


# The Libyan Academy

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## Test and Evaluate Quality of Service of VoIP over WiMAX and UMTS/HSPA Networks

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**A thesis Submitted to**

**Department of Electrical and Computer Sciences In Conformity with the Department Requirements as Partial**

**Fulfillment for Degree of Master of Engineering Sciences**

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# Abstract

The real-time service such as voice over internet protocol (VoIP) is considered the best type of conversational voice due to ability and the possibility of making a VoIP calls at anytime and anywhere with lowest cost. 3G/UMTS and WIMAX networks are considered among the most communication technologies that are used in this country and compete in this area. This thesis offers a comparative study between UMTS and WiMAX networks for VoIP services. The study focuses on VoIP quality when has sent under various networks loads such as HTTP, Email, FTB and Video Conferencing. In this thesis, the characteristics of voice as E2E delay and packet loss are investigated through simulations in OPNET modeler 14.5. These simulations have been designed in three scenarios for each network. The first one is VoIP network without QoS, the second one is the baseline VoIP network, while the third one is VoIP congested with different networks loads. Have been singled out these loads, with respect to UMTS network, the VoIP was congested in several cases with HTTP traffic only, while in WiMAX network the VoIP was congested in several cases with many types of traffic such as HTTP, Email, FTP and Video Conferencing. In particular, the effects of these traffic flows on voice traffic are examined to discover how voice traffic should be transmitted with the minimum delay and packet loss while it is in competing with others traffic for the bandwidth. The results obtained from UMTS and WiMAX simulations showed that voice E2E delay and packet loss occur more often and successively in cases of high load. In this case, the best performance of VoIP can be obtained in cases of low and medium load. Thus, the E2E performance and packet loss vary whenever the traffic load varies

# الملخص

تعتبرخدمة الوقت الحقيقي كخدمة الصوت عبر بروتوكول الإنترنت (VoIP) من افضل انواع الاتصالات لما تتميز به من قدرة فائقة وامكانية عالية لإجراء مكالمة هاتفية في أي وقت ومن أي مكان وبأقل كلفة. تعتبر شبكتى الجيل الثالث UMTS/(3G) و WiMAX من بين شبكات الاتصالات اللاسلكية التى تعمل فى هذا البلد و تتنافس فى هذا المجال. تقدم هذه الأطروحة دراسة مقارنة بين شبكتى الجيل الثالث UMTS/(3G) و WiMAX حول خدمة VoIP. الدراسة ركزت على جودة الصوت عندما يرسل تحت تاثير احمال مختلفة مثل HTTP, Email, FTP and Video Conferencing. في هذه الأطروحة، يتم التحقيق في خصائص الصوت مثل مقاييس تأخيروصول الصوت (E2E delay) وفقدان الحزمة (packet loss). من خلال المحاكاة في المحاكى OPNET14.5 تم تصميم هذه المحاكاة في ثلاثة سيناريوهات لكل شبكة, السيناريو الاول هو محاكات شبكة VoIP بدون استخدام جودة الخدمة (QoS)، و السيناريو الثاني هو شبكة VoIP مع استخدام جودة الخدمة في حين أن السيناريو الثالث هو اكتظاظ حركة VoIP بأحمال شبكات مختلفة. نخص بالذكر هذه الاحمال فيما يتعلق بشبكة UMTS تم اكتظاظ حركة VoIP بحركة HTTP فقط في حالات مختلفة ، بينما في شبكة WiMAX تم اكتظاظ حركة VoIP بحركة كل من HTTP, Email, FTP and Video Conferencing في حالات مختلفة. على وجه الخصوص، يتم فحص آثار هذا الاكتظاظ على حركة VoIP لاكتشاف الكيفية التي ينبغي أن يرسل بها الصوت مع الحد الأدنى من التأخير وفقدان الحزم أثناء وجوده في منافسة لعرض النطاق الترددي (bandwidth) مع التطبيقات الاخرى. فمن خلال النتائج التي تم الحصول عليها من المحاكاة وجد أن التأخير وفقدان الحزم للصوت يحدث في كثير من الأحيان وعلى التوالي في حالات الاحمال العالية من التطبيقات الاخرى. ففي هذه الحالة، أفضل أداء لVoIP يمكن الحصول عليه في حالات الاحمال المنخفضة والمتوسطة. وهكذا، فإن أداء تأخيرالصوت (E2E delay) وفقدان الحزم يزدادان كلما تزدادعليها الاحمال من التطبيقات المذكورة

# Acknowledgements

First, praise and thanks to ALLAH for the grace of faith and for helping me on the hardship of searching.

Second, I would like to thank my parents and my family for all the support they have provided me in all my life.

Next, I would like to express my sincere appreciation and thanks to my supervisor Dr. Ahmed Arara and Dr. Albahlul Elfgee for their guidance and for helping me with their comments on every step of this work.

Finally, I would like to thank all my Teachers at the Department of Electrical and Computer Engineering who have helped me improve myself scientifically.

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# List of Abbreviations

| | |
|---|---|
| 1G | 1st Generation |
| 2G | 2nd Generations |
| 3G | 3rd Generations |
| 3GPP | 3rd Generation partnership Project |
| 3GPP2 | Third Generation Partnership Project 2 |
| 4G | 4th Generations |
| AAA | Authentication Authorization and Accounting |
| ASN | Access Service Network |
| BS | Base Station |
| CDMA | Code Division Multiple Access |
| CN | Core Network |
| CS | Circuit Switching |
| DiffServ | Differential Service |
| DWRR | Deficit Weighted Round Robin |
| Email | Electronic Mail |
| FCFS | First Come First Serve |
| FDD | Frequency Division Duplex |
| FTP | File Transfer Protocol |
| FQ | Fair Queuing |
| GGSN | Gateway GPRS Support Node |
| GMSC | Gateway Mobile Switching Centre |
| GPRS | General Packet Radio Service |
| GSM | Global System for Mobile communication |
| GBR | Guaranteed Bit Rate |
| HSPA | High Speed Packet Access |
| HTTP | Hypertext Transfer Protocol |
| HLR | Home Location Register |
| IEEE | Institute of Electrical and Electronics Engineers |
| IMT | International Mobile Telecommunications |
| ISDN | Integrated Services Digital Network |
| LTE | Long Term Evolution |
| MIMO | Multiple Input Multiple Output |

| | |
|---|---|
| MTS | Mobile Telephone Service |
| MSC | Mobile Switching Centre |
| MSE | Mobile Switching Equipment |
| NGBR | Non – Guaranteed Bit Rate |
| OFDM | Orthogonal Frequency Division Multiplexing |
| OPNET | Optimized Network Engineering Tool |
| PS | Packet Switching |
| PSTN | Public Switch Telephone Network |
| PHY | Physical Layer |
| QoS | Quality of Service |
| QPSK | Quaternary Phase Shift Keying |
| QAM | Quadrature Amplitude Modulation |
| RNC | Radio Network Controller |
| RAN | Radio Network Access |
| SOFDMA | Scalable Orthogonal Frequency Division Multiple Access |
| SS | Subscriber Station |
| SISO | Single Input Single Output |
| SGSN | Serving GPRS Support Node |
| TDD | Time Division Duplex |
| UE | User Equipment |
| UTRA | Universal Terrestrial Radio Access |
| UTRA | UMTS Terrestrial Radio Access Network |
| TDMA | Time Division Multiple Access |
| WFQ | Weight Fair Queuing |
| UMTS | Universal Mobile Telecommunication System |
| WIMAX | Worldwide Interoperability for Microwave Access |
| WCDMA | Wideband Code Division Multiple Access |

# CHAPTER ONE

## Introduction

## 1.1 Introduction

Voice over IP (VoIP) as a communication technology supports transportation of voice data via Internet Protocol (IP) based networks, it is growing rapidly due to its increasing popularity. VoIP is a protocol that allows users to make calls over the IP networks. It is also referred to Internet telephony. VoIP is a technology that gets more useful and economical when users are on the move [1].This communication technology seems to have edge over circuit-switched Public Switched Telephone Network (PSTN) and Integrated Services Digital Network (ISDN). Consequently, many wireless technologies, such as Universal Mobile Telecommunications System (UMTS), Worldwide Interoperability for Microwave Access (WiMAX), and Long Term Evolution (LTE), are emerging to satisfy the users' growing requirements to provide anytime and anywhere access to the Internet. In fact VoIP call demands high Quality of Service (QoS). There is no network mechanism can ensures that data packets are not lost and are delivered in sequential order. Essentially, IP networks are a best effort service without QoS guarantees [2].

Real-time applications such as VoIP demands high quality because it is affected by several performance issues like voice packets end-to-end delay, jitter and packet loss as they are easily noticed irrespective of their percentage occurrence during voice/video data transmission. QoS features embedded in UMTS and WiMAX enable both networks to provide optimum transmission and delivery of interactive voice and video packets in any properly designed and configured access networks.

In order to eliminate the delay and packet loss, VoIP services equipped with mechanisms and algorithms to guarantee the service. By implementing QoS mechanism in IP networks will be offer guaranteed services regarding delay, jitter and packet loss. (Skype and Viber Media techniques are very well-known examples of successful and widely used VoIP applications that are used by millions of people over the entire world. VoIP should be use QoS features to get guaranteed services for optimum transmission and delivery of packets [3] [4].

In this thesis, the VoIP application is taken as an application scenario to study the differences of QoS performance between UMTS and WiMAX networks, in order to investigate how well these two networks cope with VoIP application. This study will help identify the strengths and

weaknesses of the two networks in terms of QoS mechanism and can study guides the applications to choose the best available network to use.

In this research, a simulation model in OPNET simulator is proposed. The system model is based on several scenarios that will be carried out extensively. The scenarios will analyze performance of packet end-to-end delay, jitter and packet loss for VoIP when loaded with different networks such as HTTP, Email, FTB and Video conferencing in both UMTS and WiMAX networks.

## 1.2 Problem Statement

Due to increasing number of real-time applications such as VoIP and video telephony, as well of other non-real-time applications such as (web browser, email, file transfer protocol (ftp) and video conferencing etc.), require wider bandwidth more over efficient resource management and service classification, the resources and corresponding bandwidth will be degraded. As a consequence, the traffic will be congested. Hence, the voice traffic could not be granted through the network. For these reasons, the QoS should be implemented in UMTS and WiMAX networks to provide better services. QoS is becoming one of the key issues to analyze and design networks and improve the services in real-time transmissions.

## 1.3 Aims and Objectives

Have been conducted a detailed comparative study between UMTS and WiMAX networks. To know how of these network can be able fulfill the requirement of VoIP telecommunication. The main objective of this thesis work is to study the performance of existing QoS in UMTS and WiMAX networks for VoIP under different traffic levels of HTTP application, furthermore observe the performance of VoIP quality when add more traffic loads such as Email, FTB and video conferencing. The study of such performance is based on some important metrics such as packet E2E delay, jitter and packet loss. From the evaluation of simulation results and analysis, the analyst will observe the performance of related parameters and recommends the following:

- Doing literature study about real-time and non-real-time applications within UMTS and WiMAX networks.

- Determine the appropriate frame size of HTTP traffic flow could UMTS and WiMAX networks been support through VoIP calling process.
- Determine the unacceptable frame size of HTTP traffic flow could be effective on VoIP calling in UMTS and WiMAX networks.

## 1.4 Scope of the Thesis

This thesis covers the technical issues and factors that need to be considered for implementation VoIP in the mobile networks. It discusses the challenging issues that need to be faced by mobile networks to transmit VoIP. This thesis gives a detail description about QoS related to 3rd generation/UMTS and WiMAX (IEEE 802.16e) networks utilized in Libya country. So this thesis may be helpful for those who interested in search in wireless communication field.

## 1.5 Research Questions

In the respect of traffic congested, QoS techniques became very handy in wireless networks such as UMTS and WiMAX. The outcome for this research depends on successfully answering these questions:

Q1. What is the impact on the packet end-to-end delay and packet loss performance for VoIP when GBR and Non-GBR bearers are established in UMTS and WiMAX networks?

Q2. What is the impact on the packet end-to-end delay and packet loss performance for VoIP traffic under different HTTP network loads?

Q3. What is the impact on the packet end-to-end delay and packet loss performance for VoIP traffic under HTTP, Email, FTB and video conferencing network loads with different bearers?

## 1.6 Research Methodology

The research methodology presented in this thesis is based on both qualitative and quantitative approach suggested by John W. Creswell [5]. In this qualitative approach, three steps are considered:

- Identify the key factors influencing VoIP performance in UMTS and WiMAX environment. It should be noted other researches that investigated this topic thererghly.

However, this research will focus on the extensive loading of HTTP, Email, FTB and Video Conferencing traffic on VoIP calls.

- Create suitable model to support VoIP and other applications.
- Justify the VoIP performance thresholds on the basis of strong facts and well known standards. For the purpose of the present study on quantitative approach, the UMTS and WiMAX networks models have designed on the work-space of the OPNET (Optimized Network Engineering Tool) modeler 14.5 with the help of different network entities. This methodology is widely used as an effective method to tune, debug and optimize the network infrastructures.
- With respect to this quantitative approach, following three steps are considered:
- Design network models need to validate or invalidate the hypothesis mentioned above. With respect to the present study, multiple experiments are deployed for VoIP calls without QoS and with QoS scheme to observe the performance of packet voice E2E delay metric. These experiments include different traffic loads to investigate the effecting of these traffic loading on VoIP quality.
- Collect quantitative data regarding packet voice end-to-end delay, jitter and packet loss.
- Evaluate the performance of different simulation scenarios in terms of VoIP when UMTS and WiMAX networks are deployed.

## 1.7 Motivation

VoIP is an attractive choice for voice transmittal compared to traditional circuit-switching technology for following reasons: lower equipment cost, integration of voice and data applications and wide availability of IP. VoIP service within packet-switching network can offer good QoS as the current circuit-switching services, but VoIP through packet-switching consumes lower bandwidth compared with circuit-switching networks. For this motivation VoIP could reduce their phone bills and make cost savings. Furthermore, VoIP services will operate seamlessly over a converged network referred to as a Next Generation Network (NGN).

## 1.8 Contribution

This thesis is focused on the comprehensive analysis for VoIP performance metrics such as end-to-end delay, jitter and packet loss for real-time applications over UMTS and WiMAX networks. Thesis will show the extent of changes on VoIP performance when voice traffic sent with set of other applications such as HTTP, Email, FTB and Video Conferencing. By simulate several scenarios will be determine the appropriate sizes of these applications that can be sent with VoIP. The model consists of UMTS and WiMAX scenarios as an input. The output of model will be a set of reports, tables, and curves as results. To meet various requirements and to reach the intended goals, six scenarios have been created for both networks to simulate different entities. Primarily, two scenarios are created for VoIP without QoS, and two scenarios for VoIP with QoS, and other two for congested VoIP with various traffic flows. Different sets of result from six scenarios will give us the opportunity to investigate the performance of the QoS.

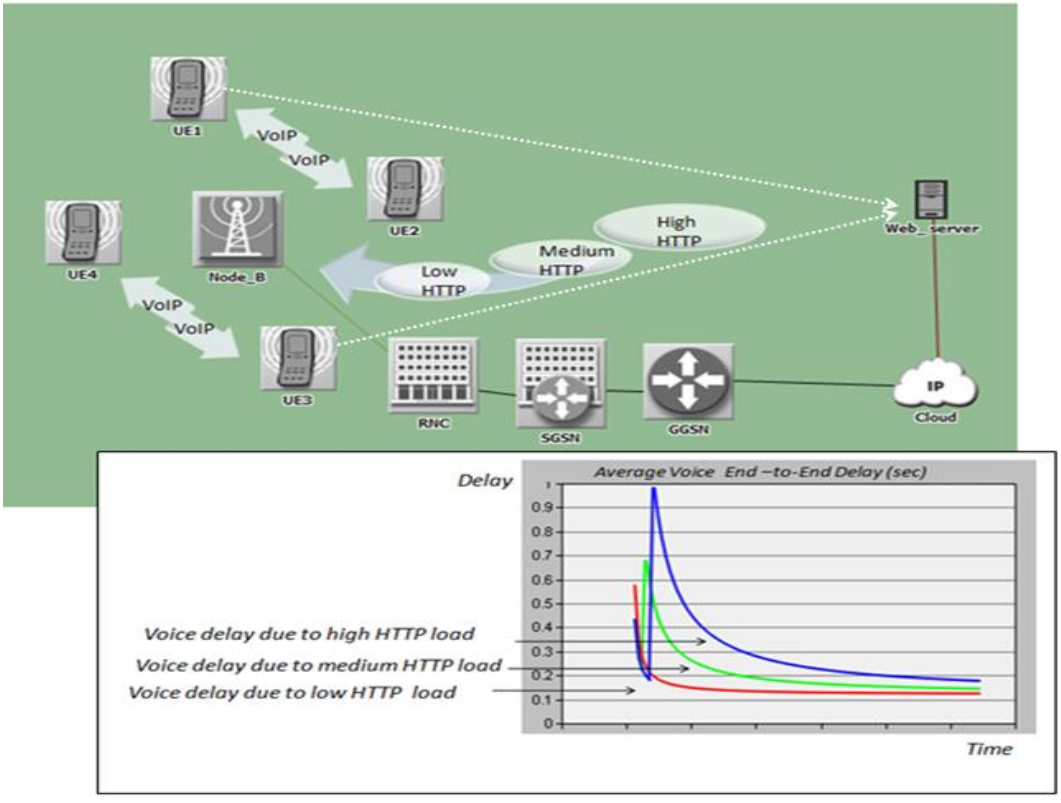



Figure 1.1: UMTS model- explains VoIP congested with Three HTTP networks

For example, Figure 1.1 and Figure 1.2 explain voice end-to-end delay of simple UMTS and WiMAX models. The UMTS model consists of three cases represent VoIP when congested with low, medium and high HTTP networks. The WiMAX model consists of four cases represent VoIP when congested with low, medium, high and very high HTTP networks.

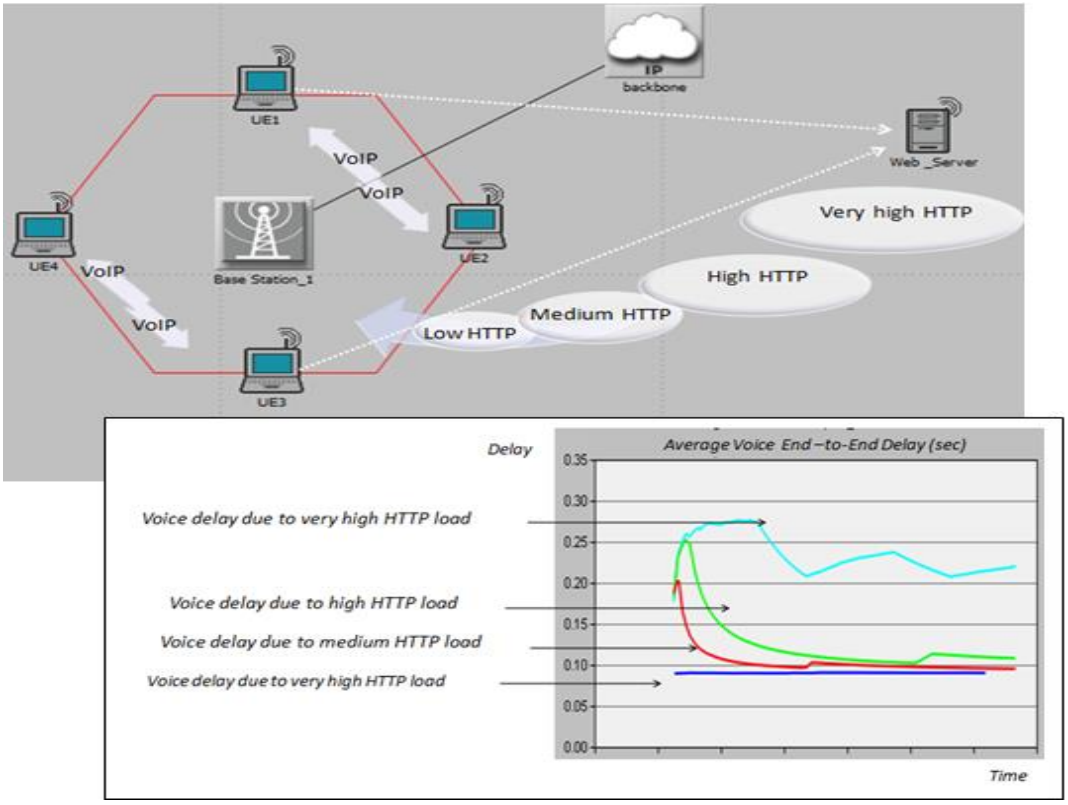



Figure 1.2: WiMAX model- explains VoIP congested with Four HTTP networks

## 1.9 Thesis Outline

The outline of this thesis research is organized as following structure:

Chapter 1 provides the introduction, problem statement, research methodology, aims and objectives, motivation and contribution.

Chapter 2 covers the general overview of the 3GPP UMTS and IEEE 802.16e WiMAX standard and network architecture.

Chapter 3 presents the Quality of Service (QoS) of UMTS and WiMAX networks where Internet protocol (IP), RTP and RTCP Protocols. More-over, Voice over IP (VoIP) and

principle, codec and characteristic, VoIP Protocols, VoIP Performance, VoIP Requirements, Web Browsing Requirements, Integrated Service (IntServ) and IP Differentiated Services Network (DiffServ) are focused in this chapter.

Chapter 4 dedicates a discussion the experiment setup, network scenarios and the parameters required to configure them.

Chapter 5 explains the simulation result followed by chapter 4.Chapter 6 concludes the entire thesis work and future work.

# CHAPTER TWO

## Literature Review and Mobile Evolution and Networks Architecture

## 2.1 Literature Review

Countless researches are on-going in areas of wireless technologies deployment (especially WiMAX and UMTS ) using Voice over IP based network system, all in a bid to come up with a communication system that will be able to provide optimal wireless services so as to meet the increasing user demands. Some related works are explained in this section.

In [6] Dina Mohammed Hassan Al-Zubair and Amen Babikr Abdal-Nabi compared the performance of QoS between UMTS 3G network and WiMAX 4G networks. Two network models were proposed has objective of studying a realistic approach in UMTS and WiMAX networks, Analysis the performance of some metrics such as throughput, jitter, queuing delay and delay. Multiple users are placed in the network requesting some applications such as ftp, web browsing, Email, video conferencing and VoIP. The performance of some parameters as well as meet the goal to analysis the QoS in 3G an 4G wireless networks. From the comparing of two network models and the observed result it is concluded that WIMAX is more efficient than UMTS. WIMAX network has high performance; it has higher throughput and less delay. The queuing delay time in UMTS is longer than WIMAX which caused more jitter in voice and RTP applications.

In [7] Ali Heideri Khoei, Ghasem Mirjally and Mehdi Sarram investigated the efficiency of the main scheduling algorithms as FIFO, WFQ, PQ and MDRR. The performance of each scheduler is evaluated to support the various classes of quality of service and various applications. VoIP traffic with Interactive Voice (6) ToS and UGS servicing class are used in this simulation. For video traffic, Video Conferencing (Light) frames/s and frame size 120*128 pixel, Interactive Multimedia (5) ToS and rtPS servicing class are used. Web Browsing (Heavy) with class Best Effort (0) ToS is used in simulation of this traffic. File Transfer (Heavy) traffic model is used for nrtPS servicing class that is a traffic generation FTP with fixed data packet as 50000 byte and ToS standard (2) class is used. Each mobile station in each cell is put in one of the servicing classes by BS based on its application.

The results showed that PQ and WFQ had the minimum jitter and are the best choice for the users using VoIP. For videoconferencing applications, the PQ scheduling algorithm is the best algorithm due to the highest received traffic and low delay. Finally, MDRR scheduling

algorithm is the best for file sending and web browser in terms of priority of throughput parameter in servicing classes nrtPS and BE

In [8] Mrs.S. Sudarvizhi, Ms. A. Sivasankari,Arulmozhi. B analyzed mobile WiMAX performance for small, medium and large network scenarios under FTP, HTTP, VoIP and Video conferencing traffic with varied node mobility. There is no significant object and page response time changes when using light and medium HTTP traffic transmitted. Download and upload response times are nearly unchanged for light HTTP traffic. Object and page response times for light and medium HTTP traffic have little increase. For heavy HTTP traffic transmitted causes an increase in response time. Packet loss ratios for VoIP and Video Conferencing are irregularly high and increase when the traffic volume of the network also increases.

Sheetal Jadhav, Haibo Zhang and Zhiyi Huang in [9] focused on convergence of different Radio Access Technologies (RATs) providing good Quality of Service (QoS) for applications such as Voice over IP traffic (VoIP) and video streaming. The voice applications over IP networks are growing rapidly due to their increasing popularity and cost. To meet the demand of providing high-quality of VoIP at any time and from anywhere, it is imperative to design suitable QoS model. They have conducted simulation study to evaluate the QoS performance of WiMAX and UMTS for supporting VoIP. They designed simulation modules in OPNET for WiMAX and UMTS, and carried out extensive simulations to evaluate and analyze several important performance metrics such as Mean Opinion Score (MOS), end-to-end delay, jitter and packet delay variation. According to results it was shown that WiMAX outscores the UMTS with a sufficient margin, and is the better technology to support VoIP applications compared with UMTS.

In [10] Al-Imran analyzed and evaluated the performance of existing QoS for voice and video transmission over 3G/UMTS networks. The performance evaluated based on some QoS parameters such as packet loss, delay, throughput and jitter. WFQ scheduling has been used in order to obtain better performance. In this thesis work, two types of applications have been used are voice and video conferencing in several scenarios with QoS and without QoS. From simulation results, have been observed that the packet loss and delay can occur due to several reasons. One reason is large frame size of video conferencing and another is network congestion. From simulation results, can be observe that the packet loss has great impact on

transmission throughput. The throughput is reduced due to increase the packet loss. Jitter is another key issue in terms of voice transmission. The performance of jitter depends on the encoding scheme and number of frames per packet. In the simulation it can be seen that the jitter of voice transmission without QoS is higher than the jitter of voice transmission with QoS. Proper adaption of QoS and adequate bandwidth can provide reliable jitter in voice conference.

## 2.2 Mobile Evolution

With the rapid growth of wireless packet-switched networks, sending data began through the internet rather than the Public Switched Telephone Network (PSTN). Packet-switched networks have better options in terms of reducing cost for both users and service providers. Furthermore, packet-switched networks can support mobile phone users to make voice/video calls through the internet with better communication quality. Becomes increasingly important of using VoIP with the continued proliferation of mobile telecommunication networks such as UMTS, which uses wideband code division multiple access (W-CDMA) radio access technology to offer greater spectral efficiency and bandwidth to mobile network operators [11]. The objective of the Mobile WiMAX standard is to fill the gap between the WLANs (which provide very high data rate with short-range coverage) and the 3G cellular systems (which provide highly mobile long-range coverage with low data rate) by providing a specification that supports a mobile broadband access system which includes functions to enable handoff between base stations (BSs) or sectors. Furthermore, the target is to greatly reduce the cost of WiMAX infrastructure per unit data rate by significantly increasing the system capacity compared to the current 3G standard systems [12].

### 2.2.1 Evolution of Mobile Generations

The cellular wireless Generation (G) generally refers to a change in the nature of the system, speed, technology and frequency. Each generation have some standards, capacities, techniques and new features which differentiate it from the previous one.

### 2.2.1.1 First Generation (1G)

1G network are considered to be the first analog cellular systems, which started early 1980s. 1G network were conceived and designed purely for voice calls with almost no consideration of data services (with the possible exception of built-in modems in some headsets). 1G technology operates with CS only [13].

### 2.2.1.2 Second Generation (2G)

2G mobile system, i.e. GSM (Global System for Mobile Communications). 2G are the first digital cellular systems launched early 1990s, offering improved sound quality. GSM supports circuit-switched data (CSD), allowing users to place dial-up data calls digitally and offers services, such as text messaging (SMS), Multimedia Messaging Service (MMS) and e-mail were a few packet-based services introduced by 2G, a packet delivery with peak data rate of 9.6 Kbps. 2G technology operate with CS only

2.5G networks are the enhancement for 2G version by adding General Packet Radio Service (GPRS) for supporting packet switching traffic in GSM network. No voice channel reservation (no QoS, only best effort service), it has theoretical data rates up to about 144 kbps. These networks have increased the peak data rates to 384 Mbps by introduced Enhanced Data rates for GSM Evolution (EDGE).

#### 2.2.1.3 Third Generation (3G)

Nowadays known as UMTS technology, with this technology, data is transported in packets without the establishment of dedicated circuits. This offers more flexibility and efficiency, and provides better bandwidth and higher speed. 3G supports broadband, packet-based transmission of voice, text, multimedia, and video data at rates from 384 kbps to 2 Mbps [14]. The 3G introduced several Releases has explained below:

**Release 99:** The first Release of 3GPP was Release 99 in early-mid 1999 and published by 3GPP in 2000, development of the new radio access network Universal Terrestrial Radio Access Network (UTRAN) (more detail about UTRAN below). Release 99 was evolution for WCDMA, and introduced Multimedia Message Service (MMS). MMS allows users to send and receive messages exploiting a large array of media types e.g. text of almost unlimited length, images, audio and video clips, enabled peak data rate 2 Mbps in theory, but in practice

gave 384 kbps through the introduction of a 5 MHz UMTS carrier. This is an important milestone for enabling support of end-to-end QoS in PS domain, although QoS is not supported in this Release.

**Release 4:** Release 4 was published in 2001, originally called the Release 2000 - added features including an all-IP Core Network, the peak data rate is 1.28 Mbps.

**Release 5:** Release 5 published in 2002 Introduction of the Internet Multimedia Subsystem (IMS), which supports guaranteed end-to-end QoS in the PS domain. Release 5 which has continued to be developed throughout all following releases. The specification of High Speed Downlink Packet Access (HSDPA.

**Release 6:** Release 6 was completed in March 2004, Introduction of High Speed Packet Uplink (HSUPA), it pushes the peak rates to and 14 Mbps.

**Release 7:** 3GPP Release 7 introduced HSPA in 2007; it supports mobility and handover technique. It reached peak rates to 28 Mbps in the downlink. It Focuses on decreasing latency, improvements to QoS and real-time applications such as VoIP. 3GPP Release 7 introduced HSPA +, it supports data rates of up to 42Mbps. The HSPA + has enhancements by Dual Carrier High-Speed Downlink Packet Access (DC-HSDPA) in combination with the MIMO (Multiple Input Multiple Output) feature, HSPA + could be support peak data rates up to 84 Mbps [15].

#### 2.2.1.4 LTE Networks

Release 8 provided the details for the Long Term Evolution (LTE) in 2008 System, Architecture Evolution (SAE), all-IP network architecture. In LTE, the Evolved Packet Core (EPC) is the latest evolution of the 3GPP core network architecture provides the high capacity, low latency and reached data rate up to 100 Mbps. The advanced Releases of LTE networks which are reached data rate up to 1Gbps. For more detail about advanced Releases of LTE network see (WWW.3GPP.Org ). As mentioned before Figure 2.1 represents the development was introduced to circuit and packet domain.

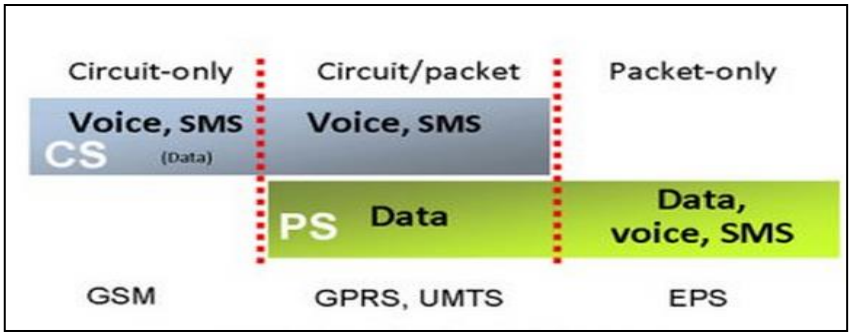



Figure 2.1: Circuits and Packet Domain [14]

### 2.2.2 Evolution of IEEE 802.16 Standards

WiMAX technology is based on the IEEE 802.16 standard of the Institute of Electrical and Electronics Engineers (IEEE). The standard is based on WMAN (Wireless Metropolitan Area Network) [16].

Since then, the 802.16 Working Group have developed several versions, which is now becoming widely implemented globally. Mobile WiMAX was a replacement candidate for cellular phone technologies such as GSM and WCDMA , or can be used as an overlay to increase capacity.

The versions of IEEE 802.16 standard will be described below:

#### 2.2.2.1 IEEE 802.16-2000

This 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.

#### 2.2.2.2 IEEE 8020.16 a2003

IEEE 8020.16a2003 standard enhances the MAC layer to support multiple physical layer specifications. The IEEE 802.16 standard has undergone a number of updates such as, fixed WiMAX standard 802.16d (802.16-2004) uses orthogonal frequency-division multiplexing (OFDM) . It has no mobility and used for fixed and nomadic access.

### 2.2.2.3 IEEE 802.16-2005

The IEEE 802.16-2005 standard is approved in the early 2005, named a Mobile WiMAX. It introduced many new features to enhance QoS and provides high mobility. The Mobile WiMAX air Interface adopts Scalable Orthogonal Frequency Division Multiple Access (SOFDMA) to support channel bandwidths from 1.25 to 20 MHz use of Adaptive Modulation and Coding (AMC). Mobile WiMAX is an attractive wireless technology for VoIP, it offers mobile communication low latency and provides data rates up 75 Mbps, aiming to provide wireless access over long distances in a variety of ways from point-to-point (P2P) and point-to-multi point (P2MP) communication to mobile cellular access as shown in Figure 2.2. For these reason Mobile WiMAX has been taken as field of our thesis in competition with UMTS/HSPA.

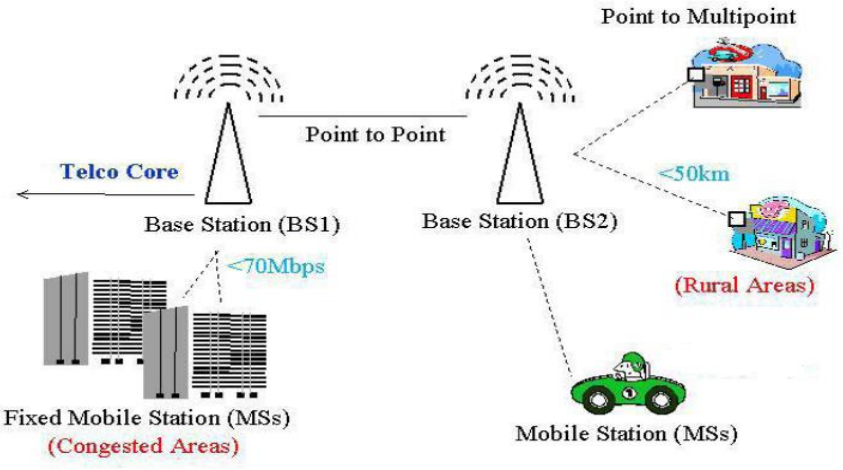



Figure 2.2: P2P and P2MP mobile communication

### 2.2.2.4 IEEE 802.16m

The beyond evolutionary update of the IEEE 802.16 e standard is the IEEE 802.16m (802.16-2011). IEEE 802.16m enables operators with access to multiple channels or licenses to achieve significantly higher peak and average data rates than is achievable with individual channels. Aggregating several 20 MHz channels, for example, could support peak data rates exceeding 1Gbps [17]. Figure 2.3 shows the comparison between 3GPP releases and IEEE 802.16 standards.

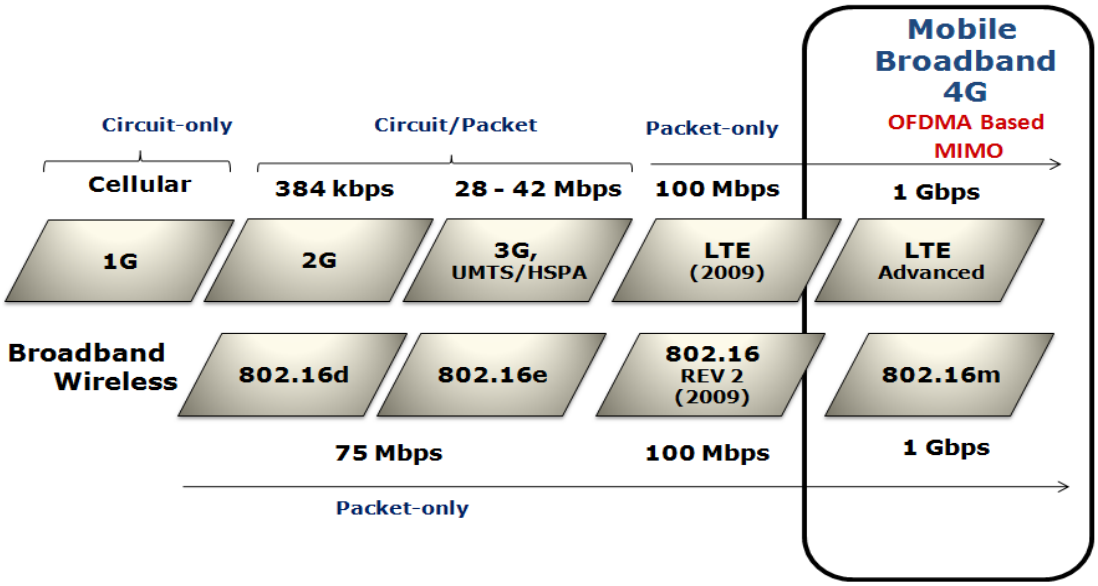



Figure 2.3: Peak Data Rate Evolution for Mobile Generations and IEEE 802.16 Standards

The rest of this chapter is organized as follows. Briefly explaining for system architectures of UMTS and WiMAX networks and giving an overview of ratio access network and physical layer, then explaining radio interface protocols, multiple access, modulation schemes and multi-antenna types for both networks

## 2.3 UMTS Network Architecture and References Model

The UMTS consists of a number of logical network elements that each has a defined functionality. UMTS uses the same well-known architecture that has been used by all 2G systems. Network elements can be grouped based on similar functionalities In addition to the improvements that have been developed to 3G systems.

Figure 2.4 shows UMTS/HSPA architecture with several entities that allows the division of the UMTS into sub-network, User Equipment (UE), UMTS Terrestrial RAN (UTRAN) which consists of Node Bs and Radio Network Controller (RNC).The Core Network(CN) consists of two parts, Circuit Switching (CS) and Packet Switching (PS). The complete explanation about system architecture and its References Model (RM) is found below.

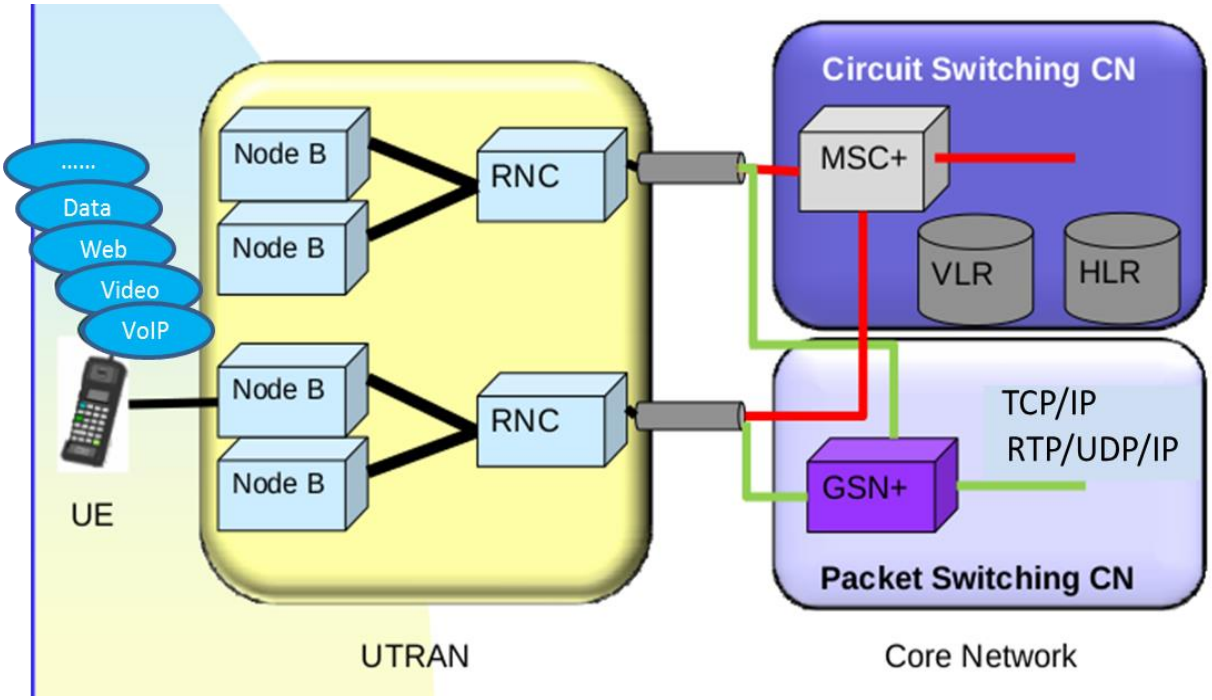



Figure 2.4: UMTS System Architecture

Figure 2.5 shows the elements and interfaces in sub-networks called a Public Land Mobile Network (PLMN). Typically, one PLMN is operated by a single operator, and is connected to other PLMNs as well as to other types of network, such as Integrated Services Digital Network(ISDN), Public Switched Telephone Network (PSTN), Internet, and so on [18].

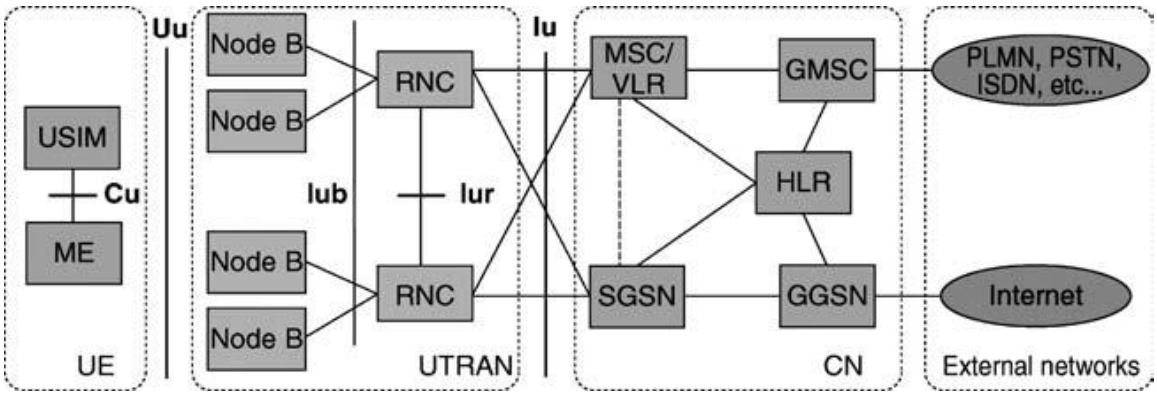



Figure 2.5: UMTS References Model Elements in a PLMN [18]

The UMTS system architecture is presented with a few explanations in the following sections.

### 2.3.1 User Equipment (UE)

UE is responsible for user and network telecommunications, the UE that interfaces with the user where the radio interface is defined UE consists of two parts:

**Mobile Equipment (ME):** ME can be a mobile phone, laptop or a desktop, which can support GSM. It supports the voice call and/or GPRS, hence supporting the Internet based applications.

**UMTS Subscriber Identity Module (USIM):** USIM is a smart card that stores the subscriber identity, authentication information and encryption keys.

### 2.3.2 Universal Terrestrial Radio Access Network (UTRAN)

UTRAN also consists of two distinct elements; the UTRAN is responsible for all functions linked to the radio that handles all radio-related functionality. UTRAN elements as following:

**Node B:** Node-B is connected between RNC and UE. When a call is initiated the Node-B functions convert the data flows from UE to Radio Network Controller (RNC) through (Uu) interface (the interfaces are defined below. It also performs modulation/demodulate-on, transmission/reception. In UMTS /HSPA, the Node B is responsible for the local scheduling instead of RNC, where occurs Release 99 all scheduling. As a consequence, the response of the system is faster and signaling is reduced, especially in the link between Node B and RNC. The moving of scheduling from RNC to Node B implies a change in the overall Radio Resource Management (RRM) architecture.

**Radio Network Controller (RNC):** RNC controls the logical resources of UTRAN access points and controlling cells used by the mobile terminal and allows handover between NodeBs. RNC contains Control RNC (CRNC) unit, which controls Node B (i.e. terminating the Iub interface towards the Node B) is indicated as the Controlling RNC (CRNC) of the Node B. The CRNC is responsible for the load and congestion control of its own cells, and also executes the admission control and code allocation for new radio links to be established in those cells.

### 2.3.3 Core Network (CN)

The core network is the main unit of UMTS network. It controls the operations of the network and it is responsible for switching and routing calls to outward networks. core network is

responsible for establishing, maintaining and terminating the connection with the user and it can be interact with external networks for different types of communication such as data, voice and video etc.

The UMTS core network is combination of Packet Switching (PS) and Circuit Switching (CS). PS domain is adapted from the GSM network and handles the GSM calls. The PS domain handles IP-based calls, while the circuit switching domain handles switching calls.

### 2.3.3.1 Circuit Switched (CS) elements of Core Network

These provide circuit-switched connections, like the existing telephony
service. ISDN and PSTN are examples of CS networks. The circuit switched elements is responsible for performing circuit switching and consists of following elements:

**Home Location Register (HLR):** HLR is a massive database that stores information, such as a user identity. HLR stores the master copy of the user's service profile. The service profile consists of, for example, information on allowed services, forbidden roaming areas.

**Mobile Switching Centre/Visitor Location Register (MSC/VLR):** it is used for moving user from one cell to another; it acts the handover of operation when call is done.

**Gateway MSC (GMSC):** GMSC is a switch for connection to the external network on the CS domain such as PSTN, ISDN.

### 2.3.3.2 Packet switched (PS) elements of Core Network

These provide connections for packet data services. The internet is one example of a PS network. The PS consists of two main components:

**Service GPRS Support Node (SGSN**): The SGSN is one of the important elements to support the packet switched network. It takes care of some important tasks, including routing, handover and IP address assignment. The SGSN converts mobile data into IP and is connected to the Gateway GPRS Support Node (GGSN) via a tunneling protocol. It is also responsible for user authentication, user location update, data encryption and decryption, establishment, maintaining, terminating sessions and mobility management procedure for UMTS users.

**Gateway GPRS Support Node (GGSN):** The GGSN is the main component to support the packet switched network. is basically a gateway of external Packet Data Network (PDN), router and firewall rolled into one. The function of the GGSN is IP address assignment, authentication, IP address mapping and Packet Data Protocol (PDP) context. It has convert packets coming from the SGSN into the appropriate PDP format and then sent to the respective packet network.

### 2.3.4 UMTS Interfaces

Figure 2.6 represents many interfaces that contact between the logical network elements act the reference points define interfaces between different network elements. The following main open interfaces are specified:

- **Uu interface:** Uu is the WCDMA radio interface, which links the UE to Node.
- **Iub interface**: Iub is a link connects a Node B with an RNC.
- **IuCs, luPs interfaces:** luCs , which links the RNC with MSC (CS core), while luPs links RNC with SGSN (PS core).
- **Cu interface,** this is the electrical interface between the UMTS subscriber identification module (USIM) smart card and the mobile equipment (ME). The interface follows a standard format for smart cards
- **Iur interface,** the Iur connects two RNCs with each other. The open Iur interface allows soft handover between RNCs from different manufacturers.

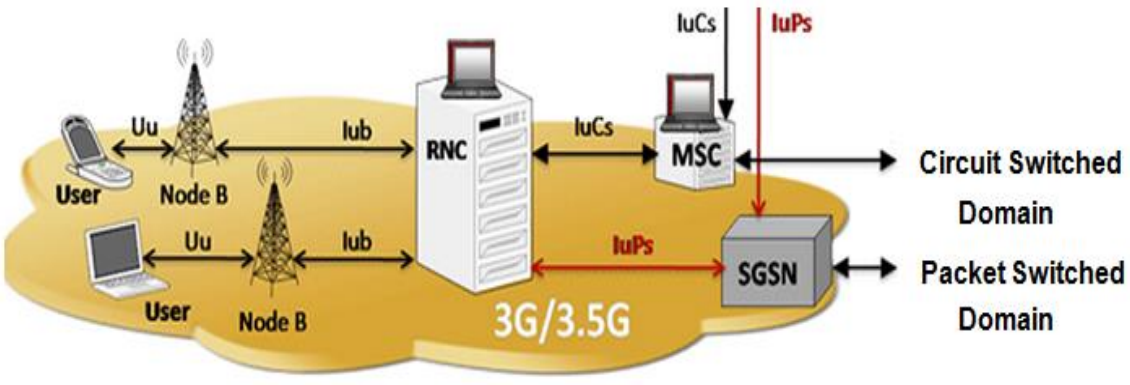



Figure 2.6: UMTS Interfaces [19]

### .2.3.5 UMTS Duplex Technique

The UMTS supports three types of interfaces: duplex modes: Frequency Division Duplexing (FDD), Time Division Duplexing (TDD) air interfaces and Dual mode. Dual mode supports FDD and TDD. In the FDD mode a physical channel is characterized by the frequency code. In the TDD mode the physical channels are characterized by the timeslot. The access scheme is Direct-Sequence Code Division Multiple Access (DS-CDMA) with information either spread over approximately 5 MHz (FDD and 3.84 Mcps TDD) bandwidth, thus also often denoted as Wideband CDMA(WCDMA) due that nature, 10MHz (7.68 Mcps TDD) bandwidth [20]. The interfaces modes also often denoted as a narrow band CDMA. UTRA has two modes, FDD & TDD, for operating with paired and unpaired bands respectively. The possibility to operate in either FDD or TDD mode allows for efficient utilization of the available spectrum according to the frequency allocation in different regions

### 2.3.6 Multiple Accesses

The multiple access schemes is an important feature for mobile communication system. Multiple access allows to differentiate the connections of multiple UEs. In UMTS the main multiple access scheme is a Code Division Multiple Access (CDMA) based technique named Direct-Sequence Code Division Multiple Access (DS-CDMA). DS-CDMA uses orthogonal sequences to spread the transmitted information to the target bandwidth.

. Table 2.1: Summary of features of UMTS evolution

| **Attribute** | **UMTS evolution** |
|---|---|
| Base Standard | WCDMA |
| Duplex Mode | FDD |
| Multiple Access | CDMA |
| Channel Bandwidth (MHz) | 5.0 |
| Frequency (GHz) | 2 |
| Modulation – DL | 16QAM, 64QAM |
| Modulation – UL | QPSK, 16QAM |
| Peak Data Rate (Mbps) | 42.2 (MIMO 2×2, 64QAM) |
| Scheduling | Fast Scheduling |

The UMTS frequency bands [18] are radio frequencies used by third generation (3G). The main UMTS-FDD bands are 1,920-1980 MHz for uplink and 2110–2170MHz for downlink. This spectrum is in International Mobile Telecommunications (IMT-2000) use in Europe. Such use does not preclude the use of these bands by other services to which they are allocated. Table 2.1 summarized the principal attributes of the UMTS/HSPA system. DL and UL peak data rate are referred over the air. The higher value for DL is considering the MIMO (2x2) mode

## 2.4 WiMAX Architecture and Reference Model

The IEEE 802.16e-2005 standard provides the air interface for WiMAX, but does not define the full end-to-end WiMAX network [21]. The WiMAX Forum's Network Working Group (NWG) is responsible for developing the end-to-end network requirements, architecture, and protocols for WiMAX, using IEEE 802.16e-2005 as the air interface. WiMAX network architecture can be divided logically into three parts: Mobile Station (MS), Access service network (ASN) and Connectivity service network (CSN) as shown in Figure 2.7.

The ASN and CSN parts are consist of some components as following:

- WiMAX Base Station.
- ASN-GW (Access Service Network – Gateway).
- CSN-GN (Core Service Network – Gateway).
- AAA proxy or server for user, device and services authentication, authorization, and accounting (AAA).
- The home subscriber server (HSS), or user profile server function (UPSF), is a master
- user database that supports the IMS network entities that actually handle calls.
- MIP-HA( Mobile IP-Home Agent).
- DHCP (dynamic host configuration protocol).
- HSS (Home Subscriber Server).
- ASP: Application Services Providers.

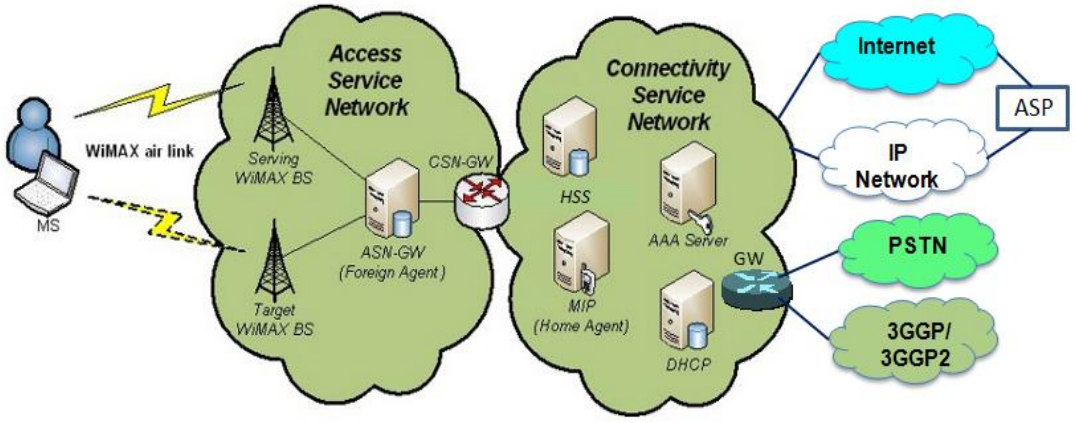



Figure 2.7: IP-Based WiMAX Network Architecture [22]

In such figure the network model developed by the WiMAX Forum NWG defined external networks that the WiMAX network can be connected with.

### 2.4.1 Mobile Station

Mobile Station used by the end user to access the network, MS can be a mobile station device, laptop or PC which uses the network to access the service, MS connects the subscriber to the BS.

### 2.4.2 Access Service Network (ASN)

The ASN has the responsibility of controlling and providing radio access connection to subscribers and can be divided logically into two parts

**Base Station (BS)**: BS is the element that is responsible for physical layer and MAC mechanisms for providing the air interface to the MS, and also contributes to the scheduling for user and exchange signaling messages with the ASN-GW. BS is responsible for radio resource management (RSM), QoS policy enforcement, traffic classification, DHCP (Dynamic Host Control Protocol) proxy, key management and it is responsible for handover control.

**Access Service Network Gateway (ASN-GW):** ASN gateway typically acts as a layer 2 traffic aggregation points within an ASN. Additional functions that may be part of the ASN gateway include intra-ASN location management and paging, radio resource management, and

admission control, caching of subscriber profiles, and encryption keys, AAA client functionality, establishment, and management of mobility tunnel with base stations, QoS and policy enforcement, foreign agent functionality for mobile IP, and routing to the selected CSN

### 2.4.3 Connectivity Service Network

The CSN provides connectivity to the Internet, ASP, other public networks, and corporate networks. The CSN is owned by the NSP and includes AAA servers that support authentication for the devices, users, and specific services. The CSN also provides per user policy management of QoS and security. The CSN is also responsible for IP address management, support for roaming between different NSPs, location management between ASNs, and mobility and roaming between ASNs. Further, CSN can also provide gateway and provides interworking with other networks, such as PSTN (public switched telephone network), 3GPP, and 3GPP2.

### 2.4.4 WiMAX Reference Model

Figure 2.8 shows the WiMAX network reference model (NRM) [22], which is a logical representation of the network architecture. The NRM identifies the functional entities in the architecture and the reference points between the functional entities over which interoperability is achieved. WiMAX network working group (NWG) defines a reference point (RP), as a conceptual link that connects two groups of functions that reside in different functional entities of the ASN, CSN or MS points are not necessarily a physical
interface, except when the functional entities on either side of it are implemented on different physical devices. The RP defined as following:

- ***Reference Point R1***Implements the air-interface (IEEE 802.16e) specifications, to support legacy MS to communicate with the legacy BS.
- ***Reference Point R2***For authentication, authorization, IP host configuration management, and mobility management. Only a logical interface and not a direct protocol interface between MS and CSN.

***Reference Point R3***consists of the set of control protocols between the ASN and the CSN to support AAA, policy enforcement and mobility management capabilities.

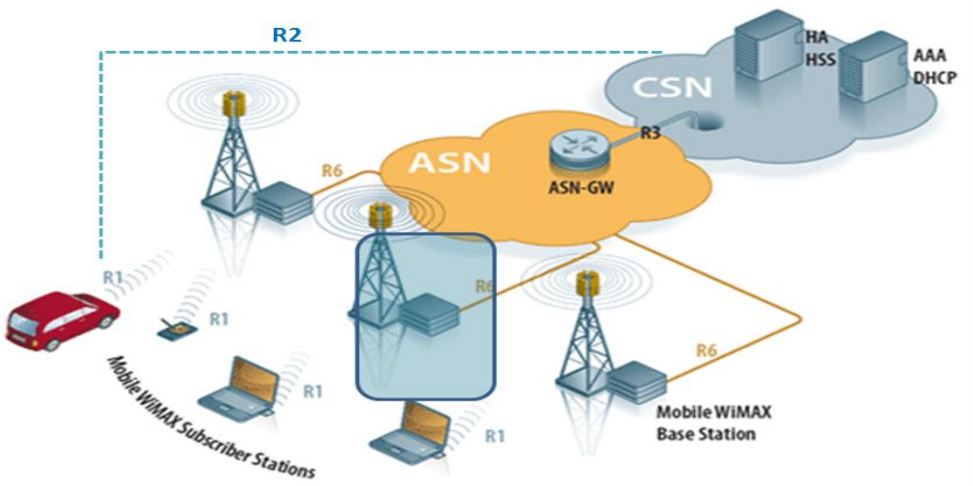



Figure 2.8: WiMAX References Model [23]

- ***Reference Point R4***A set of control and bearer plane protocols originating/terminating in various entities within the ASN that coordinate MS mobility between ASNs
- ***Reference Point R5***A set of control and bearer plane protocols for interworking between the home and visited network.
- ***Reference Point R6***is a set of control and bearer plane protocols for communication between the BS and the ASN-GW.

### 2.4.5 802.16e Duplex Technique

The 802.16e PHY supports TDD and Full and Half-Duplex FDD operation. However, TDD is the most suitable mode of operation, featuring the possibility to adjust the DL/UL ratio in order to adapt to an asymmetric type of traffic [15].

### 2.4.6 Multiple Accesses

In IEEE 802.162004 standard, the OFDM signal is divided into 256 carriers and IEEE 802.16e will use Scalable OFDMA. The using of (SOFDMA) instead of OFDM, to provide more flexibility to deal with a higher variety of antennas and devices to offer better support for QoS. The concept of scalability being means this technique has to scale the size of the Fast Fourier Transform (FFT) to the channel bandwidth with the purpose of keeping the carrier spacing constant across different channel bandwidth. The supported FFT sizes are 128, 512, 1024 and

2048 but only 512 and 1024 are mandatory for Mobile WiMAX profiles, **[24].** OFDMA provides more flexibility to deal with a higher variety of antennas and devices. Sub-channelization allows the operator to manage the available bandwidth and transmit power, which increases the efficiency of the use of resources. OFDMA is an access technique that divides the carrier space.

IEEE Standard 802.16 is an air interface standard for Broadband Wireless Access (BWA). It supports fixed, nomadic and mobile systems, and it enables combined fixed and mobile operation in licensed frequency bands below 6 GHz. In terms of frequency, the frequency used by IEEE 802.16e WiMAX depends on the region or country where the system is implemented. The licensed bands comprise 2.3, 2.5, 3.3 and 3.5 GHz. A license-exempt band can be used for unlicensed WiMAX. In United States of America, the reported WiMAX frequency bands are 2.3, 2.5 and 5.8 GHz, and in Europe the frequency bands are 2.3, 3.5 and 5.8 GHz. Although the WiMAX standards allow operation in any band from 2 to 66 GHz, but the mobile operation is best in the lower bands which are also the most crowded, and therefore most expensive, for this reason in this thesis would apply the range of 5.8 GHz [25].

Table 2.2 summarizes the principal attributes of the WiMAX system. DL and UL peak data rate are referred over the air. The higher value for DL is considering the MIMO (2x2) mode.

Table 2.2: Summary of features and characteristics of UMTS and WiMAX technology [20] [26].

| **Attribute** | **UMTS evolution** | **Mobile WiMAX** |
|---|---|---|
| Base Standard | WCDMA | IEE 802.16e-2005 |
| Duplex Mode | FDD | TDD |
| Multiple Access | CDMA | SOFDMA |
| Channel Bandwidth (MHz) | 5.0 | Scalable: 1.25, 5,10,15, 20 |
| Frequency (GHz) | 2 | 2.5, 3.5, 5.8 |
| Modulation – DL | 16 QAM, 64QAM | QPSK, 16 QAM, 64QAM |

| | | |
|---|---|---|
| Modulation – UL | QPSK, 16QAM | QPSK, 16 QAM, 64QAM |
| Peak Data Rate (Mbps) | 42.2 (MIMO 2×2, 64QAM) | 75 (DL/UL= 1:0, Channel Bandwidth = 20 MHz, MIMO 2×2) |
| Scheduling | Fast Scheduling | |

## 2.5 Requirements for UMTS and WiMAX 802.16e Systems

To gain widespread success due to rapid growth of multimedia application over Internet, the broadband wireless systems must deliver multimegabit per second throughput to end users, with robust QoS to support a variety of services, such as voice, data, and multimedia. Given the remarkable success of the Internet and the large variety of emerging IP-based applications, it is critical that broadband wireless systems be built to support these IP-based applications and services efficiently. Fixed broadband systems must, ideally, deliver these services to indoor locations, using subscriber stations that can be easily self-installed by the end user. Mobile broadband systems must deliver broadband applications to laptops and handheld devices while moving at high speeds. Customers now demand that all this be done without sacrificing quality, reliability, or security [18], [21] and [27].

### 2.5.1 Requirements for UMTS System

Some of the key technical design challenges for UMTS requirements are:

- Bit rates up to 2 Mbps;
- Variable bit rate to offer bandwidth on demand;
- Multiplexing of services with different quality requirements on a single connection, e.g. speech, video and packet data;
- Delay requirements from delay-sensitive real-time traffic to flexible best-effort packet data;
- Coexistence of second - and third-generation systems and inter-system handovers for coverage enhancements and load balancing;
- Support of asymmetric uplink and downlink traffic, e.g. web browsing causes more loading to downlink than to uplink;
- High spectrum efficiency.

### 2.5.2 Requirements for WiMAX System

Some of the key technical design challenges for WiMAX requirements are:

- Developing reliable transmission and reception schemes to push broadband data through a hostile wireless channel
- Achieving high spectral efficiency and coverage in order to deliver broadband services to a large number of users, using limited available spectrum
- Supporting and efficiently multiplexing services with a variety of QoS (throughput, delay, etc.) requirements
- Supporting mobility through seamless handover and roaming
- Achieving low power consumption to support handheld battery-operated devices
- Providing robust security
- Adapting IP-based protocols and architecture for the wireless environment to achieve lower cost and convergence with wired networks

One of the biggest challenges for the current IP based service is QoS, which is ensuring the guaranteed service through internet. For Example, HTTP and video services are bandwidth extensive, which requires less delay to maintain real-time quality. It is not always possible to maintain the quality for all requirements, especially when non real-time traffic flow is sharing voice or video traffic in bandwidth. Different mechanisms, models, schemes and policies are suggested by IETF (Institute European Task Force) to maintain the demand of QoS.

# CHAPTER THREE

## Quality of Service and Voice over IP

## 3.1 Quality of Service

QoS is defined in ITU-T E.800 as the totality of characteristics of a telecommunications service that bear on its ability to satisfy stated and implied needs of the user of the service. QoS also involves controlling and managing network resources by setting priorities for real-time services such as VoIP and video conferencing, etc. Achieving the required QoS by managing the delay, delay variation (jitter), bandwidth, parameters and manage congestion on a network becomes the secret to a successful end-to-end business solution [28].

Much data traffic has been very tolerant of delays and packet drops, however, VoIP traffic has different characteristics, it is low volume but is sensitive to both delay and jitter, plus it has the added complication of being very bursty at times when added other traffic such as Email HTTP and FTB traffic to the network. Under this condition the voice traffic will access the network under congestion state. The result, voice traffic may dropped and delayed, this will intermittent impaired QoS and resulting in customer dissatisfaction.

### 3.1.1 Quality of Service Classification

Classification and marking is an essential component to establishing end-to-end quality of service (QoS), which ensures a VoIP infrastructure. Classification is a system of identifying packets or traffic flows and assigning certain parameters within the packet headers in order to group them. Upon classifying a traffic flow using a particular protocol, a predetermined policy can be applied to it and other flows to either guarantee a certain quality (as with VoIP or media streaming service) or to provide best-effort delivery. This may be applied at the ingress point (the point at which traffic enters the network) [29].

Operators often distinguish three broad types of network traffic: Sensitive, Best-Effort, and Undesired.

- Sensitive traffic is traffic the operator has an expectation to deliver on time. This includes VoIP, online gaming and video conferencing.
- Best effort traffic is all other kinds of non-detrimental traffic. This is traffic isn't sensitive to QoS metrics (jitter, packet loss, latency). A typical example would be peer-to-peer and email
- Peer-to-peer and email and web browser applications.

- Undesired traffic is generally limited to the delivery of spam and traffic created by worms, and other malicious attacks.

### 3.1.1.1 QoS Classes of UMTS Network

UMTS QoS classes were defined in TS 23.107 by taking the restrictions and limitations of the air interface into account. The QoS mechanisms provided in the cellular network have to be robust and capable of providing reasonable QoS resolution. Four classes of services are Conversational, Streaming, Interactive and Background with different QOS parameters and attributes used for prioritization, scheduling and queuing [29].

The UMTS QoS classes:

- Conversational
- Streaming
- Interactive
- Background

#### 3.1.1.1.1 Conversational Class

The most well-known use of this scheme is telephony speech (e.g. GSM). But with Internet and multimedia a number of new applications will require this scheme, for example VoIP and video conferencing tools. Real time conversation is always performed between peers (or groups) of live (human) end-users. Real time conversation scheme is characterized by that the end-to-end delay should be low because of the conversational nature of the scheme and at the same time the time variation between information entities in the stream should be preserved in the same way as for real time streams.

#### 3.1.1.1.2 Streaming Class

When the user is looking at or listening to real time video (audio) the scheme of real time streams applies. The real time data flow is always aiming at a live (human) destination. It is a one way transport. This class tolerates some delay variations that are hidden by the jitter buffer in the receiver.

#### 3.1.1.1.3 Interactive Class

When the end-user, that is either a machine or a human, is on line requesting data from remote equipment (e.g. a server), this scheme applies. Examples of human interaction with the remote equipment are: web browsing, data base retrieval, server access. Interactive traffic is a communication scheme that is characterized by the request response pattern of the end user. Round trip delay is the most important attribute, another important one being the error rate that should be very low in data transfer.

#### 3.1.1.1.4 Background Class

When the end-user, that typically is a computer, sends and receives data-files in the background, this scheme applies. Examples are background delivery of E-mails, SMS, and download of databases and reception of measurement records. Background traffic is one of the classical data communication schemes that on an overall level is characterized by that the destination is not expecting the data within a certain time. The main differences between these QoS classes are how delay sensitive the traffic is. Conversational class is meant for traffic which is very delay sensitive while the Background class is the most delay insensitive class. For further differentiation is detailed in Table 3.1.

Table 3.1: UMTS QoS classes

| QoS Class | Application | QoS Specification |
|---|---|---|
| **Conversational** | voice | Preserve time relation (variation) between information entities of the stream<br>Conversational pattern (stringent and low delay) |
| **Streaming** | streaming video | Preserve time relation (variation)<br>between information entities of the stream |
| **Interactive** | Web browsing | Request response Pattern<br>Preserve payload content |
| **Background** | background<br>download of emails | Destination is not expecting the data within a certain time<br>Preserve payload content |

### 3.1.1.2 QoS Classes of WiMAX Network:

The IEEE 802.16e WiMAX standard offers in [30] five categories for the prioritization of traffic:

Unsolicited Grant Scheme (UGS)

- Real Time Polling Service (rtPS)
- Non Real Time Polling Service (nrtPS)
- Best Effort Service (BE).
- Extended Real Time Polling Service (ertPS)

Each of these has its own QoS parameters such as minimum throughput requirement and delay/jitter constraints.

#### 3.1.1.2.1 Unsolicited Grant service (UGS)

UGS is designed to support real-time uplink service flows that transport fixed-size data packets on a periodic basis, such as and VoIP without silence suppression. The service offers fixed-size grants on a real-time periodic basis,.

#### 3.1.1.2.2 Real-time Polling Service (rtPS)

rtPS is designed to support real-time UL service flows that transport variable-size data packets on a periodic basis, such as video streaming . It allows the subscriber unit to request bandwidth based on the needs of the data stream.

#### 3.1.1.2.3 Extended Real Time Polling Service (ertPS)

ErtPS is a middle ground between UGS and rtPS. It is designed for real-time traffic with variable data rate. However, whereas UGS allocations are fixed in size, ertPS allocations are dynamic.

#### 3.1.1.2.4 Non Real-Time Polling Service (nrtPS)

The nrtPS class provides variable size data grants for non-real-time services such as high bandwidth FTP. This service ensures that flows receive attention even during periods of network congestion.

### 3.1.1.2.5 Best effort (BE) Service

BE is intended to support data streams that don't require minimum guaranteed rate, and could be handled on best available unicast polling requests are not guaranteed in this case, requiring contention requests to be used. HTTP is one protocol that will often be defined as Best Effort. Table 3.2 lists the scheduling service types that can be used for some of the QoS WiMAX standard applications.

Table 3.2: WiMAX QoS classes [22]

| QoS Class | Application | QoS Specification |
|---|---|---|
| UGS | Constant bit rate services such as VoIP | Maximum sustained rate<br>Maximum latency tolerance<br>Jitter tolerance |
| rtPS | Streaming Audio or Video | Maximum reserved rate<br>Maximum sustained rate<br>Maximum latency tolerance |
| ErtPS | Voice with activity detection | Jitter tolerance<br>Maximum latency |
| nrtPS | FTP | Maximum reserved rate<br>Maximum sustained rate<br>Traffic priority |
| BE | Datavtransfer,WebBrowsing | Maximum sustained rate<br>Traffic priority |

Therefore it requires methods and tools to ensure that providers can deliver networks that give the users confidence to use these tools effectively. The problem of providing QoS in broadband wireless systems is one of managing radio resources effectively. Effective scheduling algorithms that balance the QoS requirements of each application and user with the available radio resources need to be developed.

The ability to maintain QoS and manage congestion in a packet-switched network requires the collaboration of many components in the QoS architecture. The main components are as follows:

### 3.1.2 Scheduling for QoS Management

Scheduling of resources such as link bandwidth and available buffers is key to providing performance guarantees to applications that require QoS support from the network. A scheduler is present in both the BS (Base Station and SS (Subscriber sub-System), It determines the order in which packets belonging to deferent flows transmit on the output link and it determines transmission periods for each connection, thus ensuring that packets from deferent applications meet their QoS constraints [31].

#### 3.1.2.1 Scheduling Goals

Following are the goals to be achieved by the new scheduling techniques to support QoS in packet switching networks:

- Sharing bandwidth;
- Providing fairness to competing flows;
- Meeting bandwidth guarantees (minimum and maximum);
- Meeting loss guarantees (multiple levels);
- Meeting delay guarantees (multiple levels);
- Reducing delay variations

#### 3.1.2.2 Scheduling Technique

Scheduling is an important part of warless layers, In order to provide different characteristics to QoS an effective scheduling is critical for the WiMAX and UMTS systems. Both networks include numerous QoS mechanisms at the Media Access Control (MAC) level for guaranteed services for voice, video and data. A number of different scheduling techniques have been used such as first in first out (FIFO), priority queuing (PQ), DWRR or weighted fair queuing (WFQ). More studies show that WFQ technique has a better quality than the other technique, [32], [33].

#### 3.1.2.2.1 First Come First Serve (FCFS)

In case of FCFS queue, the scheduler serving packets from the head of the queue and a newly arrived packet is dropped. FCFS mechanism fails to allocate max-min fair share bandwidth allocation to individual flows. A greedy source can occupy most of the queue and cause delay to other flows using the same Queue, for example VoIP and HTTP flows. There for the policy can be employed when processing sales orders, in determining restaurant seating, on a taxi stand.

#### 3.1.2.2.2 Priority Queuing (PQ)

In case of priority queue, type of service field (ToS) is used from source to request preferential treatment from the network, where multiple queues with different priority levels from 0 to n-1 are maintained. The scheduler serves packets from the higher priority queue before it attends to the lower priority queue. The highest priority has the least delay, highest throughput, and lowest loss. Here packet with lower priority serviced only if the queue is empty. This scheme has the potential of starvation of lower priority classes (i.e., the server will be unable to serve the lower class because it is always busy serving the higher class).

#### 3.1.2.2.3 Deficit Weighted Round Robin (DWRR)

DWRR can be used for the variable length packets. The advantages of these variations of RR are the ability of providing fair queuing with the simplicity of implementation [34]

#### 3.1.2.2.4 Weighted Fair Queuing (WFQ)

WFQ scheduler is used for variable size packets, it provides traffic priority management that automatically sorts among individual traffic streams without requiring that you first define access lists. WFQ can also manage duplex data streams such as those between pairs of applications, and simplex data streams such as voice or video. There are two categories of WFQ sessions: high bandwidth and low bandwidth. Low-bandwidth traffic has effective priority over high-bandwidth traffic, and high-bandwidth traffic shares the transmission service proportionally according to assigned weights. For these reasons we will use WFQ scheme in this research to share HTTP traffic in low priority and high weight with VoIP traffic [34].

### 3.1.3 Call Admission Control (CAC)

The CAC plays a significant role in providing the desired QoS in wireless networks. It is a concept that applies to voice traffic only not data traffic. CAC mechanisms extend the capabilities of the QoS tool suite to protect voice traffic from being negatively affected by other voice traffic, and to keep excess voice traffic off the network [35].

### 3.1.4 Congestion

In data networking and queuing theory, network congestion occurs when a link or node is carrying so much data that its quality of service deteriorates. Typical effects include queuing, packet loss or the blocking of new connections.

#### 3.1.4.1 TCP Congestion Control and Buffer Management

Modern networks such as WiMAX and UMTS use congestion control and congestion avoidance techniques in non-real-time applications such as HTTP and FTB, trying to avoid congestion collapse. These include: exponential backoff in protocols such as fair queuing (FQ) in device as a routers; exponential backoff is an algorithm that uses feedback to multiplicatively decrease the rate of some process, in order to gradually find an acceptable rate. Buffer management has the responsibility of discarding one or more incoming packets before the output buffer overflows. Buffer management is a proactive technique whereby the networking devices monitor their queue length and once it exceeds a certain threshold, they start dropping packets.. One of the most common packet drop strategies is Random Early Detection (RED) [36].

#### 3.1.4.2 Datagram Congestion Control Protocol (DCCP)

Datagram Congestion Control Protocol (DCCP (DCCP) is a message-oriented transport layer protocol. DCCP implements reliable connection setup. But it does not provide reliable in-order delivery. Sequenced delivery within multiple streams as in the Stream Control Transmission Protocol (SCTP) is not available in DCCP. DCCP is useful for applications with timing constraints on the delivery of data. Such applications include streaming media, multiplayer online games and Internet telephony [37]

### 3.1.5 Technical Requirements of QoS Implementation

"There are multiple issues that may well affect the QoS distribution for network based integrated services. These issues result in different traffic types requiring different services levels. Voice relies on UDP/RPT as its transport and UDP is not reliable. Delays in voice traffic create gaps in the transmission that may be heard by the receiver, resulting in unhappy customers. QoS technology features concrete priority service to voice traffic to establish predictable delivery. Usually small in size, transmission of voice packets range from 80 to 256 bytes. Unless QoS techniques are used such voice packets can be delayed between larger data packets such as (HTTP packets)If QoS mechanisms are supported on only portions of the network there are no guarantees that the traffic will get the handling end-to-end that is necessary to achieve success [38]. In order to manage the real-time applications such as VoIP, QoS mechanism should be used to serve needed parameters such as delay and jitter. For that reason the IETF defined two models: Integrated Services (IntServ) and Differentiated Services (DiffServ).

#### 3.1.5.1 Integrated Service (IntServ) Packet Network

Integrated service (IntServ) is responsible for providing services for end users. It is suitable for simple networks, which provides QoS guarantee per flow management of traffic. The major responsibility of IntServ is resource reservation and call setup. The resource with highest priority is served first, while the lowest priority gets low opportunity and the similar priorities are provisioning into queue. IntServ has different kinds of services such as best effort service, Controlled load service and guaranteed service, but the main drawback of IntServ is not scalable to work in large network .

#### 3.1.5.2 IP Differentiated Services Network

Differentiated Services (DiffServ) is a computer networking architecture and designed for large networks Differentiated Services or DiffServ is a QoS mechanism that differs greatly to the IntServ QoS mechanism. IntServ and RSVP are alike in the fact that they both have an ambitious approach to QoS, using an individual flow-based reservation model. DiffServ has a much simpler approach towards QoS, making it more scalable than other QoS mechanisms. DiffServ permits network traffic to be broken up into small flows for appropriate marking. The

flows are marked so that the devices on the network amongst the traffic path will be able to identify and provide the appropriate treatment. Flows and combining them into macro-flows and shaping and dropping traffic so it corresponds with the traffic profile. DiffServ specifies a simple, scalable and coarse-grained mechanism for classifying and managing network traffic to provide end-to-end QoS on modern IP networks. [31], [39].

DiffServ Code Point (DSCP) and Per-Hop Behavior (PHB) are the main components have been used to classify different classes of service in DiffServ domain.

- **Per-Hop-Behavior**

The DiffServ model does not define any service; it defines certain behaviors a packet may receive at each hop, it defines the policy and priority applied to a packet when traversing a hop (such as a router) in a DiffServ network. The collection of packets that have the same DSCP value (also called a codepoint) in them, and crossing in a particular direction is called a Behavior Aggregate (BA).

PHB identifies how the packet should be treated at each hop PHBs. There have been multiple standardized PHB's created to date. There are currently three PHBs. They are:

a) **Default PHB:** Best effort forwarding

b) **Expedited Forwarding PHB:** The Expedited Forwarding (EF) was defined in RFC-3246. PHB has the characteristics of low delay, low loss and low jitter. These characteristics are suitable for voice, video and other real time services. EF traffic is often given strict priority queuing above all other traffic classes. Because an overload of EF traffic will cause queuing delays and affect the jitter and delay tolerances within the class, EF traffic is often strictly controlled through admission control, policing and other mechanisms. The EF PHB polices bandwidth when congestion occurs: The EF PHB prevents starvation of other applications or classes that are not using this PHB.

c) **Assured Forwarding PHB:** Assured Forwarding (AF) was defined in RFC-259. Used for guaranteed bandwidth service. Assured forwarding allows the operator to provide assurance of delivery as long as the traffic does not exceed some subscribed rate. Traffic that exceeds the subscription rate faces a higher probability of being dropped if congestion occurs.

For example, traffic can be divided into gold, silver, and bronze classes, with gold being allocated 50 percent of the available link bandwidth, silver 30 percent, and bronze 20 percent.

- **DiffServ Codepoint**

The DiffServ Code Point is defined in RFC 2474 and 2475. It is a method that classifies the way an IP packet is queued while waiting to be forwarded within a router. DSCP is a field in the IP header that allows system to classify traffic. The objective is to provide a facility so that packets marked with specific DSCP receive well-defined performance or forwarding behavior at every hop.

- **Bandwidth Broker**

RFC 2638 defines the entity of the Bandwidth Broker in the framework of DiffServ. A Bandwidth Broker is an agent that has some knowledge of an organization's priorities and policies and allocates bandwidth with respect to those policies. The bandwidth broker is responsible for automating the process of service level-specifications (SLS) negotiation. In addition, it may also perform admission control, resource management, and network management tasks such as configuration of network devices to support the provisioned QoS according to a common set of operational policies specified for the network.

- **DiffServ Router**

Diffserv router needs a series of components such as classifier, meter, marker, shaper, and dropper commonly known as traffic conditioner. Functions of these components are provided in the following:

- *Classifier*: The packet received by the Diffserv router is first classified by a classifier module. The classifier selects packets based on the values of one or more packet header fields.
- *Meter*: A meter is used to compare the incoming flow with the negotiated traffic profile.
- Shaper: A packet may be sent to the shaper module. This module may introduce some delay in order to bring the flow into compliance with its profile

- *Dropper*: A dropper performs a policing function by simply dropping the packets that are out of profile. It is a special instance of a packet shaper with no buffer.

## **3.2** VoIP

VoIP is voice communication over a packet switched network using Internet Protocol (IP), however, voice communications can be achieved on any IP network regardless it is internet, intranets or local area networks (LAN) [40]. In IP network, the users may be connected to the network by the same VoIP service provider, or to separate VoIP service providers using different IP networks. In the latter case some interworking functionalities might be required.

### 3.2.1 VoIP Protocols

There are a number of protocols that may be employed in order to provide for VoIP communication services. This section will be focus on Real-Time Protocol (RTP) and Real-Time Control Protocol (RTCP) for transmitting audio and video packets between communicating computers.

**RTP/ RTCP:** RTP is defined by the IETF in RFC 3550. RTP also addresses issues like packet order and provides mechanisms (via the Real-Time Control Protocol, which is defined in RFC 3551). RTP provides end-to-end network transport functions, which transmits and exchanges packets of real-time services such as audio and video between communicating devices. It is suitable for multicast or unicast network services. RTP does not address resource reservation and does not guarantee quality-of-service for real-time services [41].

The data transport is augmented by a control protocol (RTCP) to allow monitoring of the data delivery in a manner scalable to large multicast networks, and to provide minimal control and identification functionality. RTP and RTCP are designed to independent of the underlying transport and network layers. RTP has important properties of a transport protocol it runs on end systems, it provides demultiplexing. It differs from transport protocols like TCP. RTCP is used to monitor transmission statistics and QoS to help address delay and jitter and aids synchronization of multiple streams. VoIP uses RTP for these reasons; firstly, VoIP doesn't use TCP because it is too heavy time applications, so instead a UDP (datagram) is used. Secondly, UDP has no control over the order in which packets arrive at the destination or how long it takes them to get there. Both of these are very important to overall voice. RTP solves the

problem enabling the receiver to put the packets back into the correct order and does not wait too long for packets that have either lost their way or those took too long to arrive [42].

### 3.2.2 VoIP Principle

VoIP principle means converting the voice signal into digital signal or packet to make it send over the packet network or Internet (with real time concept) instead of circuits switching (PSTN). Before VoIP sent, it must pass through several stages as shown in Figure 3.1. The voice signal initially is sampled, then digitized, and encoded using a given algorithm/coder. Using RTP/UDP/IP the encoded data is packetized and transmitted. At the receiver's side, data is de-packetized and forwarded to a play out buffer, which smoothies out the delay incurred in the network. Finally, the data is decoded and the voice signal is reconstructed [38], [43].

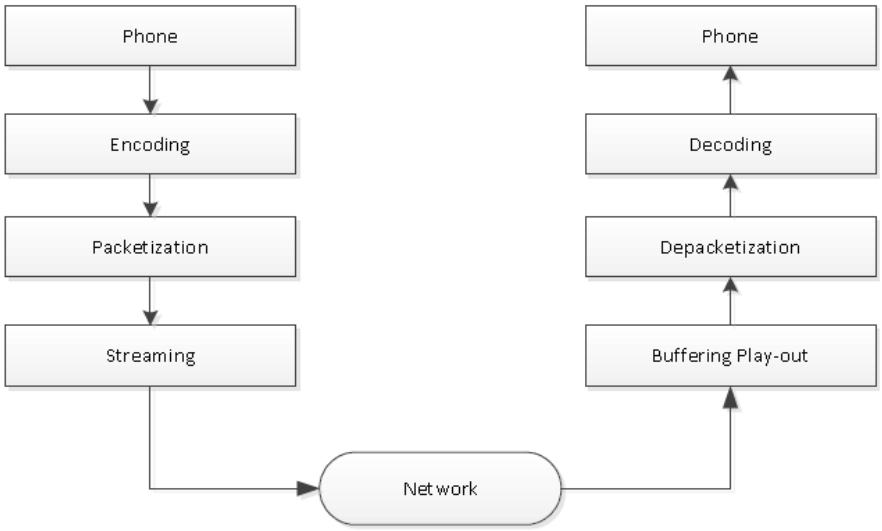



Figure 3.1: VoIP Architecture [43]

### 3.2.3 VoIP Codec

(Coder/Decoder) is a hardware circuit that performs analog-to-digital conversion (ADC) and/or digital-to-analog (DAC) conversion. When a digital device receives analog signals from a microphone or other analog source, the ADC converts them to digital audio samples [44].

Generally, the results are further compressed to save bandwidth. The goal of codec designing is to maintain audio and video quality while compressing the binary data further. There are

various codecs i.e. G.711 and G.729 were provided by ITU-T standard, while GSM-FR and GSM-EFR (Enhanced Full Rate) were provided by ETSI [45]. Basic characteristic of standard codecs are illustrated in Table 3.3. Explains default voice payload sizes of Standard Codecs.

G711 is primarily used in telephony, its formal name is Pulse Code Modulation (PCM) of voice information, this PCM voice information moved over IP network with 64 Kbps every with an encoding frame length of 10 ms and with sample size of 80 bytes. G711 provides numerous inventive features enabling high traffic rates.

G.729 codec technologies apply the Conjugate-Structure Algebraic-Code-Excited-Linear-Prediction (CS-ACELP) speech compression algorithm, approved by ITU-T. It is an 8Kpbs with an encoding frame length of 10 ms and with sample size of 10 bytes. G.729 offers tax quality speech at low bit rate and also allows reasonable transmission delays. It will be perfect for teleconferencing or visual telephony where quality, delay and bandwidth are important.

GSM-FR (6.10) is a Full Rate speech coder standardized by the ETSI and is based on RPE-LTP (Regular Pulse Excitation- Long Term Prediction) speech coding. It was the first digital speech coding standard used in GSM digital mobile phone systems. The coder has a bit rate of 13.2 kbps with an encoding frame length of 20 ms and with sample size of 33 bytes [46].

Table 3.3: Calculations for the default voice payload sizes

| **Codec Information** | | | **Bandwidth Calculations** | | | | |
|---|---|---|---|---|---|---|---|
| **Codec & Bit Rate (Kbps)** | **Codec Sample Size(Bytes)** | **Codec Sample Interval (ms)** | **Voice Payload Size (Bytes)** | **Voice Packet Size (Bytes)** | **Voice Payload Size (ms)** | **Packets Per Second (PPS)** | **Bandwidth (Kbps)** |
| G.711 (64 kbps) | 80 | 10 | 160 | 168 | 20 | 50 | 67.6 |
| G.729 (8 Kbps) | 10 | 10 | 20 | 28 | 20 | 50 | 11.6 |
| GSM-FR 06.10 (13.2 Kbps) | 33 | 20 | 66 | 74 | 20 | 25 | 14.8 |

### 3.2.4 VoIP – Per Call Bandwidth

One of the most important factors to consider when you build packet voice networks is proper capacity planning. Within capacity planning, bandwidth calculation is an important factor to consider when you design and troubleshoot packet voice networks for good voice quality. Table 3.4 explain the calculation of codec bit rate and voice payload size that need to obtain number of packets transmitted every second, which tends to bandwidth need consumed per call [44].

- 40 bytes for IP (20 bytes) / User Datagram Protocol (UDP) (8 bytes) / Real-Time Transport Protocol (RTP) (12 bytes) headers.
- Compressed Real-Time Protocol (cRTP) reduces the IP/UDP/RTP headers to 2 or 4bytes
- 6 bytes for Multilink Point-to-Point Protocol (MP).

Table 3.4: Explanation of Terms for Bandwidth Calculations

| | |
|---|---|
| **Codec Bit Rate (Kbps)** | Codec Bit Rate = codec sample size / codec sample interval<br>For example, the G.711 coder operates on sample intervals of 10 ms, corresponding to 80 bytes (640 bits) per sample at a bit rate of 64Kbps. |
| **Voice Payload Size (Bytes)** | The voice payload size represents the number of bytes (or bits) that are filled into a packet. The voice payload size must be a multiple of the codec sample size. |
| **Packets Per Second (PPS)** | PPS represents the number of packets that need to be transmitted every second in order to deliver the codec bit rate. For example, for a G.711 call with voice payload size per packet of 160 bytes (1820 bits), 50 packets need to be transmitted every second.<br>[50 pps = (6400 Kbps) / (1280 bits per packet) ] |

- **Bandwidth Calculation Formulas**
  - Total packet size = (L2 header: MP) + (IP/UDP/RTP header) + (voice payload size)
  - PPS = (codec bit rate) / (voice payload size)

- Bandwidth = total packet size * PPS

- **Sample Calculation**

For example, the required bandwidth for a G.711 call (64 Kbps codec bit rate) with cRTP, MP and the default 160 bytes of voice payload is:

- Total packet size (bytes) = (MP header of 6 bytes) + ( compressed IP/UDP/RTP header of 2 bytes) + (voice payload of 160 bytes) = 168 bytes
- Total packet size (bits) = (168 bytes) * 8 bits per byte = 1344 bits
- PPS = (64 Kbps codec bit rate) / (1280 bits) = 50 pps

Note: 1280 bits = 160 bytes (default voice payload) * 8 bits per byte

- Bandwidth per call = voice packet size (1344 bits) * 50 pps = 67.2 Kbps

For example, the required bandwidth for a GSM-FR call (13.2 Kbps codec bit rate) with cRTP, MP and the default 66 bytes of voice payload is:

- Total packet size (bytes) = (MP header of 6 bytes) + ( compressed IP/UDP/RTP header of 2 bytes) + (voice payload of 66 bytes) = 74 bytes
- Total packet size (bits) = (74 bytes) * 8 bits per byte = 592 bits
- PPS = (13.2 Kbps codec bit rate) / (528 bits) = 25 pps

Note: 528 bits = 66 bytes (default voice payload) * 8 bits per byte

- Bandwidth per call = voice packet size (592 bits) * 25 pps = 14.8 Kbps

### 3.2.5 VoIP Requirements of QoS

Primary goals of QoS include dedicated bandwidth, controlled jitter and latency or delay (required by some real-time and interactive traffic), and improved loss characteristics. QoS in packet-switched networks has to define a specific set of parameters including delay, jitter, bandwidth and loss or error rate.

#### 3.2.5.1 End-to-End delay

End-to-End delay consists of the delay incurred by the voice signal from the instant it is produced by the speaker until it is heard by the listener at the destination [46]. In VoIP network there are three types of delays occurring during the packet transverse. They are: Sender delays,

network delay and receiver delay. In one direction from sender to receiver for VoIP stream flow.

The end-to-end delay is calculated between two synchronized points A and B of an IP network and it is the time that a packet spends in travelling across the IP network from A to B. The transmitted packets need to be identified at source and destination in order to avoid packet loss or packet reordering.

The end-to-end delay can be calculated by the equation:

$$D\ (ms) = Dpack + \sum h \in path\ (Th + Qh + Ph) + Dplay \qquad (3.2)$$

Where, D is the end-to-end delay and Dpack is the delay due to packetization at the source. Dpack included encoding delay Denc and processing delay Dprocess. Dprocess is including IP phone processing and encapsulation. Dpack is 20 ms and 1ms respectively in G.711 technology. While in GSM-FR is less than 30 ms. Network packet delay corresponds to the time taken for a packet to traverse the network. The typical end-to-end ("mouth-to-ear" in the case of speech) delay ranges from 150 - 400 milliseconds (ms). Delay values up to 200 ms are acceptable even for business purposes. The G.114 ITU-T standard describes that a 150 milliseconds one-way delay is acceptable for high voice quality. In addition,

### 3.2.5.2 Jitter

Jitter is the fluctuation or variation in end-to-end delay from one packet to the next within the same packet flow; it is the variation in the delay of received consecutive packets. At the sending side, packets are sent in a continuous stream with the packets spaced evenly apart. A network with constant latency has no variation (or jitter). Jitter is caused by network congestion, improper queuing or route changes. For reasons the steady stream will become discontinuous [47]. Figure 3.2 illustrates how a steady stream of packets is handled.

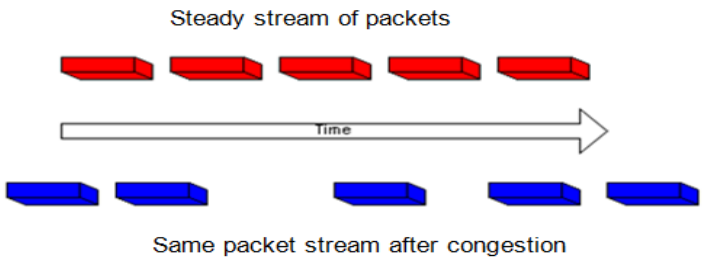



Figure 3.2: Jitter 1

If two consecutive packets leave the source node with time stamps t1 & t2 and are played back at the destination node at time t3 & t4, then:

$$\text{Jitter (ms)} = (t4 - t3) - (t2 - t1)$$

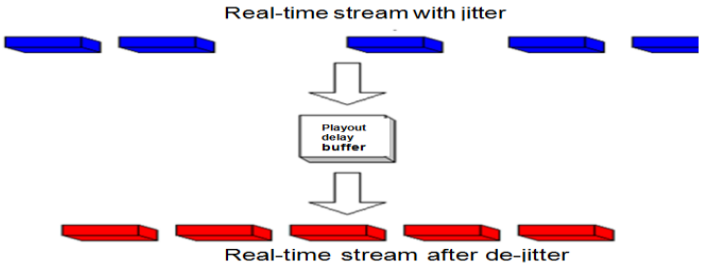



Figure 3.3: Jitter 2

Negative jitter indicates that the time difference between the packets at the destination node was less than that at the source node. When a RTP audio stream is received by a router for VoIP, the router must be compensate for the jitter. The mechanism that handles this function is the playout delay buffer. This operation will buffer these packets and play them out in a steady stream as shown in Figure 3.3, this mechanism sometimes referred to as de-jitter buffer. jitter should be small enough 50 ms at most, so that playback at the receiver remains smooth and a jitter greater than 50 ms is considered poor for VoIP quality [48].

### 3.2.5.3 Packet Loss

Packet loss rate is one of the network QoS parameters. A QoS capable network must provide some guarantees on packet loss rate based on negotiated QoS contract. For VoIP transmission, packet loss rate should not exceed 3% to obtain accepted voice quality.

UMTS and WiMAX networks require QoS to support different type of classes such as conversational voice or video telephony. In this section have look at the network parameters having an impact on VoIP quality.

QoS requirement for audio application for both networks bearer services and their attributes with different range of values are shown in table 3.5.

Table 3.5: WiMAX and UMTS QoS requirement for audio application [49]

| **Medium** | **Application** | **Degree of symmetry** | **Typical data rates** | **Key performance parameters and target values** | | |
|---|---|---|---|---|---|---|
| | | | | **One-way delay(ms)** | **jitter (ms)** | **Information loss** |
| Audio | Conversational | Two-way | 4-64 kbit/s | <150 preferred <400 ms limit | < 1 (jitter) | < 3% packet loss ratio (PLR) |
| Audio | Voice messaging | Primarily one-way | 4-32 kbit/s | < 1 s for playback < 2 s for record | < 1 | < 3% PLR |
| Audio | High quality streaming audio | Primarily one way | 16-128 kbit/s | < 10 s | << 1 ms kbit/s | < 1% PLR |

# CHAPTER FOUR

## Simulations Design and Implementation

## 4.1 Evaluation Platform

This chapter discusses the networks modeling, implementation and their configuration. To implement different scenarios, student version of OPNET Modeler 14.5 has been used [50]. This tool defines user interface and a rich set of modules where users can efficiently create suitable simulation environments.

## 4.2 Networks Model Configuration

The network model has been created to drop network elements to create scenarios, the first type is dragging and drop has used for UMTS scenarios and the second type is startup wizard has used for WiMAX scenarios.

### 4.2.1 Networks Elements and Attributes

Different network elements and attributes have been used to create UMTS and WiMAX networks in order to fulfill the requirements of UMTS and WiMAX scenarios. The networks elements as following:

#### 4.2.1.1 UMTS Network Elements and Attributes

- Umts Workstations nodes: Are used to represent a workstation with source-destination application running over TCP/IP and UDP/IP.
- Umts_sgsn_ethernet_atm_slip9: Includes an ATM, Ethernet, or IP interface to connect to GGSN node. The GPRS Tunneling Protocol (GTP) runs in the IP module on these nodes and sets up GTP tunnels between the SGSN and GGSN.
- Umts_ggsn_slip8, IP32_ cloud_adv: Includes the SGSN module and ATM stacks. The GPRS Tunneling Protocol (GTP) runs in the IP module on these nodes and sets up GTP tunnels between the GGSN and SGSN.
- Ethernet4_slip8_gtwy: Is a router node protocol stack with an IP module and IP interfaces running other layer-2 technologies.
- ATM_OC3 (Optical Carrier 3): Is an ATM- link, with transmission data rate up to 155.52 Mbps.

- PPP_DS3 link: Is used to represent the Ethernet connection operating 44.736 Mbps. This node is connected to two nodes in running IP. Type of this link is duplex. 100BT.

- Application Configuration: Comprises a name and a description table which is specified different parameters for the various applications (i.e. voice, HTTP etc.).
- Profile_Config node: Is used to create user profiles. These user profiles can be precise on various nodes in the network to generate application layer traffic.
- Ethernet_server_adv: Includes UE and client/server application functionality.
- 100BT: Is a baseband transmission. The T designates twisted pair cable duplex link represents an Ethernet connection operating at 100Mbps. It can connect any combination nodes (except Hub-to-Hub, which cannot be connected),
- IP32_cloud_adv: Provides shared processing resources and data to computers and other devices on-demand. The IP cloud supporting up to 32 serial.
- QoS Profile Configuration is used to Specify the expected maximum and guaranteed bit rates for the uplink and downlink communication
- The UMTS network elements mentioned in this section are shown below in Figures 4.3 and 4.5.

#### 4.2.1.2 WiMAX Network Elements and Attributes

- Wimax_Configuration: is a global configuration object used to configure parameters such as service classes and PHY profiles
- Backbone: Is a network is a part of computer network infrastructure that interconnects various pieces of network, providing a path for the exchange of information between different LANs or subnetworks.
- Workstations: Are a mobile subscriber station node model with workstation functionality.
- Wimax_bs_ethernet4_slip4_router: Is fixed base station node model with router functionality. This node has 4 Ethernet interfaces, 4 SLIP interfaces, and one WiMAX interface.
- Wimax_ss_server: Are a mobile subscriber station nodes model with server functionality.

- PPP_adv link: Is a point-to-point advanced duplex link connects two nodes with serial interfaces (e.g., router with PPP ports) at a selectable data rate.
- Application configuration
- Profile configuration
- IP QoS attribute definition
- The Application Configuration, Profile Configuration and IP QoS attributes are defined in previous section. The WiMAX network elements mentioned in this section are shown below in Figures 4.4 and 4.6.

### 4.2.2 Networks traffic Generation

***Application Configuration*:** In order to design simulation model in OPNET, an object is presented called Application Definition attribute consists of several predefined applications i.e. Database, E-mail, FTP, HTTP, Video, Voice, etc. Type of service is used within Application Definition attribute to define applications (HTTP, Email, FTB, Video Conferencing and VoIP). HTTP application is modeled for set up best-effort service in all simulation scenarios. Email is modeled for set up background service, and FTB is modeled for set up excellent-effort service. Since Video Conferencing is modeled for set up interactive multimedia service, while the VoIP application is designed by configuring the (voice table) see right side in Figure 4.1, it is modeled for best- effort service in scenario 1 and is modeled for Interactive Voice service in scenario 2 and 3 as type of service for creating the VoIP calls. VoIP application uses G.711 encoder scheme for WiMAX and GSM FR for UMTS network, for the reasons mentioned in chapter 3.

***Profile Configuration:*** After Application Definition attribute configuration, it is necessary to configure the Profile Definition attribute, where the behavior of the application is set. Figure 4.2 illustrates the Profile Definition attribute has used in our study models. It shows the start time of the simulation, which is set to uniform (100-110) seconds. The simulation time interval is configured at 20 minute. The VoIP application is repeated continuously till the end of last application. It refers VoIP calls that established between source and destination are starting at uniform (5, 10) seconds and the calls are added continuously till the end of last application. The addition of VoIP calls are prepared by repeating the voice application for every 5 second

in the profile definition. This procedure is continuous till the end of simulation. In this approach the VoIP calls are increased continuously at fixed interval to the network model.

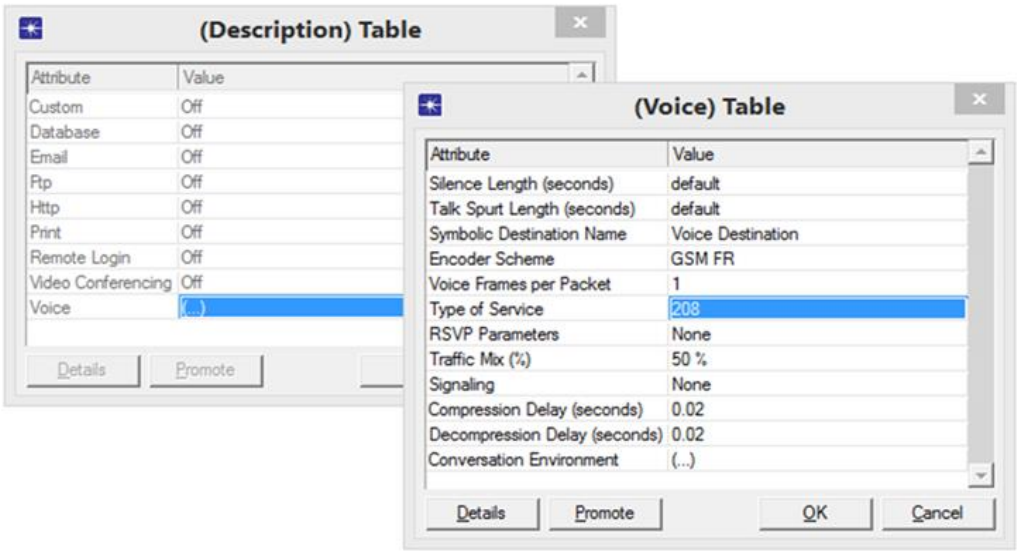



Figure 4.1: Application Definition Configurations

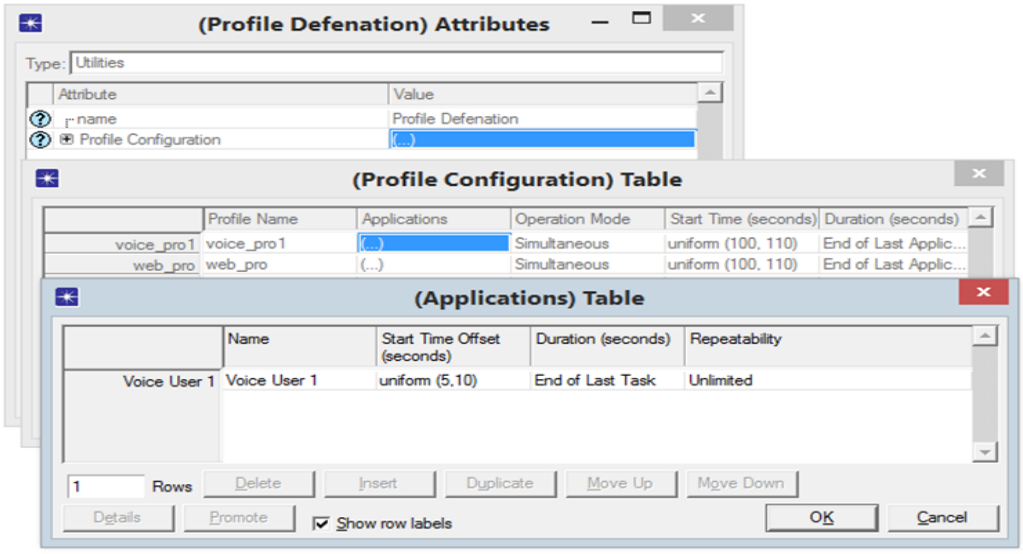



Figure 4.2: Profile Definition Configurations

## 4.3 Simulations General Parameters

Table 4.2 demonstrates the UMTS and WiMAX general parameters have used in the process of simulation models.

Table 4.1: UMTS and WiMAX Parameters

| **Parameter** | **UMTS** | **WiMAX** |
|---|---|---|
| Data Rate | Default | Default |
| Bandwidth (MHz) | 5 MHz | 20 MHz |
| Frequency Band | 1.8 – 2.2 GHz | 5. 8 GHz |
| Cell Radius | 10 km | 10 km |
| Switching Technique | Circuit- and Packet Switched | All – IP Packet |

Table 4.2: Networks Load Parameters

| **Scenario** | **Traffic Type** | **Quality of Service** | | **Traffic Load** | **Packet Size Information** |
|---|---|---|---|---|---|
| | | **UMTS Net.** | **WiMAX Net.** | | |
| One | VoIP | Best-Effort | | 50% | – |
| Two | VoIP | Interactive Voice | | 50% | _ |
| Three Case 1 | VoIP | | | | |
| | HTTP | Best-Effort | | Low | 5000 Bytes |
| Three Case 2 | VoIP | Interactive Voice | | 75% | – |
| | HTTP | Best-Effort | | Medium | 50500 Bytes |
| Three Case 3 | VoIP | Interactive Voice | | 75% | – |
| | HTTP | Best-Effort | | High | 150000 Bytes |
| Three Case 4 | VoIP | _ | Interactive Voice | 75% | – |
| | HTTP | – | Best-Effort | Very High | 750000 Bytes |
| Three Case5 / Case 6 | VoIP | _ | Interactive Voice | 75% | – |
| | HTTP | – | Best-Effort | High | 150000 Bytes |
| | Email | – | Background | High | 150000 Bytes |
| | FTB | _ | Excellent-Effort | High | 150000 Bytes |
| | Video Conferencing | _ | Expedited Forward (EF) | High Resolution Video | 128×240 pixels |

## 4.4 Simulations Design

Design several simulation models in order to test and evaluate the performance of the QoS for VoIP transmission over UMTS and WiMAX networks. Create six scenarios for networks, three scenarios for UMTS and other three scenarios for WiMAX network. For example, the first scenario of each network is designed without QoS to determine the impact on VoIP in case of best-effort class service. The second scenario is designed with QoS scheme as a base line scenario to evaluate the performance of QoS. The third scenario of UMTS consists of three cases. For example, case 1, case 2 and case 3cenario are modeled for congesting VoIP network with (low, medium and high HTTP network) respectively. The third scenario of WiMAX consists of six cases, case 1 to case 4 are modeled for congesting VoIP network with (low, medium, high and very high HTTP network) respectively. While in case 5 scenario, the traffic is mixed between VoIP, high (HTTP, Email and FTB). In case 6 scenario, the mixed traffic introduced VoIP, high (HTTP, Email, FTB and Video Conferencing) to occur more congestion on VoIP network.

### 4.4.1 UMTS and WiMAX Simulations Scenarios

- **UMTS Simulation Scenarios**

1. UMTS Scenario 1: VoIP Network without QoS
2. UMTS Scenario 2: Base Line VoIP Network
3. UMTS Scenario 3: Congested VoIP with Different Traffic Loads:
    - Case 1: Congested VoIP with Low HTTP Traffic Load
    - Case 2: Congested VoIP with Medium HTTP Traffic Load
    - Case 3: Congested VoIP with High HTTP Traffic Load

- **WiMAX Simulation Scenarios**

1. WiMAX Scenario 1: VoIP Network without QoS
2. WiMAX Scenario 2: Base Line VoIP Network
3. WiMAX Scenario 3: Congested VoIP with Different Traffic Loads
    - Case 1: Congested VoIP with Low HTTP Traffic Load

- Case 2: Congested VoIP with Medium HTTP Traffic Load
- Case 3: Congested VoIP with High HTTP Traffic Load
- Case 4: Congested VoIP with Very High HTTP Traffic Load
- Case 5: Congested VoIP with High (HTTP, Email and FTB) Traffic Loads
- Case 6: Congested VoIP with High (HTTP, Email, FTB and Video Conferencing) Traffic Loads

#### 4.4.1.1 VoIP Network without QoS Scenarios

An example for VoIP without QoS model of UMTS network is depicted in Figure 4.3 and an example for VoIP without QoS scenario of WiMAX model is depicted in Figure 4.4. The main objective of this scenario is to see the impact on voice E2E delay when voice traffic with Non-Guaranteed Bit Rate (NGBR). Type of Service (ToS) has been used for voice traffic which operates in environment of best-effort class. The workstations operate in area of one cell radius confined in 10 kilometers.

In UMTS and WiMAX simulation scenarios, the workstations are named UE1 (User Equipment), UE2, UE3 and UE4, where UE1 & UE3 are proceed as source node while UE2 & UE4 are served as destination node. UE1 creates VoIP calling to UE2 while UE3 creates VoIP calling to UE4 at the same time.

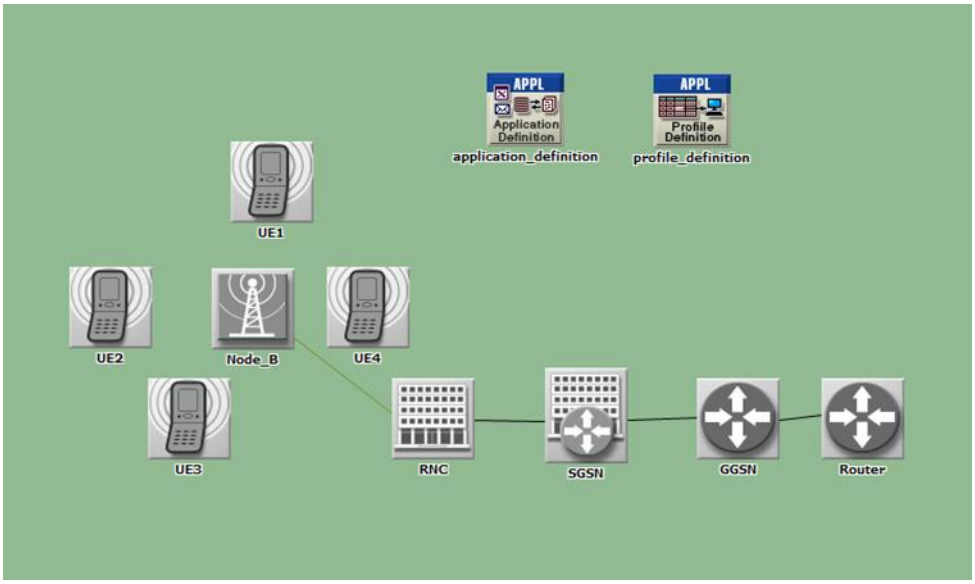



Figure 4.3: VoIP without QoS and Base Line Scenarios of UMTS Network

In all simulation scenarios, the simulation implementation interval is configured to 20 minutes, which the run time of simulation takes 20 minutes to complete processing. First call is generated at 100th second of the simulation because warmup time set as 100 seconds. After that in every 5 seconds a VoIP call is added in the simulation. It is important to mention that VoIP workstations work as stationary nods in all the simulation scenarios.

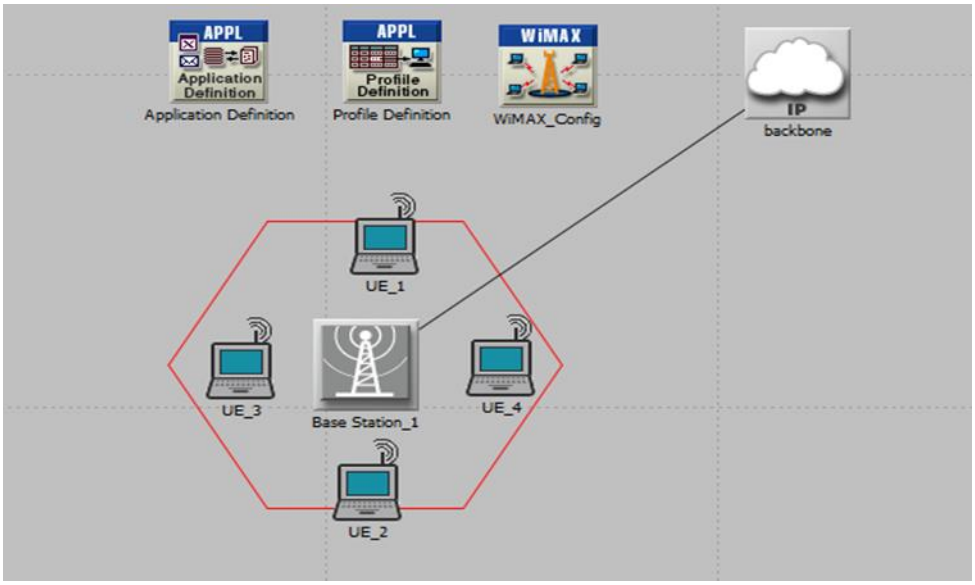



Figure 4.4: VoIP without QoS and Base Line Scenarios of WiMAX Network

### 4.4.1.2 Base Line VoIP Network Scenarios

For Base Line VoIP scenarios of UMTS and WiMAX networks, the same modules of previous scenarios have been used as shown in Figure 4.3 and 4.4. In both scenarios, the Guaranteed Bit Rate (GBR) has been used with Interactive voice sensitive to delay rather than NGBR. The GBR operates in environment conversational service class in UMTS and operates in UGS service class in WiMAX networks. The main objective of Base Line VoIP scenarios is to evaluate the performance of E2E delay of VoIP when used QoS mechanism. To evaluate the performance of the QoS, have been import IP-QoS attribute to implement QoS in both networks. Under the QoS, different types of queuing schemes are available such as FIFO, PQ, DWRR and WFQ. To obtain better performance among these queues schemes WFQ has been used, WFQ is reliable, easy to implement and efficient to define the priority of the traffic in the network for the reason mentioned in chapter three. In order to optimize the performance of QoS in networks, DiffServ mechanism should be used, and in order to enable DiffServ in the

networks, DiffServ Code Point (DSCP) has been used to classify packets. For example in simulation cases, the interactive voice sensitive to delay has used for conversational voice, which configured in (*DSCP* 208).

### 4.4.1.3 Congested VoIP with Different Traffic Loads Scenarios

The main objective of these cases scenarios is to observe the impact on VoIP calls when congested with difference several traffic. In cases (1, 2, 3 and 4) scenarios UE1 phones UE2 and requests, while UE3 phones UE4 and requests HTTP service at 100$^{th}$ second of the simulation. In cases 5 scenario UE1 phones UE2, while UE3 phones UE4 and request (HTTP, Email and FTB) services from all users at same time. In case 6 scenario, have been used the same parameters of case 5 addition to requesting for Video Conferencing service from UE3 and UE4 in 600$^{th}$ second of the simulation.

- **Case1 Scenario: Congested VoIP Network with Low HTTP Traffic Load (5000 bytes)**

The main objective of case 1 scenario is to investigate the optimal transmission of VoIP and HTTP traffic flows in UMTS network; here the HTTP traffic in N-GBR will competes with

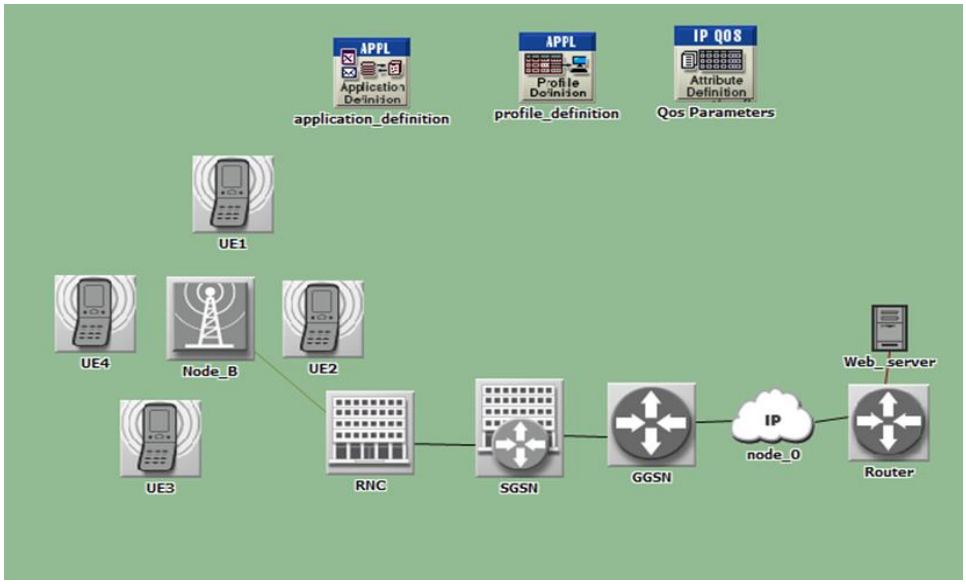



Figure 4.5: Congested VoIP with HTTP Traffic Scenario of UMTS network

VoIP in GBR. Case 1 scenario get started with (5000 bytes) of HTTP traffic as low load network to event low congestion on VoIP network. Figure 4.5 represents the UMTS network model for Congested VoIP with HTTP scenario, while Figure 4.6 represents the WiMAX network model for Congested VoIP with HTTP scenario. In these cases scenarios both UMTS and WiMAX networks use the same parameters for VoIP and HTTP applications as mentioned in table 4.2.

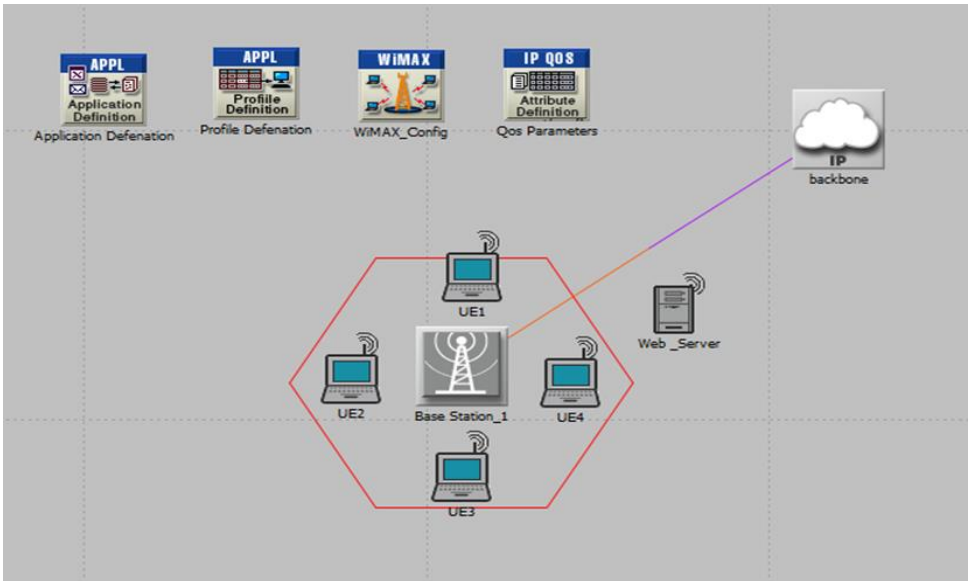



Figure 4.6: Congested VoIP with HTTP Traffic Scenario of WiMAX network

- **Case2 Scenario : Congested VoIP Network with Medium HTTP Traffic Load (50500 bytes)**

The main objective of case 2 scenario is to investigate the optimal transmission of VoIP and HTTP traffic flows in in WiMAX network; In this scenario, the HTTP value is increased to (50500 bytes) as a medium network load to occur more congestion on VoIP networks.

- **Case 3 Scenario: Congested VoIP Network with High HTTP Traffic Load (150000 bytes)**

In case 3 scenario, the HTTP value is increased to (150000 bytes) to occur a high congestion on VoIP networks. Evaluate the performance of packet voice E2E delay and determine to what extend do the performances of packet loss for interactive voice vary from medium to high traffic load within UMTS and WiMAX networks.

- **Case 4 Scenarios : Congested VoIP Network with Very High HTTP Traffic Load (750000 bytes) in WiMAX Network**

This scenario is designed for WiMAX, the frame size is increased to (750000 bytes) as a very high HTTP traffic load. The purpose of this scenario is to determine to what extent the performances of packet loss for interactive voice vary from high to very high HTTP traffic load within WiMAX network.

- **Case 5 Scenario: Congested VoIP Network with High HTTP, Email and FTB Traffic Loads of WiMAX Network.**

One of the key factors of this experiment is that it considers three different network loads corresponded to this case network scenario. Three network loads, high (HTTP, Email and FTB) traffic loads) have been realized and configured for better understanding of how the performance of QoS metrics affected when network loaded with more traffic flows. In case 5 scenario, the users make VoIP calls and send requests for HTTP browser, Email, FTB at 100th second of simulation. The example for case 5 scenario is depicted in Figure 4.7

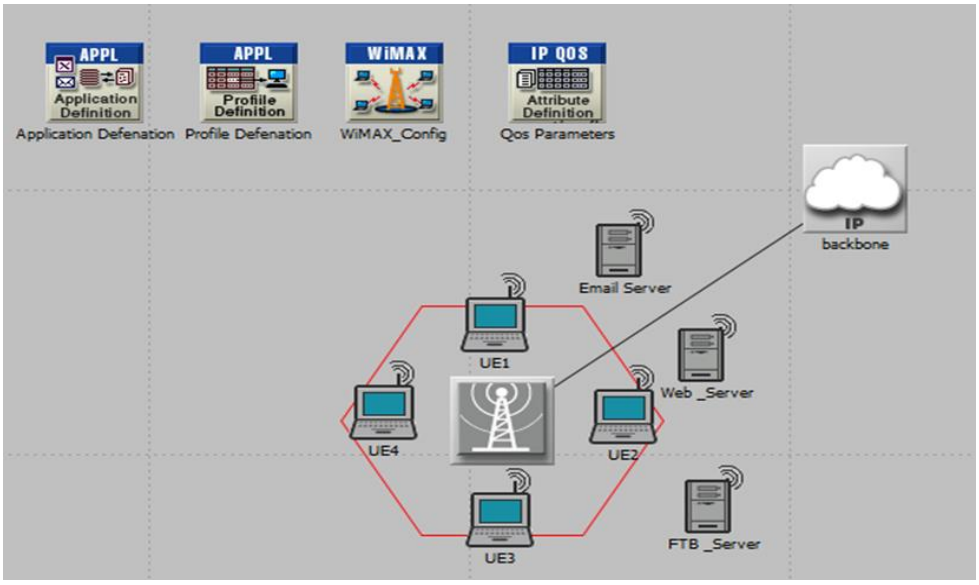



Figure 4.7: Congested VoIP Network with High (HTTP, Email and FTB) Traffic Loads in WiMAX Network

- **Case 6 Scenario: Congested VoIP Network with Very High HTTP, High Email and FTB and Low Video Conferencing Traffic Load of WiMAX Network.**

In case 6 scenario, will be use the same parameters were used in case 5 scenario in addition to request for Video Conferencing from UE2 and UE4 at 600 second of simulation. The purpose of scenario is to find out how WFQ scheme gives different weights and how it deals with different type of services. An example for VoIP congested with HTTP, High Email, FTB and video conferencing traffic load.in WiMAX network is depicted in Figure 4.8.

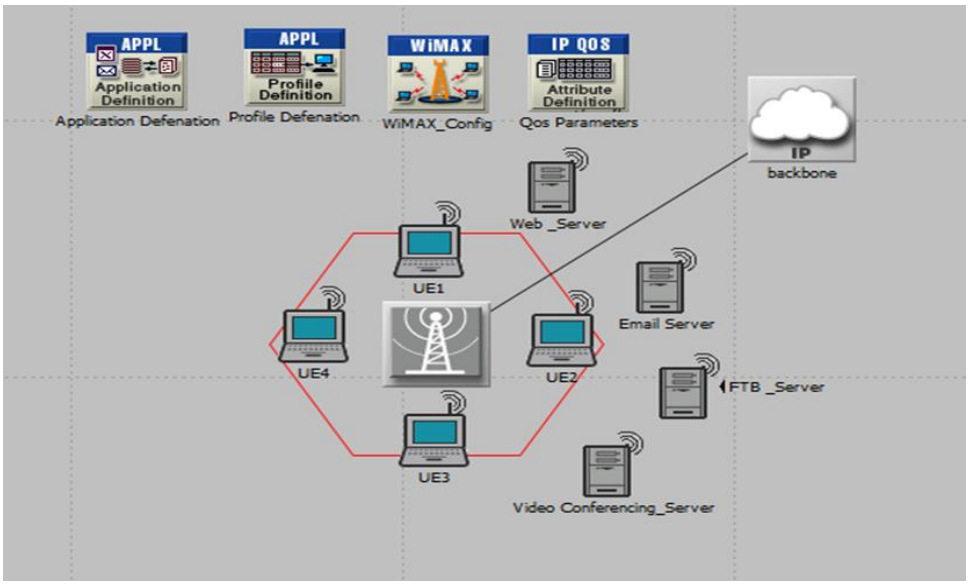



Figure 4.8: Congested VoIP Network with Very High (HTTP, High Email, FTB and Video Conferencing) Traffic Loads of WiMAX Network

## 4.5 Simulation Run-Time

All the simulations run time is set as 1200 seconds. The simulation is implemented in OPNET Modeler 14.5 on DELL laptop with Windows 8.1 Intel core i5, CPU 2.5 GHz and RAM 4.00 GB.

# CHAPTER FIVE

## Simulation Results and Analysis

## 5.1 Performance Metrics

In order to evaluate the performance of voice transmission over UMTS and WiMAX networks, have been consider three QoS parameters such as:

- End to End delay
- Jitter
- Packet Loss

### 5.1.1 End-to-End Delay Performance

In this section, the packet E2E delay results for different simulation scenarios are presented in various statistical plots. As mentioned in chapter Four, Scenario 1, 2 and 3 in UMTS and WiMAX networks, correspond to the VoIP without QoS, VoIP with QoS (Baseline), and VoIP congested with HTTP network scenarios respectively. In all simulation scenarios, the sources and destinations happen to be started between (100-110) seconds of simulation time as the start time of the profile and application configuration, where the time interval for implementing simulation is configured to 20 minute. In all figures presented in following sections, X axis represents simulation time in seconds and Y axis represents End-to-end delay in seconds too. Voice and other applications and profile configuration are discussed in details in Chapter Four. In all following simulation scenarios UE1 phones UE2 & UE3 phones UE4 at $100^{th}$ second of the simulation as mentioned in previous chapter.

#### 5.1.1.1 E2E Delay Performance for VoIP Network without QoS

The parameters have been used for VoIP traffic flow in this scenario of UMTS and WiMAX networks are explained in Table 4.2.The comparable performance of the E2E delay of VoIP network without QoS for UMTS and WiMAX networks is illustrated in Figure 5.1. It represents the average voice packet E2E delay for VoIP calling without QoS; where the blue line shows the E2E delay of UMTS while red line shows the end-to-end delay of WiMAX simulation. From figure, UMTS has E2E delay about 2 seconds on average while WiMAX has end-to-end delay of about 14 seconds on average due to best-effort class service has been used for both applications. Hence all VoIP calls are not accepted due to the packet E2E delay doesn't exceed the threshold value of 400 milliseconds as mentioned in Table 3.5. The reasons behind high end to end delay in VoIP due to voice traffic sent without offering any guaranteed bit rate, where no QoS support for voice traffic.

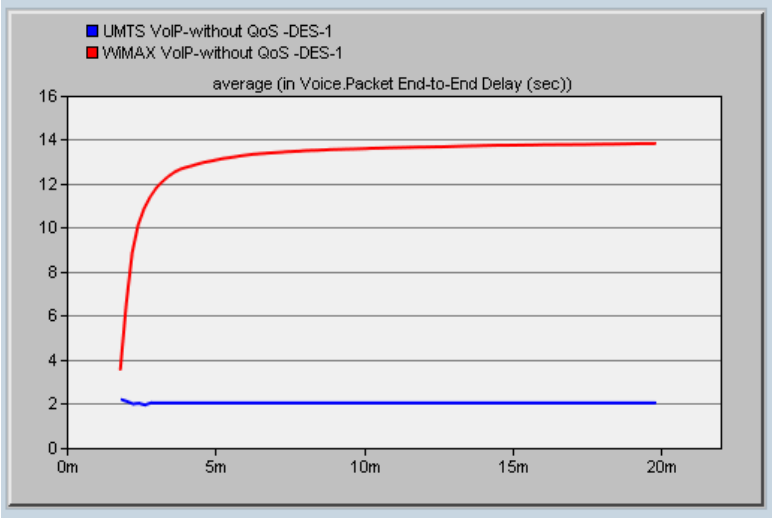



Figure 5.1: Global Statistics Voice Packet E2E Delay for VoIP without QoS of UMTS and WiMAX

### 5.1.1.2 E2E Delay Performance for Baseline VoIP Network

To minimize the end to end delay for VoIP, DiffServ mechanism has been used in order to forward the packet based on PHB to enable the QoS in both networks. From Figure 5.2, the blue line represents the UMTS E2E delay, while red line represents the WiMAX E2E delay.

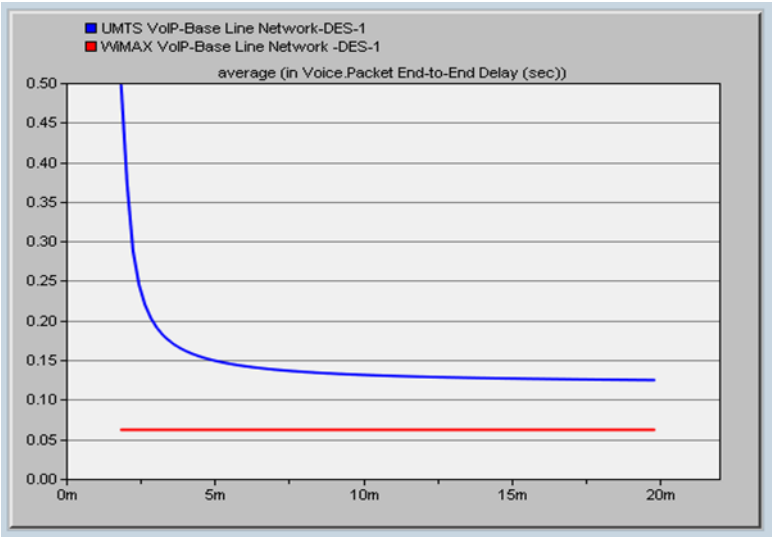



Figure 5.2: Global Statistics Voice Packet E2E Delay for Base Line VoIP of UMTS and WiMAX

UMTS E2E delay is about 120 ms on average, and the E2E delay in WiMAX is about 62 ms on average. From obtained results, it can be deduce, the E2E delay with QoS is lower than the E2E delay without QoS. It is clearly the E2E delay has maintained and minimized by implementing QoS in the networks. The E2E delay is accepted in each of the networks and be suitable for VoIP transmission as mentioned in Table 3.5. From obtained results, it is clearly that QoS mechanism plays a critical role in transfer real-time applications such as VoIP.

- **Objects Voice Packet E2E Delay for UMTS Baseline VoIP Scenario**

Comparison against users of UMTS network in terms of E2E delay, Figure 5.3 and Table 5.1 illustrate the user's voice E2E delay. The blue line represents E2E delay on UE4 with 128 ms and the red line represents E2E delay on UE1 with 120 ms and. The green line represents E2E delay on UE3 with 120 ms too; while the turquoise line represents E2E delay on UE2 with 119 ms, the results on all users are acceptable for due Table 3.5.

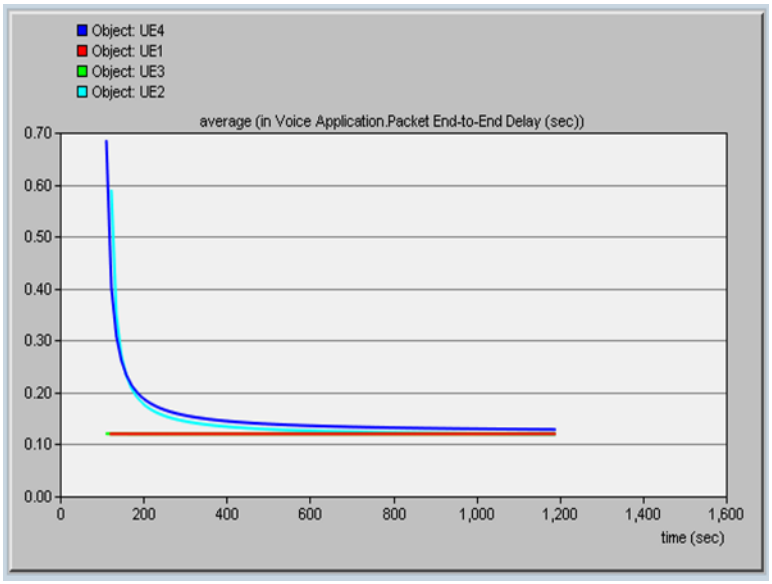



Figure 5.3: Objects Statistics Voice Packet End-to-End Delay for UMTS Baseline VoIP Scenario

Table 5.1 Objects Statistics Voice Packet End-to-End Delay (ms) for UMTS Baseline VoIP Scenario

| Object Name | Minimum | Average | Maximum |
|---|---|---|---|
| UE4 | 122 | 128 | 684 |
| UE1 | 120 | 120 | 120 |
| UE3 | 120 | 120 | 120 |
| UE2 | 120 | 119 | 590 |

- **Objects Voice Packet E2EDelay for WiMAX Baseline VoIP Scenario**

On the other hand, the results of WiMAX users in terms of end-to-end delay is referred to Table 5.2 and observed in Figure 5.4. All users voice E2E delay are equally about 62 ms, the results on all users are acceptable and all users at normal state.

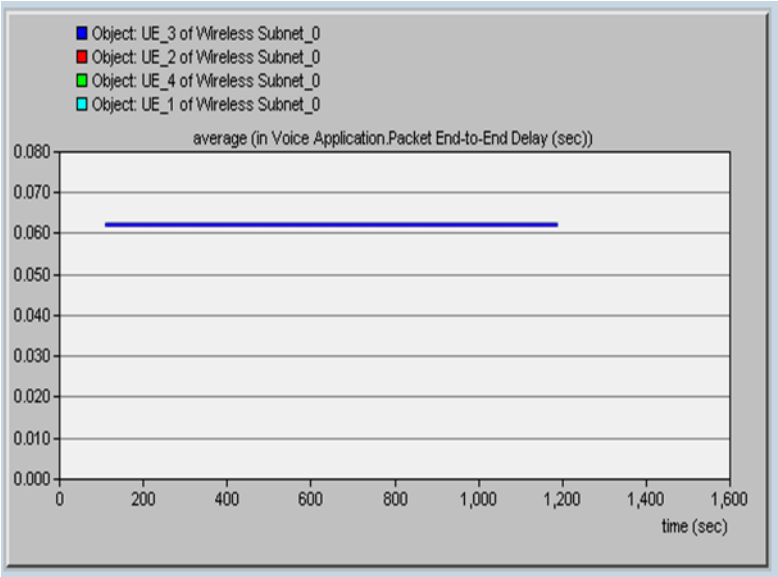



Figure 5.4: Objects Statistics Voice Packet End-to-End Delay for WiMAX Baseline VoIP Scenario

Table 5.2 Objects Statistics Voice Packet End-to-End Delay (ms) for WiMAX Baseline VoIP Scenario

| Object Name | Minimum | Average | Maximum |
|---|---|---|---|
| UE3 | 62 | 62 | 62 |
| UE2 | 62 | 62 | 62 |
| UE4 | 62 | 62 | 62 |
| UE1 | 62 | 62 | 62 |

#### 5.1.1.3 E2E Delay Performance for Congested VoIP Network with Different Traffic Loads

In all following cases scenario, UE1 phones UE2, while UE3 phones UE4, in cases (1, 2, 3 and 4) HTTP browser has been requested from UE1 & UE3 only at 100$^{th}$ second of the simulation as a traffic load on VoIP calls from such users. In cases 5 and 6, through VoIP calls, the four users are request high (HTTP, Email and FTB) at 100$^{th}$ second of the simulation, In case 6 scenario, the same traffic loads have been requested in addition to Video Conferencing has requested from UE2 & UE4 only at 600$^{th}$ second of the simulation. The parameters have used in both networks are explained in Table 4.2.

- **Case1 Scenario: E2E Delay Performance for Congested VoIP with Low HTTP Traffic Load**

In case 1 scenario, VoIP is congested with 5000 bytes of HTTP network, in this simulation should be demonstrate the effect on voice transmission when loaded with 5000 bytes of HTTP network. From explained results in Figure 5.5, the blue line represents the UMTS E2E delay which is approximately 124 ms, while the red line represents WiMAX E2E delay in 90 ms. From observation curves, the E2E delay in UMTS is increased from 120 ms in previous scenario to 124 ms, and in WiMAX is increased from 62 ms to 90 ms due to requested low HTTP browser from UE1 &UE3. There is no significant difference in E2E delay of both

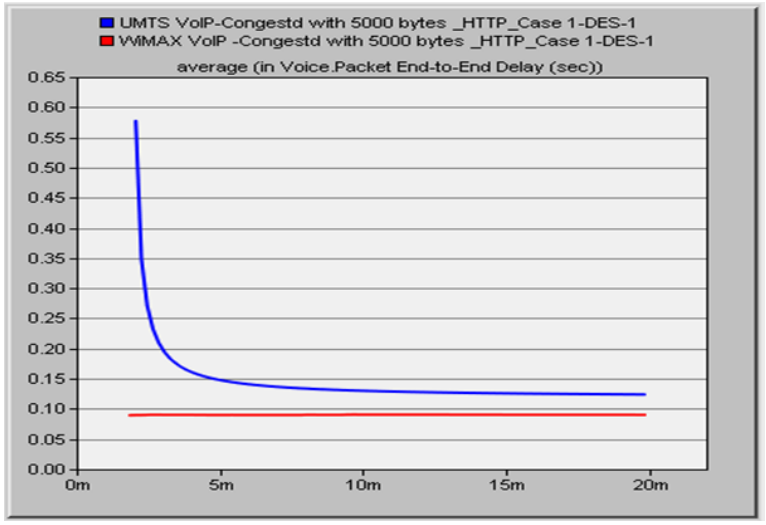



Figure 5.5: Global Statistics Voice Packet End-to-End Delay of UMTS and WiMAX Case 1 Scenario

networks when VoIP loaded with low HTTP network. VoIP calls are accepted due to the packet E2E delay doesn't exceed the threshold value of 400 milliseconds.

- **Objects Voice Packet E2E Delay of UMTS Case 1 Scenario**

From the results occurred in UMTS case 1 scenario, Figure 5.6 and Table 5.3 represent user's voice E2E delay. The blue line represents E2E delay on UE2 with 133 ms and the red line represents E2E delay on UE4 with 121 ms and. The green line represents E2E delay on UE3 with 120 ms too; while the turquoise line represents E2E delay on UE1 with 120 ms, the results on all users are acceptable due Table 3.5.

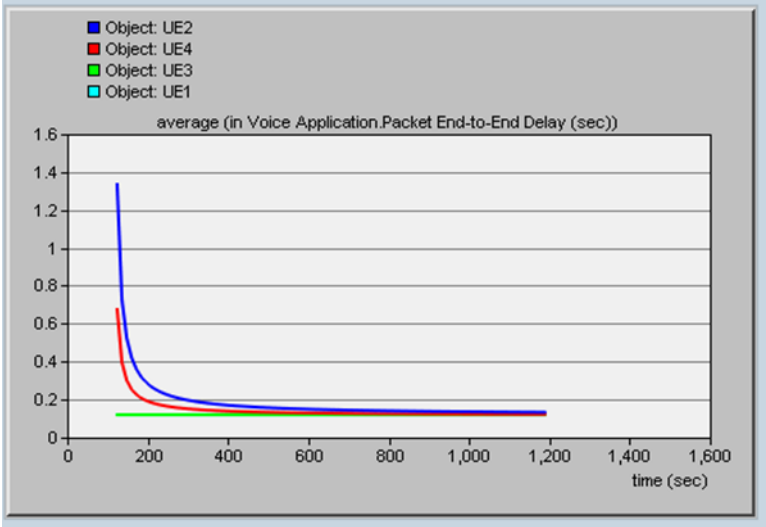



Figure 5.6: Objects Statistics Voice Packet End-to-End Delay of UMTS Case 1 Scenario

Table 5.3 Objects Statistics Voice Packet End-to-End Delay (ms) of UMTS Case 1 Scenario

| Object Name | Minimum | Average | Maximum |
|---|---|---|---|
| UE2 | 119 | 133 | 1346 |
| UE4 | 115 | 121 | 685 |
| UE3 | 120 | 120 | 120 |
| UE1 | 120 | 120 | 120 |

- **Objects Voice Packet E2E Delay of WiMAX Case 1 Scenario**

On other hand on WiMAX case 1 scenario, the results on user's voice had illustrated in Figure 5.7 and Table 5.4.The average E2E delay is ranged between 91 ms on UE4 and 84 ms on UE1. scenario. All these VoIP calls are acceptable due to the packet E2E delay doesn't exceed the threshold value of 400 ms.

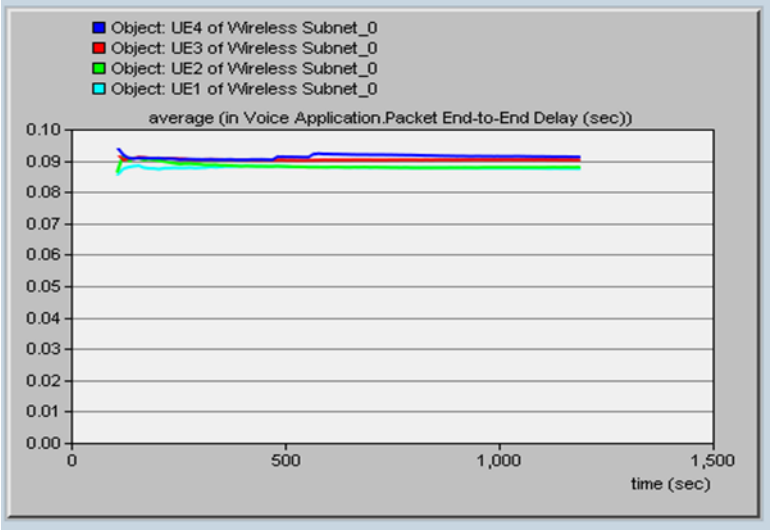



Figure 5.7: Objects Statistics Voice Packet End-to-End Delay of WiMAX Case 1 Scenario

Table 5.4 Objects Statistics Voice Packet End-to-End Delay (ms) of WiMAX Case 1 Scenario

| Object Name | Minimum | Average | Maximum |
|---|---|---|---|
| UE4 | 87 | 91 | 127 |
| UE3 | 87 | 90 | 93 |
| UE2 | 84 | 88 | 96 |
| UE1 | 83 | 84 | 92 |

With respect to HTTP traffic flow in case 1 scenario in both networks, Figure 5.8 represents sent and received HTTP traffic in UMTS case 1 scenario and Figure 5.9 represents sent and received HTTP traffic in WiMAX case 2 scenario. X-axis represents the simulation time in minute and Y-axis represents the sent and received data in bytes per second. Red line indicates the sent data while blue line represents the received data in both figures. Through curve

observation of scenario VoIP congested with 5000 bytes, there is no difference between sent and received HTTP traffic in UMTS and WiMAX case 1 scenario. From observation both graphs, the HTTP traffic did not has any deference between sent and received data. This means both networks can be support HTTP traffic with 5000 bytes.

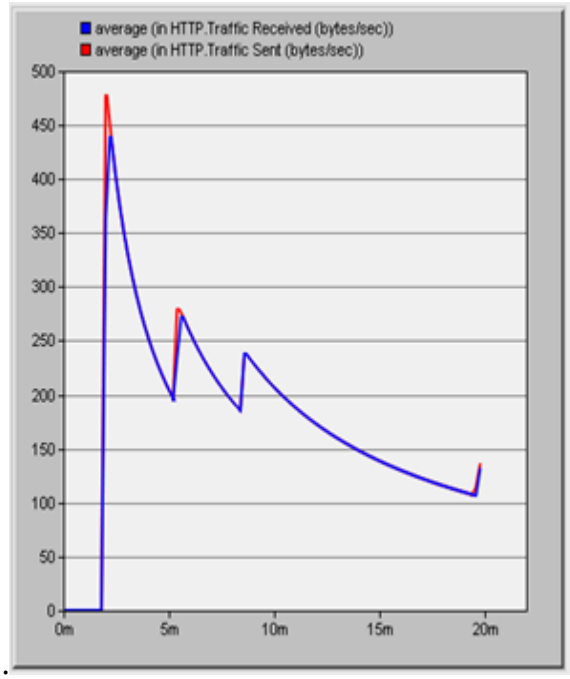



Figure 5.8: Sent and Received Low HTTP Data in UMTS Case 1 Scenario

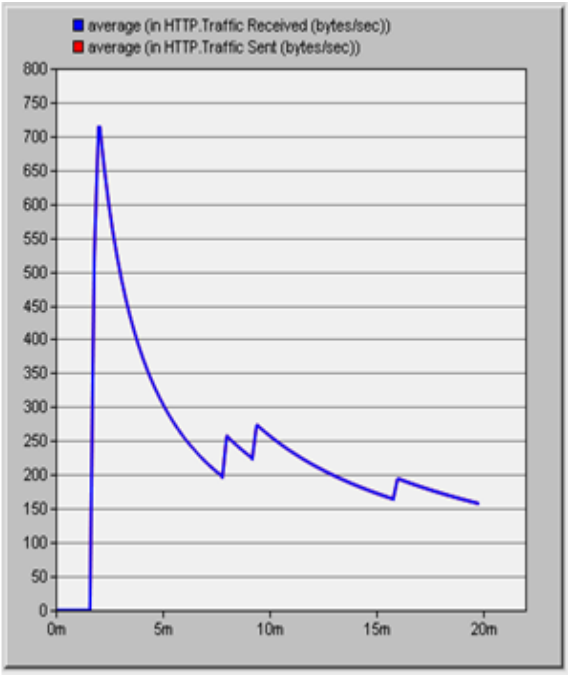



Figure 5.9: Sent and Received Low HTTP Data in WiMAX Case 1 Scenario

- **Case 2 Scenario : E2E Delay Performance for Congested VoIP with Medium HTTP Traffic Load**

Figure 5.10 illustrates the comparable performance of the E2E delay in UMTS and WiMAX under case2 scenario of VoIP congested with medium load of HTTP network. In this network, have been investigate the effect of HTTP traffic on VoIP network when HTTP increased from 5000 bytes to 50500 bytes. Figure 5.10 shows the blue line for E2E delay UMTS case 2 scenario, while red line shows the E2E delay for WiMAX case 2 scenario. As shown in figure the E2E delay in WiMAX is about 100 ms on average and in UMTS is about 150 ms on average. The E2E delay in UMTS case2 is increased comparable with case1, the delay is varied from 124 ms to 150 ms, while E2E delay in WiMAX has a little variation, which

varied from 90 ms to 100 ms. VoIP calls are acceptable due to the packet E2E delay doesn't exceed the threshold value of 400 milliseconds.

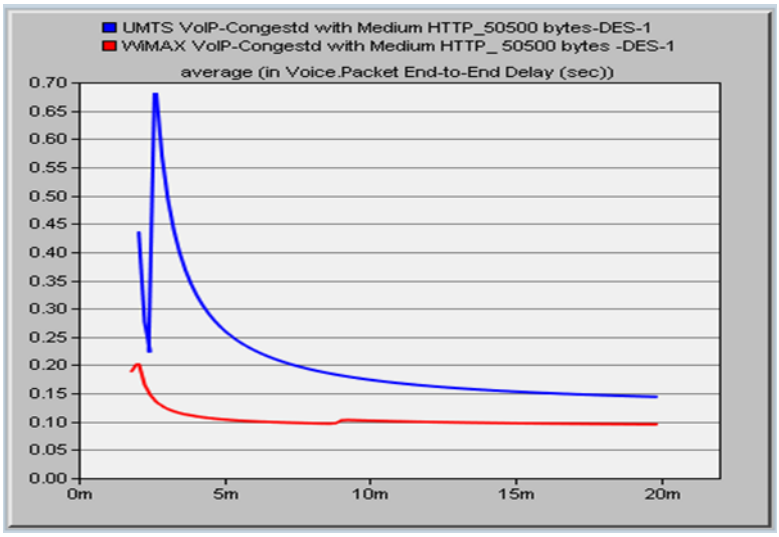



Figure 5.10: Global Statistics Voice Packet E2E Delay of UMTS and WiMAX Case2 Scenario

- **Objects Voice Packet E2EDelay of UMTS Case 2 Scenario**

The E2E delay results of UMTS users in case 2 scenario are shown in Figure 5.11 and Table 5.5, it can be seen that the highest E2E delay is represented with blue line, which is recorded

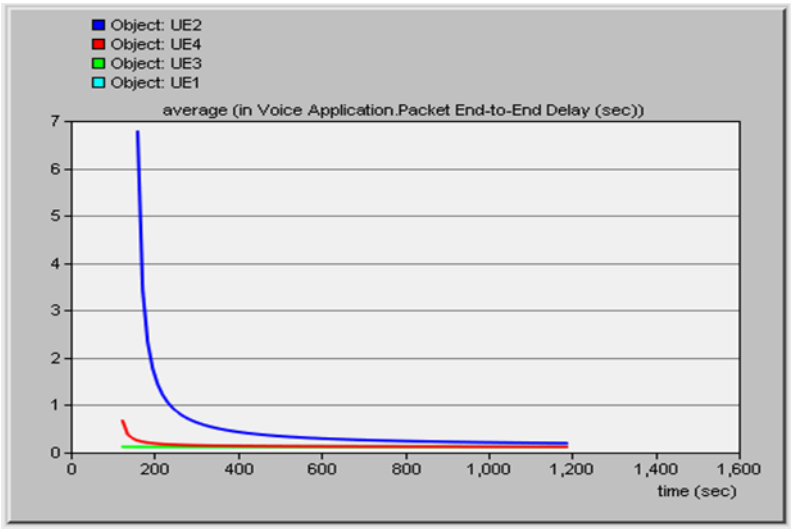



Figure 5.11: Objects Statistics Voice Packet End-to-End Delay of UMTS Case 2 Scenario

on UE2 with 196 ms on average, the ratio is about 32% higher than UE2 in case 1 scenario. The curve referred to UE2 rose suddenly to 678.5 ms, after that is fall down gradually to 119 ms till end of the simulation. The lowest E2E delay is represented with turquoise line, which is recorded on UE2 with 120 ms on average,

Table 5.5 Objects Statistics Voice Packet End-to-End Delay (ms) of UMTS Case 2 Scenario

| Object Name | Minimum | Average | Maximum |
|---|---|---|---|
| UE2 | 119 | 196 | 6785 |
| UE4 | 115 | 121 | 685 |
| UE3 | 120 | 121 | 120 |
| UE1 | 120 | 120 | 120 |

- **Objects Voice Packet E2E Delay of WiMAX Case 2 Scenario**

As shown in Figure 5.12 and Table 5.6, the highest E2E delay is represented with blue line on UE4, which recorded 104 ms on average. The curve referred to UE4 was rose suddenly to 544.ms, after that the curve fall down gradually to 78.17 ms till end of the simulation. The E2E delay is about 12.5% higher than UE4 in case 1 scenario. The lowest E2E delay is represented the with turquoise line on UE1, which recorded 87 ms on average

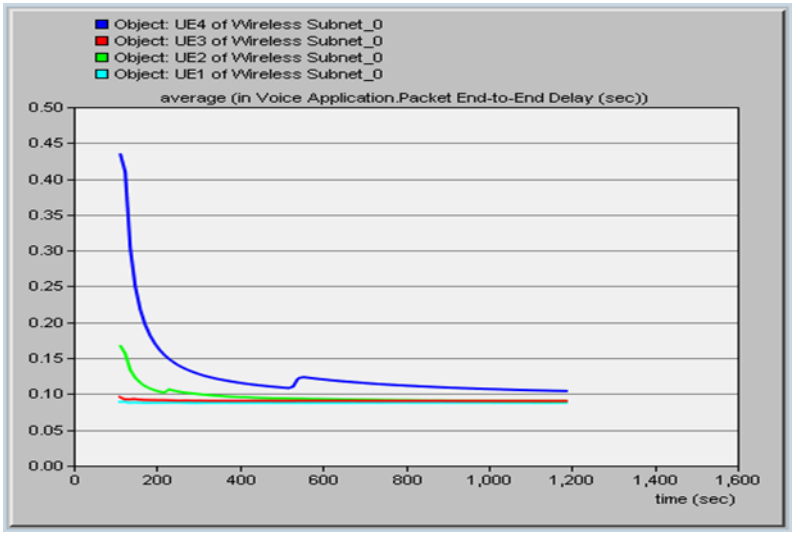



Figure 5.12: Objects Statistics Voice Packet End-to-End Delay of WiMAX Case 2 Scenario

Table 5.6 Objects Statistics Voice Packet End-to-End Delay (ms) of WiMAX Case 2 Scenario

| Object Name | Minimum | Average | Maximum |
|---|---|---|---|
| UE4 | 87 | 104 | 544 |
| UE3 | 87 | 90 | 96 |
| UE2 | 84 | 89 | 168 |
| UE1 | 83 | 87 | 92 |

With respect to HTTP, traffic flow in case 2 scenario in both networks, X-axis represents the simulation time in minute and Y-axis determines the sent and received data in bytes per second. Red line indicates the sent data while blue line represents the received data in both

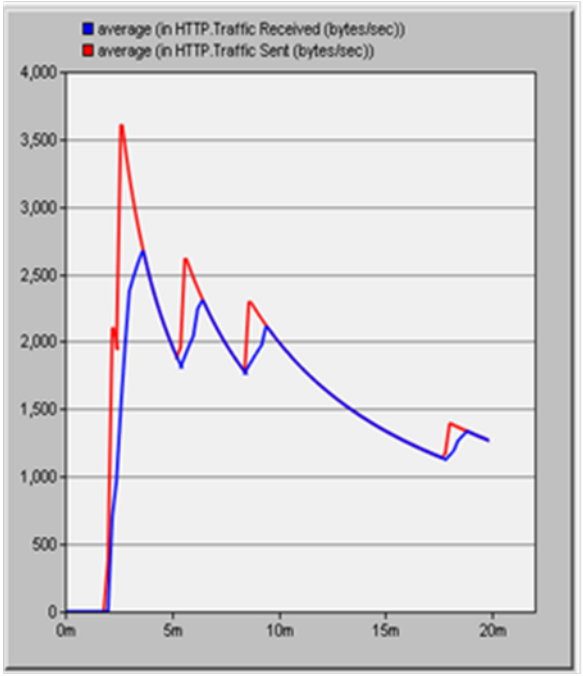



Figure 5.13: Sent and Received Medium HTTP Traffic in UMTS Case 2 Scenario

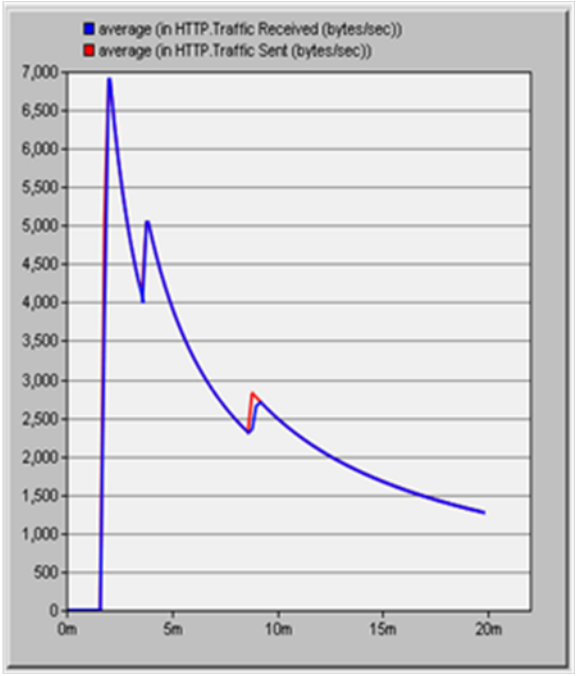



Figure 5.14: Sent and Received Medium HTTP Traffic in WiMAX Case 2 Scenario

from two graphs there is a significant difference between sent and received traffic in UMTS case 2 scenario, this is indicator to presented packet loss in HTTP traffic received in UMTS case 2 scenario, while no difference between sent and received traffic in WiMAX case 2 scenario. Thus, WiMAX case 2 scenario has no packet loss in its HTTP traffic. From the obtained results case 2 scenarios, the UMTS network could not support HTTP traffic with (50500 bytes), but it can be just support HTTP traffic with (5000 bytes). Hence have been

decide the optimal HTTP frame size which UMTS network can be support is (5000 bytes), while WiMAX network is still support HTTP traffic with (50500 bytes).

- **Case 3 Scenario: E2E Delay Performance for Congested VoIP with High HTTP Traffic Load Scenarios**

The purpose of case3 scenario is to increase the traffic congestion on VoIP network by increasing frame size HTTP network from (50500 bytes) to (150000 bytes) and observe packets E2E delay of VoIP network. As shown in Figure 5.15, the blue line represents the result of voice packet E2E delay of UMTS network and the red line represents the result of voice packet E2E delay of WiMAX network. E2E delay in UMTS reached approximately to190 ms on average due to VoIP calling and requested web browser with frame size 150000 bytes.

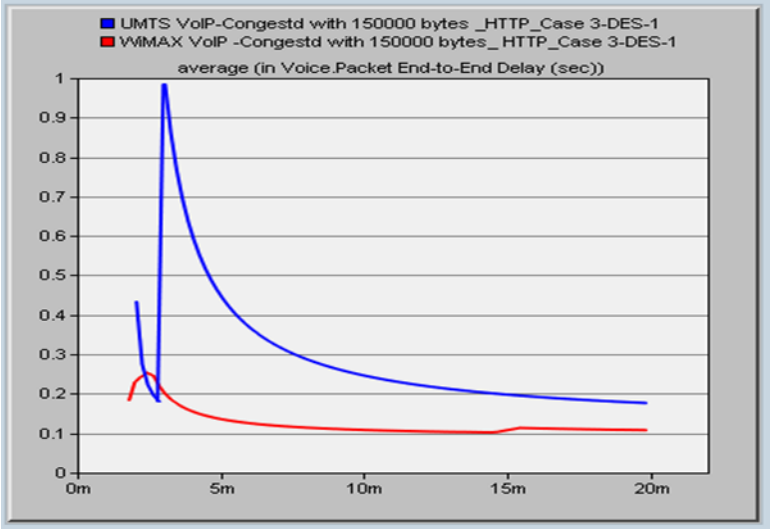



Figure 5.15: Global Statistics Voice Packet E2E Delay of UMTS and WiMAX Case 3 Scenario

On other hand, the voice packet E2E delay of WiMAX network is reached approximately to120 ms on average due to VoIP calling and requested HTTP browser with frame size 150000 bytes. The E2E delay in UMTS is higher than case 2 scenario, the delay is varied from 150 ms to190 ms and in WiMAX is varied from 100 ms to 120 ms. There is significant difference in UMTS E2E delay under high load of HTTP traffic; the E2E delay reached high level.

- **Objects Voice Packet E2E Delay of UMTS Case 3 Scenario**

The E2E delay of users UMTS in case 3 scenario is shown in Figure 5.16 and Table 5.7. The highest E2E delay is represented with blue line on UE2; it is about 386.73 ms on average,. The curve referred to UE2 is rose suddenly to 2283.5 ms, after that fall down gradually to 119.47 ms till end of the simulation, while the E2E delay on other users is still quiet. The delay on UE2 is about 65.67% higher than UE2 in case 1 scenario.

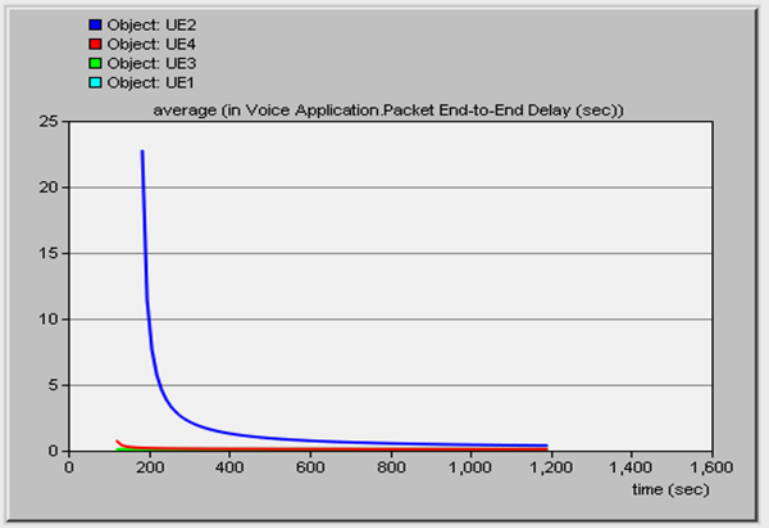



Figure 5.16: Objects Statistics Voice Packet End-to-End Delay of UMTS Case 3 Scenario

By observing the results shown in Figure 5.16 and Table 5.7 for UMTS users, the E2E delay on UE2 is high due to large packet sizes of HTTP traffic. E2E delay on UE2 usually acceptable but may be leads to packet losing.

Table 5.7 Objects Statistics Voice Packet End-to-End Delay (ms) of UMTS Case 3 Scenario

| Object Name | Minimum | Average | Maximum |
|---|---|---|---|
| UE2 | 119 | 386.73 | 2283 |
| UE4 | 115 | 122 | 686 |
| UE3 | 120 | 120 | 120 |
| UE1 | 120 | 120 | 120 |

- **Objects Voice Packet E2E Delay of WiMAX Case 3 Scenario**

For WiMAX users in case 3 scenario, a highest E2E delay is recorded on UE4 as shown in Figure 5.17 and Table 5.8. The curve referred to UE4 is represented with blue line, it is rose suddenly to 579 ms, after that fall down gradually to 87.62 ms till end of the simulation. It can be seen that the average E2E delay is about 133.47 ms, which is about 31.8% higher than case 1 scenario. The E2E delay on other users is still quiet and ranged between 88 and 96 ms.

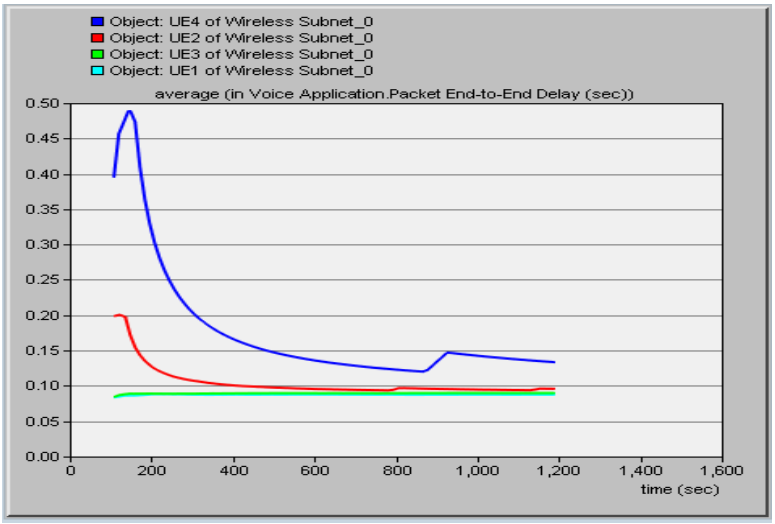



Figure 5.17: Objects Statistics Voice Packet E2E Delay of WiMAX Case 3 Scenario

It can be illustrated that all VoIP calls through WiMAX network are confirmed. From these results can be insure the WiMAX network has more efficient and it can be continue for more HTTP network loaded on VoIP.

Table 5.8 Objects Statistics Voice Packet End-to-End Delay (ms) of WiMAX Case 3 Scenario

| Object Name | Minimum | Average | Maximum |
|---|---|---|---|
| UE4 | 87 | 133 | 579 |
| UE2 | 82 | 96 | 216 |
| UE3 | 85 | 90 | 92 |
| UE1 | 82 | 88 | 93 |

From networks simulation, the UMTS network can't be sustain HTTP traffic with (150000 bytes). For this reason, will end the simulation for loading VoIP with more than (150000 bytes), and will continue with WiMAX network only for loading VoIP. Where the voice End-to-End delay of some users reached high level delays about 386.73 ms and packet loss performance was found abuse 3.63%, which is not acceptable ratio (results related packet loss performance have been explained later).

With respect to web traffic in case 3 scenario, in brief the WiMAX cannot supports HTTP traffic when through VoIP calls. The reason behind the lack of network to support web browser with (150000 bytes) is due to the large frame size of HTTP traffic with best-effort service class, because this class has no guaranteed bit rate to support with big frame size.

- **Case 4 Scenario : E2E Delay Performance for Congested VoIP with Very High HTTP Traffic Load of WiMAX Network**

In order to reach a value of HTTP traffic has effect on VoIP traffic in term E2E delay. Various HTTP frame sizes have been simulated beyond (150000 bytes) until reached (750000 bytes) of HTTP application.

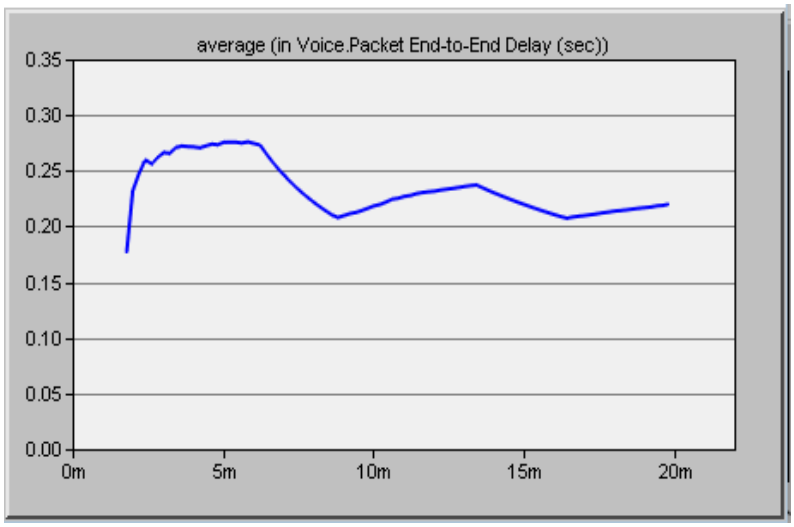



Figure 5.18: Global Statistics Voice Packet E2E Delay of WiMAX Case 4 Scenario

The main purpose of case 4 scenario is to improve to what extent the E2E delay of interactive voice varies when loaded with 750000 bytes of HTTP traffic in WiMAX network. In this

section will be explain in detail the simulation results related to frame size of 750000 bytes and will not explain other simulation results. As shown in Figure 5.18, the voice packet E2E delay reached approximately to 275 ms on average due to VoIP calls from UE1 to UE2 and from UE3 to UE4 at 100$^{th}$ second of the simulation adding to sending requesting for web browser from UE1and UE3 with frame size 750000 bytes in such time. Then E2E delay decreased to 220 ms then rises slowly up to 240 ms and reached 235 ms at the end of simulation. So can conclude the E2E delay varies high in WiMAX network.

- **Objects Statistics Voice Packet E2E Delay of WiMAX Case 4 Scenario**

As shown in Figure 5.19 and Table 5.9. The blue line represents maximum E2E delay on UE4 with 399 ms, red line represents E2E delay of UE2 with 111 ms and green line represents E2E delay of UE3 with 90 ms while turquoise line represents E2E delay of UE1 with 88 ms. It can be seen the maximum E2E delay is recorded on UE4 with 399 ms due to VoIP calling from UE3 to UE4 and requesting web browsing with frame size (750000 bytes) from UE3. The E2E delay on some users is satisfying; but UE4 is dissatisfied because E2E delay does reach the threshold value of 400 ms.

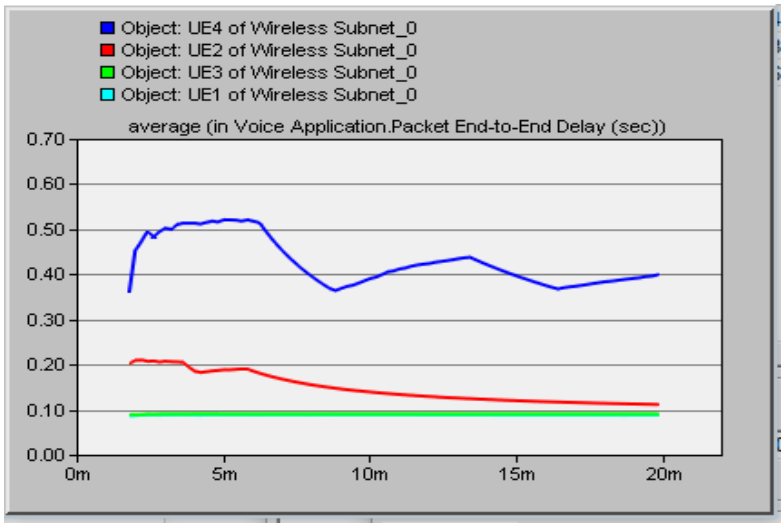



Figure 5.19: Objects Statistics Voice Packet E2E Delay of WiMAX Case 4 Scenario

Table 5.9 Objects Statistics Voice Packet E2E Delay (ms) of WiMAX Case 4 Scenario

| Object Name | Minimum | Average | Maximum |
|---|---|---|---|
| UE4 | 88 | 399 | 669 |
| UE2 | 84 | 111 | 216 |
| UE3 | 87 | 90 | 94 |
| UE1 | 83 | 88 | 92 |

- **Case 5 Scenario : E2E Delay Performance for Congested VoIP with High HTTP, Email, and FTB Traffic Loads of WiMAX Network**

It is important to mention for the sake of better understanding and analyzing how the packet E2E delay is impacted by different network loads. Here many of the applications have been dealt with at the same time. In this simulation voice traffic network has faced many of other traffic flows such as HTTP, Email and FTB. All traffic flows are competed voice traffic in bandwidth which caused increased in voice delay. As shown in Figure 5.20, the voice packet E2E delay rises suddenly approximately to 1300 ms due to high congestion of mentioned traffic then fall down to 350 ms at end of simulation. It is clearly the E2E delay level became high in WiMAX network under different network loads, where all traffic are initiated at 100$^{th}$ second of simulation.

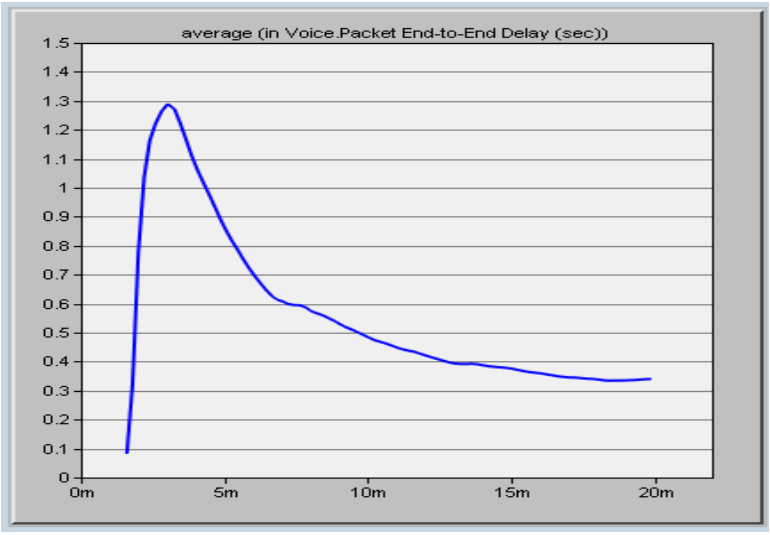



Figure 5.20: Global Statistics Voice Packet E2E Delay of WiMAX Case 5 Scenario

- **Objects Statistics Voice Packet E2E Delay of WiMAX Case 5 Scenario**

Figure 5.21 and Table 5.10 show high E2E delay on all users. In WiMAX Case 5 scenario all users do VoIP calls and request high (HTTP, Email and FTB) services. The blue line shows the E2E delay of the UE1 while red, green and turquoise lines show the E2E delay of UE 2, UE 3 and UE 4 respectively. As can be seen in Figure 5.21 the highest E2E delay is recorded on UE1 with 379.60 ms on average, on UE2 with 345.06 ms and on UE3 with 323.27 ms while E2E delay on UE4 is 311.90 ms. This means after UEs requested high HTTP, Email and FTB networks at VoIP calls, the E2E delay became high in WiMAX case 5 compared with WiMAX case 4 and case 3 scenarios.

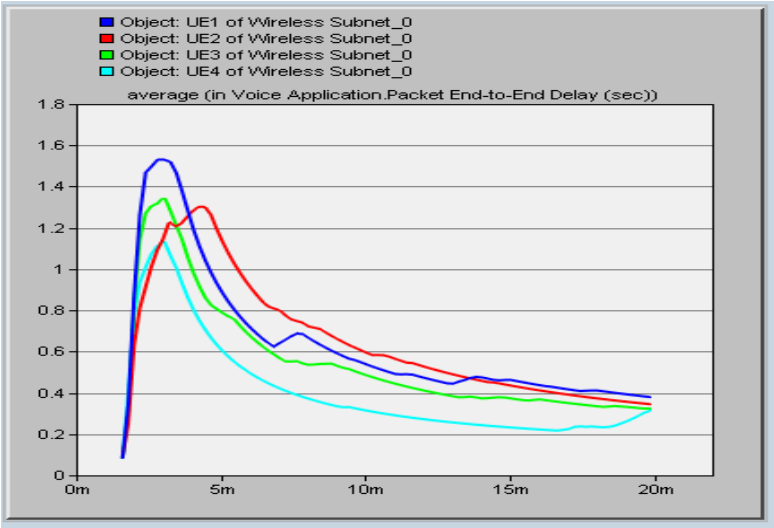



Figure 5.21: Objects Statistics Voice Packet E2E Delay of WiMAX Case 5 Scenario

For example, if have been done comparing for delay performance between case 5 and case 3 scenarios, where VoIP network in case 3 scenarios was loaded with high HTTP network only while here VoIP network is loaded with high (HTTP, Email and FTB) networks. Thus the different between results as following: The E2E delay on UE1 varied from 88 ms to 379.60 ms and on UE2 is varied from 96 ms to 345.06 ms, while on UE3 is varied from 90 ms to 323.27ms, finally E2E delay on UE4 is varied from 87 ms to 311.90 ms. This means E2E delay on all UEs are affected by increasing number of applications have requested at VoIP calls.

Table 5.10 Objects Statistics Voice Packet E2E Delay (ms) of WiMAX Case 5 Scenario

| Object Name | Minimum | Average | Maximum |
|---|---|---|---|
| UE1 | 84.16 | 379.60 | 2444.1 |
| UE2 | 86.39 | 345.06 | 1849.5 |
| UE3 | 83.65 | 323.27 | 2066.3 |
| UE4 | 84.45 | 311.90 | 1640.3 |

From simulation results can conclude, all users sustain from high delay. The UEs dissatisfied but E2E delay is still acceptable for a VoIP call due to the packet E2E delay does not exceed the threshold value of 400 ms as mentioned in Table 3.5.

- **Case 6 Scenario : E2E Delay Performance for Congested VoIP with High HTTP, Email, FTB and Video Conferencing Traffic Loads in WiMAX Network**

In this simulation have been used same parameters had used in case 5 scenario addition to Video Conferencing. The purpose of this simulation is to find out how WFQ scheme deals with much more traffic flows. Every traffic flow here has its specificity in terms of type of service. The parameters have used in simulation were explained in Table 4.2.

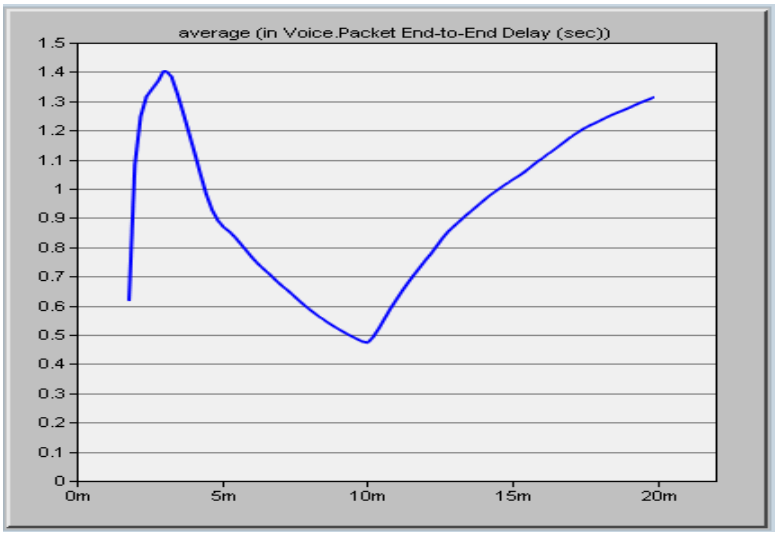



Figure 5.22: Global Statistics Voice Packet E2E Delay of WiMAX Case 5 Scenario

From the curve observation in Figure 5.22, the curve represents high E2E delay in two stages. The first, when all user do VoIP calls and requested high (HTTP, Email and FTB) services at100th second of simulation, the E2E delay rises up approximately to 1400 ms and then fall down to 475 ms. The second when two users transmitted high video conferencing with type of service (Interactive Multimedia) at 600th second of simulation, the E2E delay rises up to 1300 ms. It is clearly the E2E delay became high in WiMAX network when loaded with real-time application such as video conferencing.

- **Objects Statistics Voice Packet E2E Delay of WiMAX Case 6 Scenario**

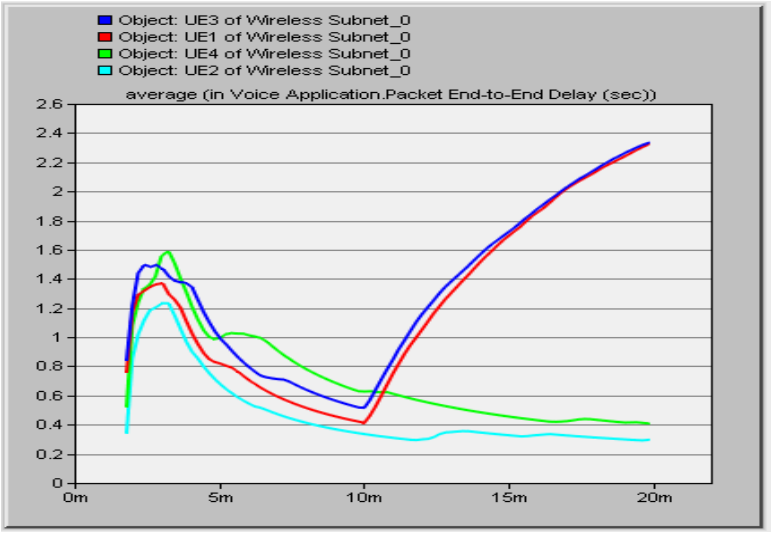



Figure 5.23: Objects Statistics Voice Packet E2E Delay of WiMAX Case 6 Scenario

Topmost curves in Figure 5.23 show very high delay especial on UE3 which represented with blue line and on UE1represented with red line. As shown in Figure 5.23 and illustrated in Table 5.11, the highest delay was recorded on UE3 with 2331.5 ms and on UE1 with 2321.1 ms. The phenomena behind rising delay on UE3 and UE1 due to Video Conferencing transmission from UE4 and UE2 at 600th second of simulation.

The green line and turquoise lines show the E2E delay of UE4 and UE 2 respectively. As can be seen in Figure 5.23 and Table 5.11, the E2E delay is recorded on UE4 408.5 and on UE2 with 295.3 ms. Delays for these four bearers are dissatisfied and not acceptable from QoS point of view. In order to maintain the quality for VoIP, the delay should be equal or less than 400

ms. If have seen to UE2 with delay 295.3 ms, possibly the delay accepted, but should be seen to UE1 which has delay with 2321.1 ms, where the VoIP calling was happened between both UE1& UE2, and the two users can not be listening to gather. Thus VoIP calling between them cannot be accepted.

Table 5.11 Objects Statistics Voice Packet E2E Delay (ms) of WiMAX Case 6 Scenario

| Object Name | Minimum | Average | Maximum |
|---|---|---|---|
| UE3 | 87.752 | 2331.5 | 4311.2 |
| UE1 | 87.828 | 2321.1 | 4631.1 |
| UE4 | 87.546 | 408.5 | 2370.7 |
| UE2 | 88.092 | 295.3 | 1462.6 |

### 5.1.1.4 Summary of E2EDelay Performance

As it known in this study, six scenarios have created for each network; three scenarios for UMTS and three scenarios for WiMAX network. Scenario 1for each network acts as VoIP without QoS and scenario 2 acts as VoIP with QoS (Base Line Scenario), while scenario 3 acts as VoIP congested with different networks. In UMTS Scenario 3, VoIP is congested with HTTP network only which implemented in three cases (low, medium and high). In WiMAX Scenario 3, the scenario is separated to six cases. From case 1 to case 4, VoIP is congested with (low, medium, high and very high) HTTP network respectively. In case 5, VoIP is congested with high (HTTP, Email and FTB networks). Finally, in case 6, VoIP is congested with high (HTTP, Email, and FTB and Video Conferencing networks).

As shown in Table 5.12, in UMTS without QoS scenario, the voice E2E delay is reached about 2 seconds on average, while in WiMAX without QoS scenario, the voice E2E delay is reached about 14 seconds on average. In UMTS Base Line Scenario, the voice E2E delay is about 120 ms, while in WiMAX Base Line Scenario, the voice E2E delay is about 62 ms. In UMTS scenario 3 in case 1 scenario the voice E2E delay is about 124 ms, while in WiMAX scenario 3 in case 1 scenario the voice E2E delay is about 90 ms. In UMTS case 2 the voice E2E delay is about 150 ms, while in WiMAX case 2 the voice E2E delay is about 100 ms. In UMTS case 3 the voice E2E delay is about 190 ms, while in WiMAX case 3 scenario the voice E2E delay is about 120 ms. In WiMAX case 4 the voice E2E delay is about 240 ms, WiMAX case 5 the

voice E2E delay is about 340 ms and in case 6 scenario the voice E2E delay reached to 1340 ms on average.

Table 5.12 10 Comparison of Average E2E Delay of UMTS and WiMAX networks

| **Scenarios** | | **End-to-End Delay (ms)** | |
|---|---|---|---|
| | | **UMTS Network** | **WiMAX Network** |
| **Scenario 1 VoIP without QoS** | | 2 Seconds | 14 Seconds |
| **Scenario 2 Base Line VoIP** | | 120 ms | 62 ms |
| **Scenario 3** | | | |
| **VoIP Congested with HTTP** | Case 1 | 124 ms | 90 ms |
| | Case 2 | 150 ms | 100 ms |
| | Case 3 | 190 ms | 120 ms |
| | Case 4 | – | 240 ms |
| **VoIP Congested with HTTP, Email and FTB** | Case 5 | – | 340 ms |
| **VoIP Congested with HTTP, Email, FTB and Video Conferencing** | Case 6 | – | 1339 ms |

From collected results, have been compare the results with respect to case 1 scenario of each network. In UMTS network with respect to UMTS case 1, the E2E delay in case 2 is higher than 17.3 %, while in case 3, E2E delay is higher than 34.7%.

In WiMAX network with respect to case 1, the E2E delay in case 2 is higher than 10 % and in case 3, E2E delay is higher than 25% and in case 4 is higher than 62.5 % while the in case 5 is higher than 73.5 %, while in case 6 scenario is higher than about 93 %, Figure 5.24 will shows the comparison between different cases scenarios in UMTS and WiMAX networks.

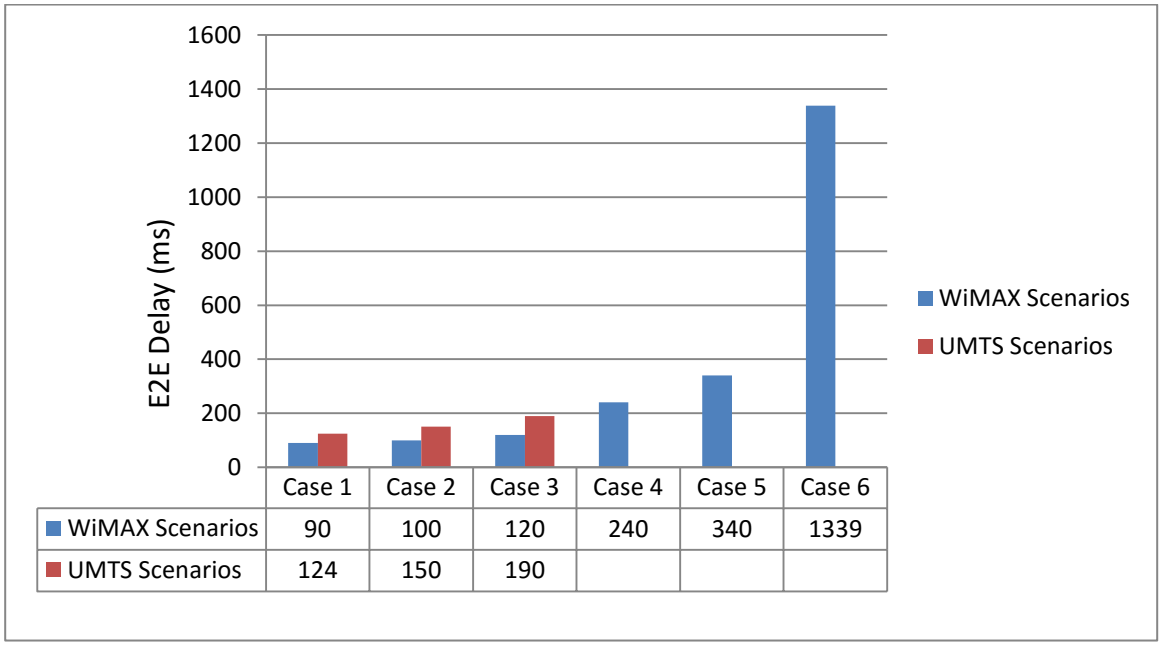



Figure 5.24: Comparison of Average E2E Delay Cases of UMTS and WiMAX Networks

### 5.1.2 Jitter Performance

Jitter in the VoIP is the variation of delay over different packets. It is measured as the time difference between the consecutive packets. Jitter significantly impacts the quality of the transmission real-time applications and is included as a performance parameter due to its importance at the transport layer in packetized data systems. There are various sources of jitter, such as limited buffer size, Network congestion and delay due to longer routes for the packets, which affect the communication, link quality.

#### 5.1.2.1 Jitter Performance for VoIP Network without QoS

Figure 5.25 shows the average voice jitter for VoIP without QoS scenario. The X-axis shows the simulation time in minute and Y-axis represents the voice jitter in second. Initially it can be seen that the voice jitter without QoS for UMTS is represented with blue line and for WiMAX is represented with the red line. The jitter in UMTS is fall down from approximately 7.5 ms to 1.6 ms and remained stable at this value till end of simulation, while in WiMAX, the jitter rises sharply up to 16 ms then fall slowly to 1ms at $12^{th}$ minute of simulation. This jitter dissatisfied

from starting calling till reached 12th minute of simulation due to non-guaranteed bit rate through this interval.

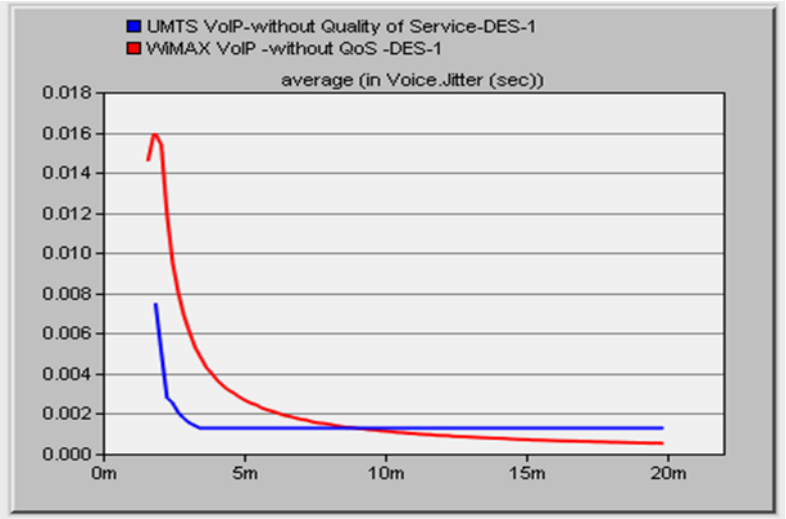



Figure 5.25: Comparisons Global Statistics Voice Jitter for VoIP without QoS between UMTS and WiMAX

Finally after 12 minute, the jitter without QoS begins deceases slowly till the end of simulation. From observed curves, the two values of voice jitter are not suitable for VoIP calls and are not acceptable according to the table 3.5, because the jitter value should be less than 1ms. Thus, it can be conclude the jitter must be maintained and minimized for appropriate VoIP calls. For the reason, have been implement QoS mechanism such as (DiffServ) in the networks to jitter be better.

### 5.1.2.2 Jitter Performance for Base Line VoIP Network

Figure 5.26 shows the average voice jitter of Base Line VoIP scenario for both networks. The X-axis shows the simulation time in minute and Y-axis represents the voice jitter in second. Initially it can be seen that the voice jitter with QoS for UMTS is represented with blue line and for WiMAX is represented with the red line. From Figure 5.34 can be observe the jitter in UMTS network is rose sharply from -6 ms then closed to 0 ms, while in WiMAX network, the jitter is closed to 0 ms from initial of simulation, it is clearly the jitter has maintained and minimized by implementing QoS in the networks. The jitter value is accepted in both networks

and suitable for VoIP calls. From given results, it is clearly that QoS mechanism plays a critical role in transfer real-time applications such as VoIP.

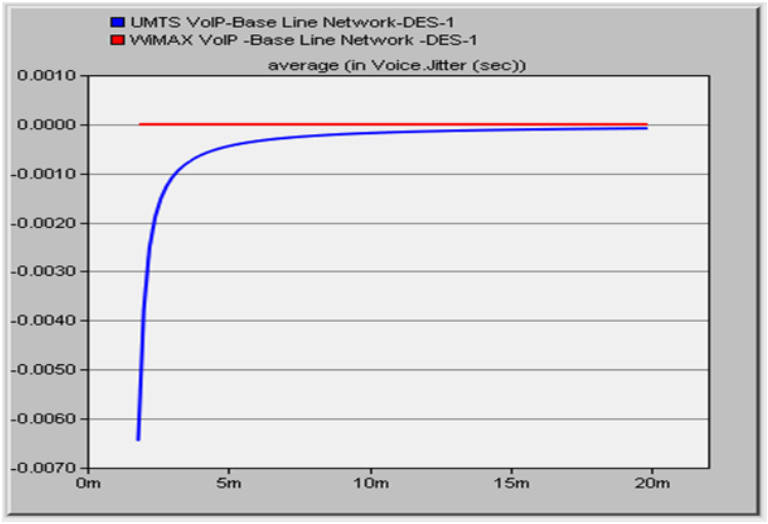



Figure 5.26: Comparison Global Statistics Voice Jitter for Base Line VoIP network between UMTS and WiMAX

### 5.1.2.3 Jitter Performance for Congested VoIP with Different Traffic Loads

In this section, the jitter results for Congested VoIP with Different Traffic Loads scenarios are presented in various statistical plots

- **Jitter Performance for Congested VoIP with HTTP Traffic Load**

Figure 5.27 (a) shows comparison between different cases of voice jitter for VoIP Congested with different network. The X-axis shows the simulation time in minute and Y-axis represents the jitter in second for voice jitter. Red line represents UMTS case 3, blue line represents UMTS case 2 and green line represents UMTS case 1 scenario, yellow line represents WiMAX case 1 scenario, turquoise line represents WiMAX case 2 and medium violet red line represents WiMAX case 3. From curves shown in Figure 5.27 (a), average voice jitter in UMTS case 1 and case 2 are in stable state, while UMTS case 3 scenario has bad jitter, this bad jitter due to high HTTP traffic loads (150000 bytes). It can be seen from the graph with red line, the jitter is

decreased sharply to minus (− 6 ) ms, then increased slowly to −1 ms at 12th minute of the simulation, this low negative value of jitter represents unstable received packet at

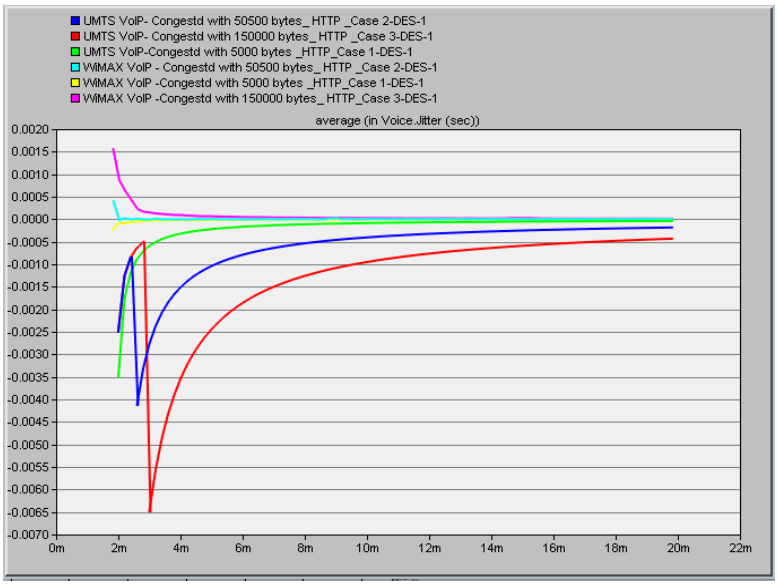



Figure 5.27 (a): Comparison Global Statistics Voice Jitter for Case 1, 2 and 3 Scenarios of UMTS and WiMAX

destination. The negative jitter is referred to the time difference between the packets at the destination node was less than time at the source node or the packet arrives earlier than expected time as mentioned in chapter three. On other hand in WiMAX cases scenarios, Figure 5.27(a) shows the average voice jitter of WiMAX case 1, 2 and 3 scenarios in suitable state, jitter nearly closed to zero ms.

- **Jitter Performance for Congested VoIP with High HTTP, Email, FTB and Video Conferencing Traffic Loads of WiMAX Network**

Figure 5.27 (b) shows comparison between different cases of voice jitter for VoIP Congested with different network. The X-axis shows the simulation time in minute and Y-axis represents the jitter in second for voice jitter. Green line represents WiMAX case 4, Red line represents WiMAX case 5 and Blue line represents WiMAX case 6 scenario, At the beginning of the simulation, the Jitter at case 4 scenario decreases sharply from 1.8 ms and in fast convergence to closed nearly 0 ms at 300th second of simulation . In case 5 scenario, the average voice jitter is rose up from 0.5 ms to 2.5 ms from starting simulation to 300th second of simulation, this

interval of simulation has bad consecutive of packets, after that jitter is stable to end of simulation. The jitter in beginning of simulation was not stable it causes some delay in packets

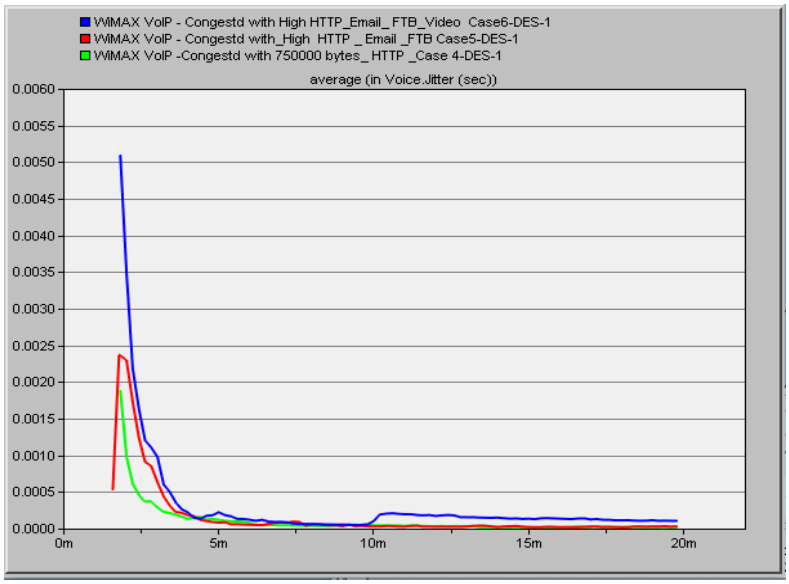



Figure 5.27 (b): Global Statistics Voice Jitter for Case 4, 5 and 6 Scenarios of WiMAX

sent. It can be seen that the jitter WiMAX case 6 scenario increasing sharply to 5 ms at the beginning of the simulation then decreases slowly until 300th second of simulation, this interval of simulation has bad consecutive of packets, therefore packets maybe unbuffered and might delayed. At 600$^{th}$ of simulation, jitter increases sharply to 0.5 ms second of simulation due to high load of Video Conferencing from UE2&UE4.

### 5.1.2.4 Summary of Jitter Performance

For WiMAX network, it has high jitter in scenario without QoS. In case 1, 2, 3, 4 scenario, it has a narrow range and has a fast convergence to closed nearly 0 ms. In case5 scenario the jitter begin varies and increases due high traffic load which caused significant delay. In case 6 scenario, the jitter has a significant variation, is reached unacceptable state at beginning of simulation, and takes some time to converge to the stable state, her the buffer cannot be re arrangement packets due to a large range of jitter variation, thereby leading to a poor quality for the user. For UMTS network it has high jitter in scenario without QoS, which is recorded 1.6 ms. In other scenarios, it has a negative jitter this phenomenon can be explained as follows: as the loaded of traffic increases, the congestion in the system also increases which caused to

the slow packet scheduling. The real-time application such as VoIP will end up in more increased time for buffering the conversational voice. This will cause delay in transferring packet at receiving terminal, thereby leading to a poor quality voice for the users.

### 5.1.3 Packet Loss Performance

Packet loss is an important metric to measure the quality of data transmission. The packet loss refers to the losing one or more packets when transmit from source to destination across a network. A VoIP user is satisfied if more than 97% of its speech packets are delivered successfully, The following equation 5.1 is used to determine the packet loss percentage.

$$Packet\,Loss = \left(\frac{Sent\ Packet - Received\ Packet}{Sent\ Packet}\right) * 100 \qquad (5.1)$$

#### 5.1.3.1 Packet Loss Performance for VoIP Network without QoS

Figure 5.28 represents VoIP without QoS scenario for UMTS network, and Figure 5.29 represents VoIP without QoS scenario for WiMAX network. X-axis represents the simulation

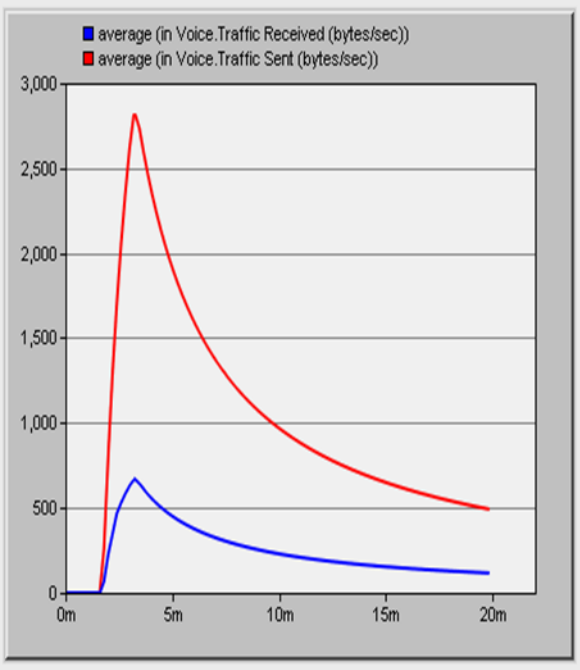



Figure 5.28: Total Sent and Received Data for VoIP without QoS of UMTS Network

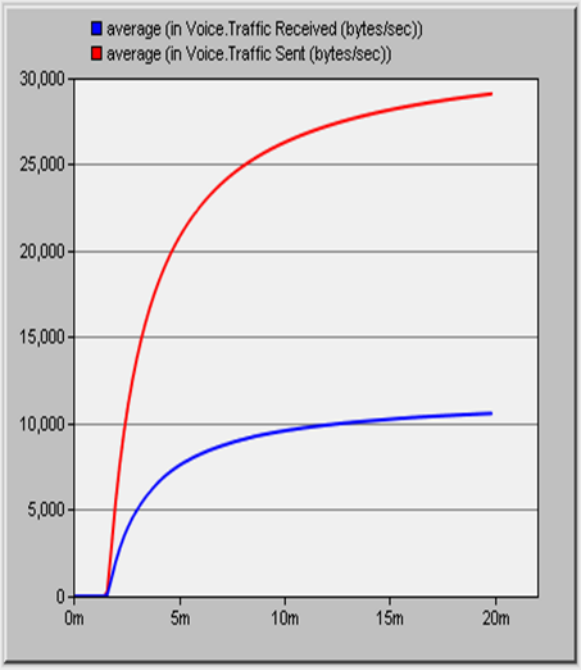



Figure 5.29: Total Sent and Received Data for VoIP without QoS of WiMAX Network

time in minute and Y-axis determines the sent and received data in bytes per second. Red line indicates the sent data while blue line represents the received data. The difference between sent and received data starts increasing significantly from beginning simulation in both UMTS and WiMAX networks. The packet loss for VoIP without QoS in UMTS is recorded 76%, while in WiMAX is recorded 63% as shown in Table 5.13 and 5.14. As a result of the packet loss is more than 3% that is not acceptable in VoIP transmission as mentioned in table 3.5.

Table 5.13: Total Sent and Received Voice Traffic (bytes/sec) for VoIP without QoS of UMTS Network

| UMTS VoIP without QoS Scenario | Minimum | Average | Maximum | Average Packet Loss |
|---|---|---|---|---|
| Voice traffic sent | 0 | 29086.4 | 32000 | 63% |
| Voice traffic received | 0 | 10581.8 | 12313.3 | |

Table 5.14: Total Sent and Received Voice Traffic (bytes/sec) for VoIP without QoS of WiMAX Network

| WiMAX VoIP without QoS Scenario | Minimum | Average | Maximum | Average Packet Loss |
|---|---|---|---|---|
| Voice traffic sent | 0 | 492.97 | 6600 | 76% |
| Voice traffic received | 0 | 116.73 | 6600 | |

### 5.1.3.2 Packet Loss Performance for Base Line VoIP Network

Figure 5.30 represents VoIP with QoS scenario for UMTS network, and Figure 5.31 represents VoIP with QoS scenario for WiMAX network. X-axis represents the simulation time in minute and Y-axis determines the sent and received data in bytes per second. Red line indicates the sent data while blue line represents the received data. In order to reduce the packet loss, DiffServ mechanism has been used to support QoS. DiffServ has described in in detail in chapter three. Figure 5.30 depicts the packet loss in VoIP when QoS is employed in UMTS network. From this figure, it is apparent that the red line is overlapped by the blue line. That means that the total sent and received data for VoIP is almost equal. It means that the packet

loss in UMTS scenario with QoS is much less than the scenario without QoS. However at the 15th minute, the received data is a little bit lower than the data sent which is difficult to distinguish. This implies that the packet loss is 2.2% as shown in Table 5.15, while in WiMAX both sent and received voice traffic are overlapped and the packet loss is 0.0% as shown in Table 5. 16. The results of both networks are below 3%. The packet loss ratios are acceptable for VoIP transmission according to table 3.5.

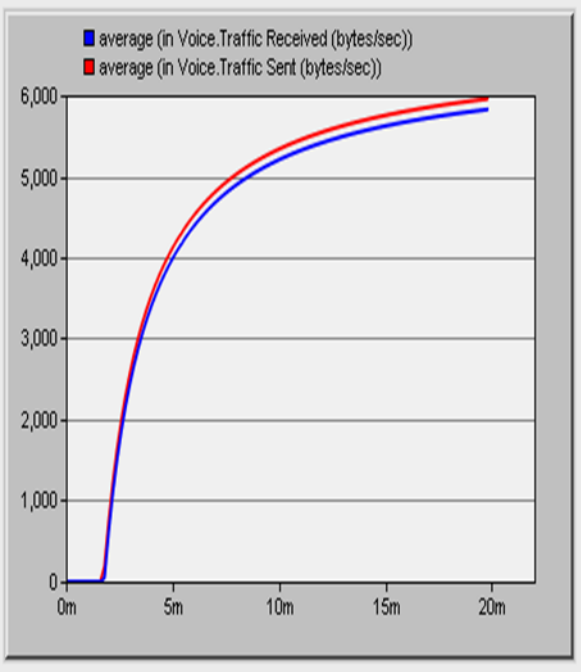



Figure 5.30: Total Sent and Received Data for Base Line VoIP of UMTS

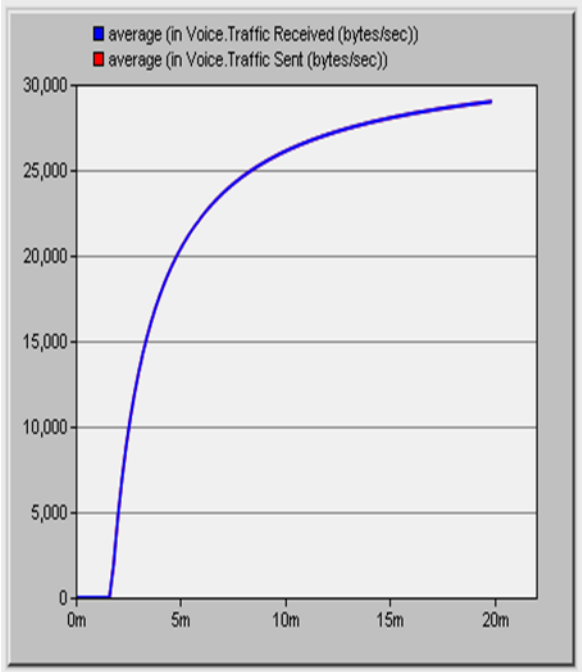



Figure 5.31: Total Sent and Received Data for Base Line VoIP of WiMAX

Table 5.15: Total Sent and Received Voice Traffic (bytes/sec) for Base Line VoIP of UMTS Network

| **UMTS Base Line VoIP Scenario** | **Minimum** | **Average** | **Maximum** | **Average Packet Loss** |
|---|---|---|---|---|
| Voice traffic sent | 0 | 5958.1 | 6600 | 2.20% |
| Voice traffic received | 0 | 5827.5 | 6616.5 | |

Table 5.16: Total Sent and Received Voice Traffic (bytes/sec) for Base Line VoIP of WiMAX Network

| WiMAX Base Line VoIP Scenario | Minimum | Average | Maximum | Average Packet Loss |
|---|---|---|---|---|
| Voice traffic sent | 0 | 28996 | 64000 | 0.00% |
| Voice traffic received | 0 | 28996 | 64000 | |

### 5.1.3.3 Packet Loss Performance for Congested VoIP with Different Traffic Loads

In this section, the packet loss performance results for Congested VoIP with Different Traffic Loads scenarios are presented in various statistical plots

- **Case 1 Scenario: Packet Loss Performance for VoIP Congested with Low HTTP Traffic Load**

Figure 5.32 represents packet loss performance of UMTS case 1 scenario, and Figure 5.33 represents packet loss performance of WiMAX case 1 scenario. X-axis represents the

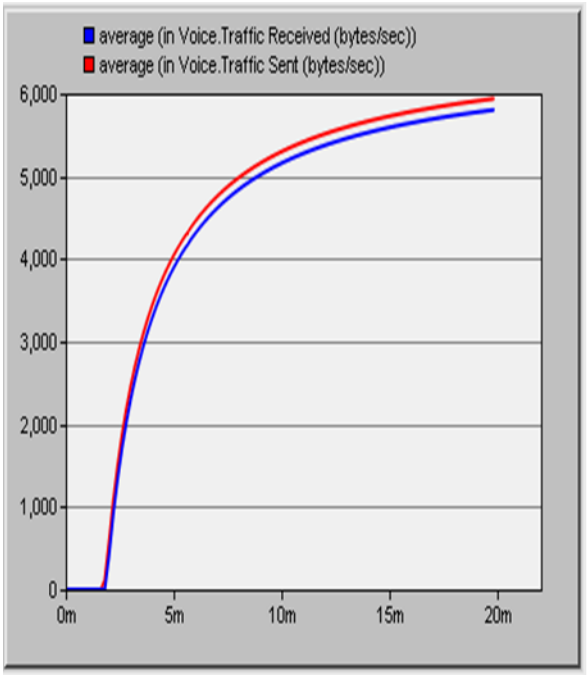



Figure 5.32: Total Sent and Received Data for UMTS Case 1 Scenario

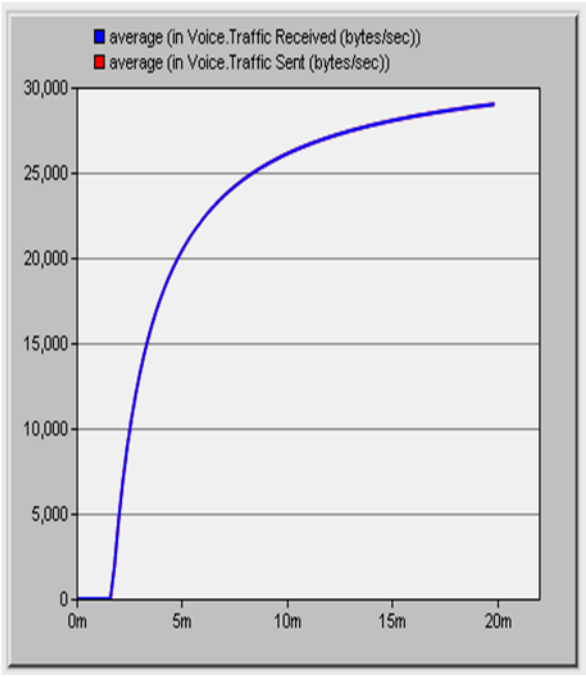



Figure 5.33: Total Sent and Received Data for WiMAX Case 1 Scenario

simulation time in minute and Y-axis determines the sent and received data in bytes per second. Red line indicates the sent data while blue line represents the received data in both Figures. Figure 5.32 appears packet loss performance of UMTS case 1 scenario that the blue line is overlapped with redline, which means that the total sent and received data of voice traffic is almost equal. However at the $10^{th}$ minute of simulation, the received data is a little bit lower than the data sent which is difficult to distinguish. This implies that the packet loss UMTS case 1 scenario is 2.27% as illustrated in Table 5.17, which is below 3%. The packet loss ratio is acceptable for VoIP in case 1 scenario of UMTS network according to table 3.5.

Figure 5.33 appears packet loss performance of WiMAX case 1 scenario that the blue line is overlapped with redline, which means that the total sent and received data of voice traffic is almost equal. This implies that the packet loss is almost Zero as illustrated in Table 5.18, which is acceptable for VoIP in case 1 scenario of WiMAX network according to table 3.5. From Figures 5.32 and 5.33, it can be conclude that the packet loss increases something in UMTS network due to congestion with low HTTP traffic and did not increased in WiMAX network. From observation, Figure 5.32 and 5.33, explain that packet loss can be reduced and traffic congestion avoided by implementing proper QoS mechanism in the both networks.

Table 5.17: Total Sent and Received Voice Traffic (bytes/sec) for Case 1Scenario of UMTS Network

| **UMTS Case 1: Scenario** | **Minimum** | **Average** | **Maximum** | **Average Packet Loss** |
|---|---|---|---|---|
| voice traffic sent | 0 | 5939.3 | 6600 | 2.27**%** |
| voice traffic received | 0 | 5804.1 | 6644.1 | |

Table 5.18: Total Sent and Received Voice Traffic (bytes/sec) for Case 1Scenario of WiMAX Network

| **WiMAX Case 1: Scenario** | **Minimum** | **Average** | **Maximum** | **Average Packet Loss** |
|---|---|---|---|---|
| Voice traffic sent | 0 | 4332.5 | 4800 | 0.00% |
| Voice traffic received | 0 | 4332.5 | 9660 | |

- **Case 2 Scenario : Packet Loss Performance for Congested VoIP with Medium HTTP Traffic Load Scenarios**

Figure 5.34 represents UMTS case 2 scenario, and Figure 5.35 represents WiMAX case 2 scenario. In case 2 scenarios for both networks the interactive voice is congested with medium HTTP traffic load (50500 bytes). Figure 5.34 shows sent data represented with red line is overlapped by received data represented with blue line. The sent and received data are overlapped till 300 seconds of simulation, and then the difference between sent and received data is increased significantly which means the packet loss is increased.

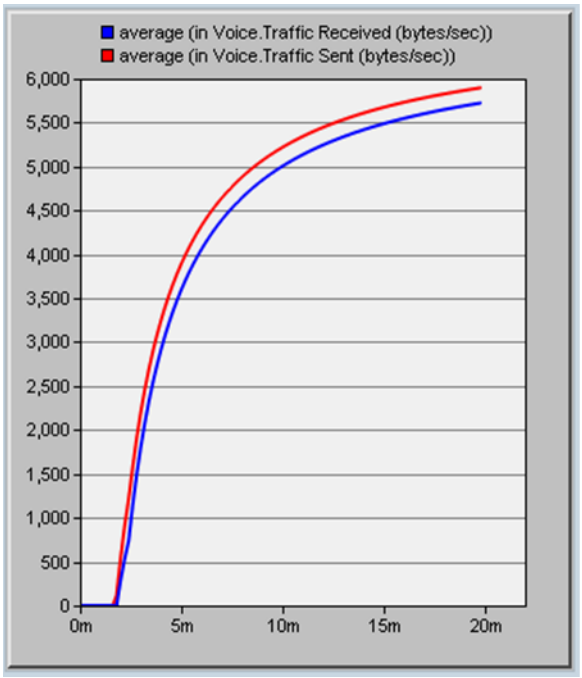



Figure 5.34: Total Sent and Received Data for UMTS Case 2 Scenario

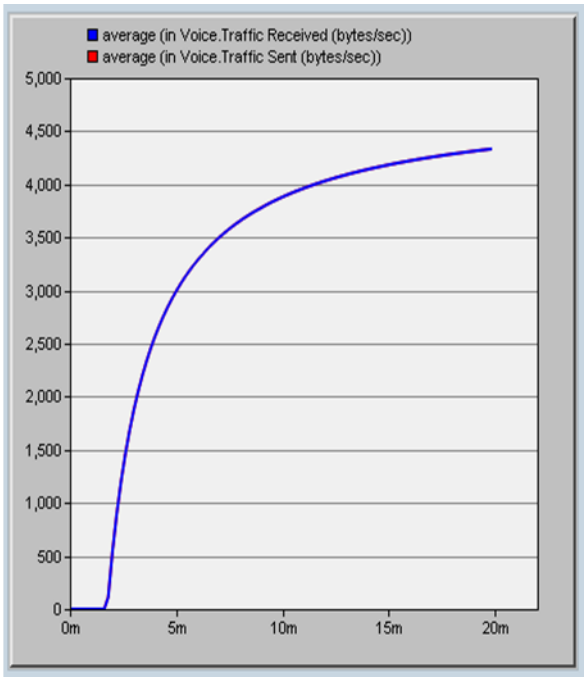



Figure 5.35: Total Sent and Received Data for WiMAX Case 2 Scenario

The packet loss in UMTS case 2 scenario is more than the packet loss in UMTS case 1 scenario. However the packet loss is important to distinguish. Table 5.19 illustrates the packet loss is about 2.94%, which is still below 3%. The packet loss ratio is still acceptable for VoIP transmission according to table 3.5. Figure 5.35 illustrates the packet loss when VoIP is congested with medium HTTP traffic load (50500 bytes) in WiMAX network. The total sent and received data for voice traffic is much close to each other. That means that the total sent

and received data are almost equal. It is clearly the packet loss ratio is very negligible as illustrated in Table 5.20 with 0.0023%. The packet loss performance is acceptable for VoIP transmission according to table 3.5.

Table 5.19: Total Sent and Received Voice Traffic (bytes/sec) for Case 2 Scenario of UMTS Network

| **UMTS Case 2: Scenario** | **Minimum** | **Average** | **Maximum** | **Average Packet Loss** |
|---|---|---|---|---|
| Voice traffic sent | 0 | 5898.2 | 6600 | 2.94% |
| Voice traffic received | 0 | 5724.9 | 6886.1 | |

Table 5.20: Total Sent and Received Voice Traffic (bytes/sec) for Case 2Scenario of WiMAX Network

| **WiMAX Case 2: Scenario** | **Minimum** | **Average** | **Maximum** | **Average Packet Loss** |
|---|---|---|---|---|
| Voice traffic sent | 0 | 4332.3 | 4800 | 0.0023**%** |
| Voice traffic received | 0 | 4332.2 | 4820.1 | |

- **Case 3 Scenario: Packet Loss Performance for Congested VoIP with Large HTTP Traffic Load**

Figure 5.36 illustrates UMTS case 3 scenario and Figure 5.37 illustrates WiMAX case 3 scenario. Here VoIP is congested with high HTTP network (150000 bytes). Red line indicates the sent data while blue line represents the received data in both Figures. As observed in figure 5.36, sent and received data are different in all simulation intervals, where the sender began in the loss packets from the beginning of simulation. This is indicator to high packet loss in UMTS case 3 scenario. The sent and received traffic value is referred in Table 5.21; the packet loss is achieved ratio 3.63% which is not acceptable according to table 3.5, that the packet loss should not exceeds 3%. The reasons behind high packet loss in UMTS case3 scenario is large packet size of HTTP network which caused to traffic congestion in UMTS network. Figure 5.37 demonstrates the sent and received traffic of WiMAX case 3 scenario for VoIP congested

with high HTTP network scenario. As can be seen in figure 5.33, sent and received data are overlapped between red line and blue line at all case simulation intervals. It means that the packet loss in WiMAX case 3 scenario is nearly.0023% as illustrated in Table 5.22 The packet loss is negligible which is acceptable for VoIP transmission according to table 3.5. It can be concluded from Figure 5.36 and Figure 5.37 that the packet loss performance in UMTS network is higher than the standard recommendation. So the UMTS network in case 3 scenarios can't support VoIP transmission when loaded with high HTTP traffic (150000 bytes).

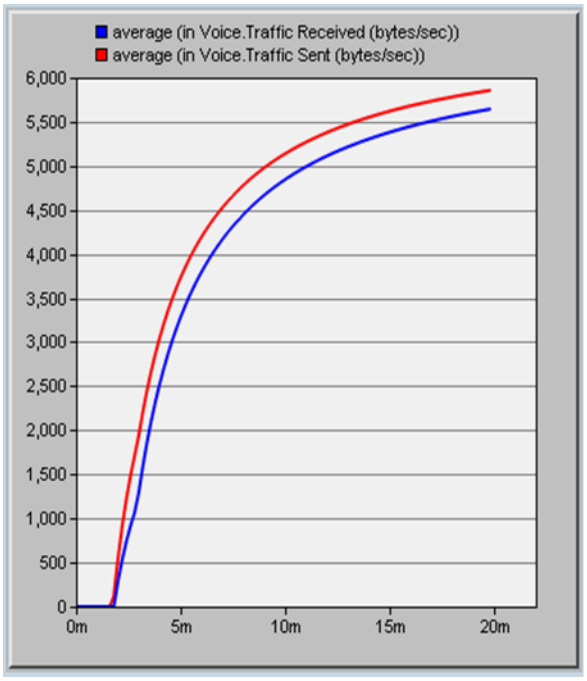



Figure 5.36: Total Sent and Received Data for UMTS Case 3 Scenario

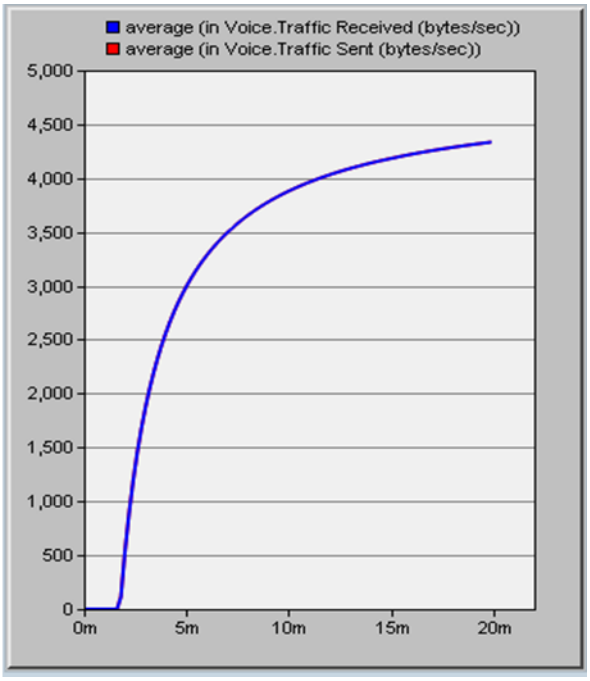



Figure 5.37: Total Sent and Received Data for WiMAX Case 3 Scenario

Table 5.21: Total Sent and Received Voice Traffic (bytes/sec) for Case 3 Scenario of UMTS Network

| **UMTS Case 3: Scenario** | **Minimum** | **Average** | **Maximum** | **Average Packet Loss** |
|---|---|---|---|---|
| Voice traffic sent | 0 | 5856.7 | 6600 | 3.63% |
| Voice traffic received | 0 | 5643.8 | 6660.6 | |

Table 5.22: Total Sent and Received Voice Traffic (bytes/sec) for Case 3 Scenario of WiMAX Network

| WiMAX Case 3: Scenario | Minimum | Average | Maximum | Average Packet Loss |
|---|---|---|---|---|
| Voice traffic sent | 0 | 4332.5 | 4800 | 0.0023% |
| Voice traffic received | 0 | 4332.4 | 4886.7 | |

- **Case 4 Scenario: Packet Loss Performance for VoIP Congested with Very Large HTTP Traffic Load**

As mentioned in Table 5.21, the packet loss ratio in UMTS case 3 scenario reached 3.63% when VoIP congested with high HTTP traffic load. Which is not acceptable? Thus, UMTS network could not sustain loaded 150000 bytes of HTTP traffic. For this reason will be end the simulation in UMTS network and will continue with WiMAX network only for congested VoIP network. As mentioned before, more frame sizes of HTTP beyond (150000 bytes) have tested to investigate the impact on E2E delay and packet loss performance. Have been found the value of HTTP frame size can effects on voice E2E delay is (750000 bytes). Therefore

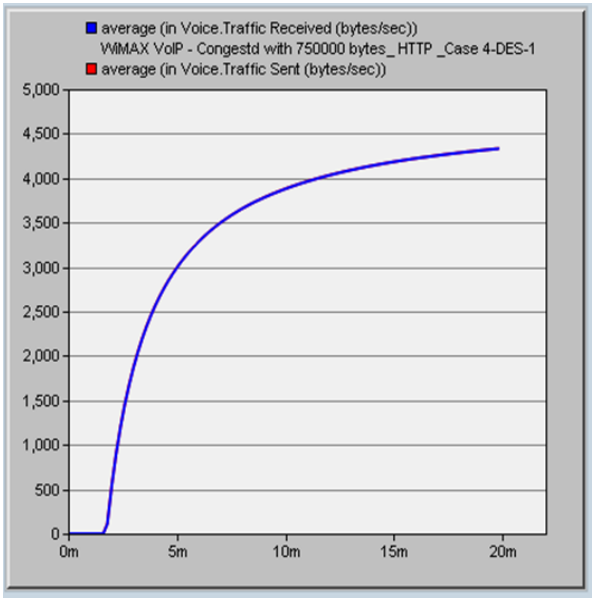



Figure 5.38: Total Sent and Received Data for WiMAX Case 4 Scenario

will be explain the result related (750000 bytes) and will not explain other tested results. Figure 5.38 shows packet loss performance of WiMAX case 4 scenario. Here VoIP network is

congested with very high HTTP network (750000 bytes). Red line indicates the sent voice data while blue line represents the received voice data. As observed in figure, sent and received data are overlapped between red line and blue line in all case simulation. The packet loss is 0.023% as illustrated in Table 5.23; thus the packet loss ratio is small. The packet loss performance is acceptable for VoIP transmission.

Table 5.23: Total Sent and Received Voice Traffic (bytes/sec) for Case 4 Scenario of WiMAX Network

| **WiMAX Case 4: Scenario** | **Minimum** | **Average** | **Maximum** | **Average Packet Loss** |
|---|---|---|---|---|
| Voice traffic sent | 0 | 4332.4 | 4800 | 0.023% |
| Voice traffic received | 0 | 4331.4 | 4920 | |

- **Case 5 Scenario: Packet Loss Performance for VoIP Congested with High HTTP, Email and FTB Traffic Loads**

Figure 5.39 shows the packet loss performance of WiMAX case 5 scenario. Red line indicates the sent data while blue line represents the received data. As can be observe from figure, sent and received data are overlapped between red line and blue line in all simulation. The packet

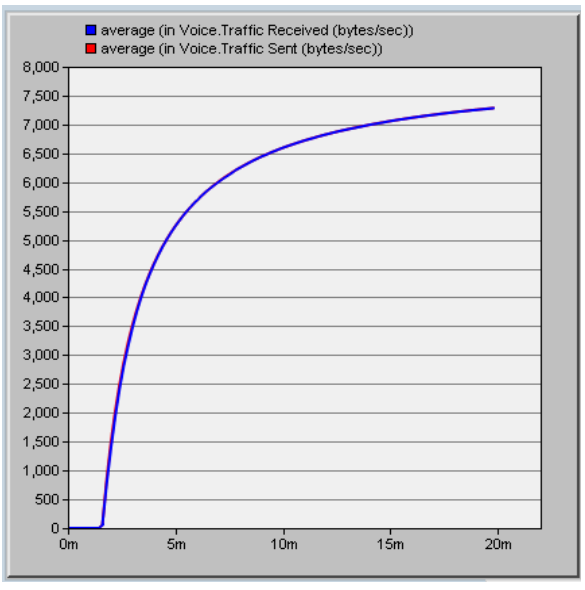



Figure 5.39: Total Sent and Received Data for WiMAX Case 5 Scenario

loss is 0.043% as illustrated in Table 5.24; the packet loss is still small, which is acceptable for VoIP transmission.

Table 5.24: Total Sent and Received Voice Traffic (bytes/sec) for Case 5 Scenario of WiMAX Network

| **WiMAX Case 5: Scenario** | **Minimum** | **Average** | **Maximum** | **Average Packet Loss** |
|---|---|---|---|---|
| Voice traffic sent | 0 | 7285.5 | 8000 | 0.043% |
| Voice traffic received | 0 | 7282.4 | 8593.3 | |

- **Case 6 Scenario: Packet Loss Performance for VoIP Congested with High HTTP, Email, FTB and Video Conferencing Traffic Loads**

Figure 5.40 shows the packet loss performance of WiMAX case 6 scenario. Red line indicates the sent data while blue line represents the received data. From figure, two lines are overlapped at starting of simulation till the 600$^{th}$ second of simulation when UE2 &UE4 loaded high video conferencing, at this time the sent and received data started in variation. The packet loss is 1.61% as illustrated in Table 5.25; the packet loss performance is significantly impaired

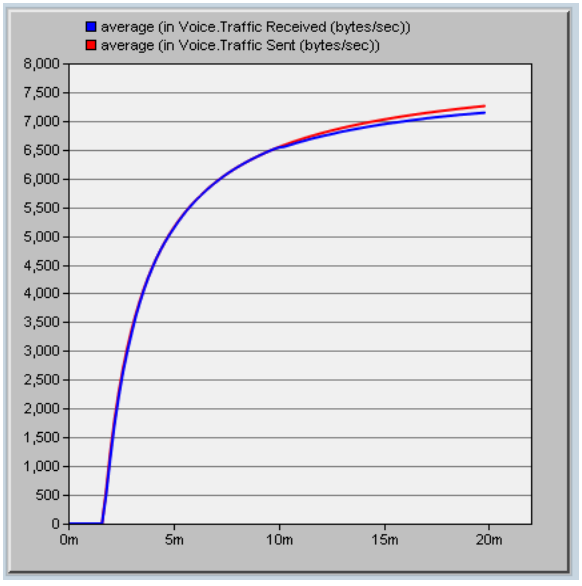



Figure 5.40: Total Sent and Received Data for WiMAX Case 6 Scenario

Table 5.25: Total Sent and Received Voice Traffic (bytes/sec) for Case 6 Scenario of WiMAX Network

| WiMAX Case 6: Scenario | Minimum | Average | Maximum | Average Packet Loss |
|---|---|---|---|---|
| Voice traffic sent | 0 | 7285.5 | 8000 | 1.61% |
| Voice traffic received | 0 | 7282.4 | 8593.3 | |

### 5.1.3.4 Summary of Packet Loss Performance

The goal of this section is to compare the packet loss performance between different simulation of UMTS and WiMAX networks. It can be conclude from Table 5.26, the packet loss performance of all voice traffic simulation corresponding to UMTS network experiences is high comparison to packet loss performance of WiMAX network experiences.

Table 5.26: Comparison for Packet Loss Performance Scenarios of UMTS and WiMAX Networks

| Scenarios | | Packet Loss Rate (%) | |
|---|---|---|---|
| | | UMTS Network | WiMAX Network |
| **Scenario 1 VoIP without QoS** | | 63 | 76 |
| **Scenario 2: VoIP Base Line** | | 2.2 | 0.00 |
| **Scenario 3** | | | |
| **VoIP Congested with HTTP** | **Case 1** | 2.27 | 0.00 |
| | **Case 2** | 2.94 | 0.0023 |
| | **Case 3** | 3.6 | 0.0023 |
| | **Case 4** | – | 0.023 |
| **VoIP Congested with HTTP, Email and FTB** | **Case 5** | – | 0.043 |
| **VoIP Congested with HTTP, Email, FTB and Video Conferencing** | **Case 6** | – | 1.61 |

## 5.2 Discussion

From simulated VoIP in 3G/UMTS and WiMAX 802.16e networks have been found the voice quality is influenced by several factors such as E2E delay, jitter and packet loss. These factors are the key parameters to measure the performance of conversational voice transmission over wireless networks. This thesis figured out reasons behind this influencing. One reason is the network congestion from large frame size; another reason is the size of bandwidth. For example UMTS network has a bandwidth with 5MHz comparison to WiMAX network has a bandwidth with 20MHz. jitter is another key issue in terms of VoIP calls. The performance of jitter depends on the encoding scheme and number of frames per packet. In order to obtain better performance for VoIP calls the application requesting should be properly sized at VoIP calling.

Appropriate queuing scheme such as WFQ is suitable to support other service classes such as best-effort through VoIP calling. By using this scheduling, low-priority queues have opportunity to transmit packets even though the high priority queues are not empty. Have the fairness is enforced in sharing the remaining bandwidth among classes, for each number of packets within WFQ scheduler a weight is assign. While in PQ, the priority is given to certain traffic, mostly real time applications (Voice and Video) over non real time applications (Email, HTTP and FTP). DiffServ has administrative control of delay, bandwidth and packet dropping preferences. The combination of these two schemes (DiffServ, WFQ) played an important role QoS. The proper adaptation in scheduling algorithm, sufficient bandwidth, avoidance traffic congestion and packet loss reduction will tend to good QoS and users satisfy.

# CHAPTER SIX

## Conclusion and Future work

## 6.1 Conclusion

In this thesis has been tested and evaluated QoS performance for VoIP in terms of three metrics (i.e., End-to-End delay, jitter and packet loss). The thesis is taken as a comparative study between 3G/UMTS and WiMAX (IEEE 802.16e) networks.

In research objective, have been investigated the performance of delivering VoIP traffic in terms GBR and Non-GBR carriers with the priority for users, In order to make this evaluation more realistic have been applied various web traffic loads with Non-GBR to make some congestion on VoIP network. Moreover other traffic loads such as Email, FTB and Video Conferencing with different service classes have been implemented to show how QoS mechanism serves VoIP traffic under these traffic loads.

So it can be seen how UMTS and WiMAX QoS work its best in the cases of highly loaded network. In this thesis OPNET simulation based study, six network scenarios (1- 6) have been simulated and analyzed. For both UMTS and WiMAX, two scenarios represent (VoIP network without QoS), other two scenarios represent Baseline VoIP network (with QoS), and other two scenarios represent VoIP congested with different networks load were separated in several cases.

According to the objectives of this thesis work, three research questions set out in chapter one were aimed to understand and investigate the performance of VoIP traffic, The Research Questions as following:

Q1. What is the impact on the packet end-to-end delay and packet loss performance for VoIP when GBR and Non-GBR bearers are established in UMTS and WiMAX networks?

Q2. What is the impact on the packet end-to-end delay and packet loss performances for VoIP traffic under different HTTP network loads?

Q3. What is the impact on the packet end-to-end delay and packet loss performances for VoIP traffic under HTTP, Email, FTB and video conferencing network loads with different bearers?

Answer 1: In both networks the E2E delay for VoIP with NGBR is found high compared to VoIP with GBR The existing QoS scheme enhanced the E2E delay and packet loss performances for VoIP transmission in both networks.

Answer 2: In cases 1and 2 scenarios of scenario three when VoIP loaded with low and medium HTTP traffic, both networks represent better performance in terms of E2E delay and packet loss. In case 3 when VoIP loaded with high HTTP traffic, the UMTS network represents bad performance in term of packet loss, but E2E delay but is still acceptable, some users were dissatisfied. While in WiMAX network, the performance remains quite. In case 4 when VoIP loaded with very high HTTP traffic, this case was applied on WiMAX network only, the E2E delay is significantly high but is still acceptable, while the packet loss performance remains quite. It has been found the VoIP in WiMAX network can be sustains the congestion of HTTP traffic when requested with high and very large sizes, while UMTS network sustained HTTP traffic when requested with small and medium sizes only.

Answer 3: This work related to case 5 and case 6 scenarios , the work was focused on the ability of VoIP quality in WiMAX network when VoIP suffered extra congestion of various traffic flows. For case 5, when VoIP network congested with high (HTTP, Email and FTB), the E2E delay is significantly high. Compared to case 4 the E2E delay began to suffer and significantly increased, many users have high delay and the packet loss performances were increased but are still acceptable. For case 6 scenario when VoIP network congested with high (HTTP, Email, FTB and Video Conferencing). The E2E delay is very much high and appeared bad. The users were recorded unacceptable delay and dissatisfied.

From thesis work, it can be concluded, the E2E delay and packet loss rate are increased as traffic congestion increased.

Finally this thesis offers five benefits; the optimal HTTP value can UMTS supports at VoIP calls was represented in UMTS case 1, the optimal HTTP value can WiMAX supports at VoIP calls was represented in WiMAX case 2, the HTTP value that affected on VoIP quality of UMTS network was represented in UMTS case 3, the HTTP value that affected in VoIP quality of WiMAX network was represented in WiMAX case 4 and suffering of E2E delay was represented in WiMAX case 5, when several traffic flows with different services are competing VoIP on bandwidth. While bad E2E delay was represented in WiMAX case 6.

## 6.2 Future Work

IEEE 802.16m represents a significant improvement over 802.16e-2005 latency and greatly the QoS for latency-sensitive real-time applications such as VoIP, on-line business and financial transactions, etc. The performance enhancements defined in IEEE 802.16m build on the capabilities established with IEEE 802.16e-2005, an OFDMA-based technology. This will provide the flexibility to meet the individual QoS parameters established for individual multimedia classes established by IMT-Advanced. The IEEE 802.16m amendment also supports channel aggregation to provide an effective bandwidth up to 100 MHz The channels do not need to have the same bandwidth nor do they need to be in the same frequency band. This capability will enable operators with access to multiple channels or licenses to achieve significantly higher peak and average data rates than is achievable with individual channels. Aggregating several 20 MHz channels, for example, could support peak data rates exceeding 1 Gigabit/sec.

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