS value is 1.7 for codec G.729, It is observed that the maximum rate of throughput is recorded for the codec G.711 with the rate of 350,000 bit /sec, the lowest rate belongs to codec G.723.1 which is oscillated from 40,000 bit/sec to about 45,000 bit/sec, Note Increased speed mobile nodes give rise to lower throughput rates.

GSM codec scheme has large value of jitter variation of (0.00037) sec and therefore yields highest curve among the codec scheme.in Mobile node 2-2. And G.711 codec has smaller value of jitter variation is (- 0.00005) sec.

The results show that G.723 codec scheme yields the highest voice packet end-to-end delay averaging around 0.13 sec, and two codec schemes i.e., G.711 and G.729 yields the lowest voice packet end-to-end delay averaging around 0.85 sec.

Effect Congestion on audio codec in Handover

The obtained results show the acceptable MOS value in our results is recorded for codecs G.711 with a value of 1.4.

The maximum rate of throughput is recorded for the codec G.711with value 550,000 bit/sec, and lowest rate belongs to codec G.723.1.

The average end-to-end delay in G.723.1 is greater than 0.11 seconds, which gives lower performance compared to other codecs, The G711 and G729A codecs have no delay and their value is approximately zero 0.085.

Jitter rate is 0 mille seconds for the G.711 encoding, indicating that there is no fluctuation in jitter or delays between VOIP packets.

6.2 FUTURE WORK

WiMAX and LTE technologies are the main competitors in the mobile communication domain. 4G is an integrated system based entirely on IP technology, able to provide users with data, voice and multimedia services anywhere and anytime, with transfer rates up to 1 GB/s with high security and quality.

The future works aims at possibility of extension by studying LTE technology (Long Term Evolution), It is considered newer than the current standards in technologies series of mobile network, One of the biggest advantages of LTE its compatibility with all mobile technologies GSM, Thus the number of users is greater, and Transfer rates are slightly higher with LTE especially in the mobile department.

Comparison between LTE and WIMAX In terms of handover and Quality of Service (Qos), could be done in future.

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