Sudan University of Science and Technology

College of Engineering

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Performance Evaluation of QoS of VoIP over WiMAX Networks

A Research Submitted in Partial fulfillment for the Requirements of the Degree of B.Eng. (Honors) in Electronics Engineering

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استهلال

قال تعالى:

أعوذ بالله من الشيطان الرجيم

(وما تَوْفِيقِي إِلَّا باللهِ عَلَيْهِ تَوَكَّلْتُ وَإِلَيْهِ أُنِيبُ)

[سورة هود - 88]

صدق الله العظيم

Dedication

To our mothers, fathers, sisters, brothers and our friends …

ACKNOWLEDGEMENT

The success and final outcome of this project required a lot of guidance and assistance from many people and we are extremely privileged to have got this all along the completion of our project. We would like to thank our supervisor, Dr. FathELrahaman Ismael for encouragement, guidance, critics and friendship all advices and supports throughout this project. Also we would like to thank our friends that never late to give support. We always thank our parents and all of our families.

ABSTRACT

Due to its large coverage area, low cost of deployment and high speed data rates, WiMAX is a promising technology for providing wireless last-mile connectivity. Voice over Internet Protocol (VoIP) is one of the rapidly growing technologies and is expected to replace the conventional circuit switched voice services. VoIP is also considered as one of the applications of WiMAX which requires careful design of QoS configurations. As real-time application VoIP requires packet delivery with low latency, jitter, packet loss, and sufficient bandwidth. WiMAX technology defines 5 different service classes that can be used in order to satisfy QoS requirements of different applications. This research evaluates the performance of VoIP traffic over WiMAX networks. This has been done through the study and evaluation of several aspects related to the WiMAX network and VoIP configurations such as: WiMAX service classes, number of nodes and VoIP Codecs. The investigated parameters include: throughput, mean opinion score (MOS), jitter delay and end-to-end delay. The results were obtained through the design and simulation of different WiMAX network scenarios using OPNET Modeler. Results showed that UGS service class has the best performance parameters serving VoIP. It is also observed that the G.711 codec has lower end-to-end delay, higher MOS and higher throughput.

المستخلص

نظراً لسعة تغطيتها وانخفاض تكلفة تركيبها والسرعة العالية لنقل البيانات عبرها، تعتبر الواي ماكس تقنية واعدة لتوفير الاتصال اللاسلكي. تعتبر تقنية نقل الصوت عبر بروتوكول الإنترنت واحدة من التقنيات التي تنمو بسرعة، ويتوقع أن تحل محل التقنيات التقليدية لتقديم الخدمات الصوتية .كما تعتبر تقنية نقل الصوت عبر بروتوكول الإنترنت أيضا واحدة من تطبيقات شبكة واي ماكس التي تتطلب تصميم دقيق لتكوينات جودة الخدمة. يتطلب تطبيق الصوت عبر بروتوكول الإنترنت- كمثال علي تطبيقات الوقت الحقيقي- تسليم الحزمة بتأخير منخفض ومعدل التغيير في التأخير منخفض ،مع إنخفاض في فقدان الحزمة وعرض نطاق ترددي كافي. تعرف تقنية الوايماكس خمس فئات مختلفة لخدمة توصيل البيانات التييمكن استخدامها من أجل تلبية متطلبات جودة الخدمة في التطبيقات المختلفة. الهدف الرئيسي من هذا البحث هو دراسة وتحليل أداء تقنية نقل الصوت عبر بروتوكول الإنترنت على شبكات واي ماكس المتنقلة .وقد تم ذلك من خلال دراسة وتحليل العديد من الجوانب المتصلة بشبكة واي ماكس وتقنية نقل الصوت عبر بروتوكول الإنترنت مثل :طبقات الخدمة، والتنقل، وعدد العقد وتقنيات ترميز وفك ترميز الصوت .تشمل عوامل الأداء التي تمت دراستها :معدل معالجة البيانات، ومعدل رضا الزبائن عن الخدمة، وزمن وصول البيانات و الاختلاف في زمن وصول البيانات .وقد تم الحصول على النتائج من خلال تصميم ومحاكاة سيناريوهات مختلفة لشبكة واي ماكس باستخدام برنامج النمذجة والمحاكاة (أوبنت 14.5) وأظهرت النتائج أن' طبقة خدمة المنح غير المرغوب لديها أفضل معايير الأداء لخدمة نقل الصوت عبر بروتوكول الإنترنت .كما تلاحظ أيضا أن الترميز G.711 يمتاز بأدنى تأخير وأعلى قيمة لمعدل رضا الزبائن عن الخدمة ويرسل بأعلى طاقة انتاجية.

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List of symbols

t(i)	Time transmitted at transmitter (sec).
t'(i)	Time received at receiver (sec).
De2e	End to end Delay (sec).
Dn	Network Delay (sec).
De	Encoding Delay (sec).
Dd	Decoding Delay (sec).
Dc	Compression Delay (sec).
Dde	Decompression Delay (sec).

List of Abbreviation

1G	First Generation
2G	Second Generation
3G	Third Generation
3GPP	Third Generation Partnership Project
4G	Fourth generation
AAA	Authentication, Authorization and Accounting
AMC	Adaptive Modulation and Coding
ASN	Access Service Network
ASP	Application Service Provider
BE	Best Effort
BS	Base Station
CSN	Connectivity Service Network
DHCP	Dynamic Host Control Protocol
DOCSIS	Data Over Cable Service Interface Specification
DSL	Digital Subscriber Line
ertPS	Extended Real-Time Polling Service
FDM	Frequency Division Multiplexing
FEC	Forward Error Correction
FFT	Fast Fourier Transform
FTP	File Transfer Protocol
GUI	Graphical User Interface
IEEE	Institute of Electrical and Electronics Engineers
IP	Internet Protocol
ISDN	Integrated Services Digital Network

ISI	Inter-Carrier Interference
MOS	Mean Opinion Score
MS	Mobile Station
NWG	Net Working Group
NLOS	Non Line-of-Sight
nrtPS	Non-Real-time Polling Service
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
OPNET	Optimized Network Engineering Tool
OSI	Open System Interconnection
PHY	Physical Layer
PSTN	Public Switched Telephone Network
QoS	Quality of Service
rtPS	Real-time Polling Service
SOFDMA	Scalable Orthogonal Frequency Division Multiple Access
SS	Subscriber Stations
TDMA	Time Division Multiple Access
UGS	Unsolicited Grant Service
VoIP	Voice over Internet Protocol
WiMAX	Worldwide Interoperability for Microwave Access
WMAN	Wireless Metropolitan Area Network

Chapter One

Introduction

Chapter One

Introduction

1.1 Preface

WiMAX stands for Worldwide Interoperability for Microwave Access. It is a wireless digital communications system, also known as IEEE 802.16 that is intended for wireless "Metropolitan Area Networks" [1]. It is one of the solutions of fourth generation (4G) wireless network which provides large network coverage and high data rates for IP networks that is capable of offering high Quality of Service (QoS). QoS is service that is used to measure the overall performance of network. Performance of network is measured to check the speed, accuracy and reliability. Different services are provided to different network traffic. Bandwidth, error ratio, delay, jitter and throughput are a different service that comes under QoS[2]. WiMAX defines five scheduling services for network applications: UGS, rtPS, nrtPS,ertPS and BE.

One of the important applications for the WiMAX is Voice over IP (VoIP) service. VoIP technology turns analog voice into digital data packets that can be transmitted over the internet. Before transmitting the analog voice signals they are compressed and encoded into digital voice streams with the help of codecs [3]. Codec is a voice/video encoding algorithm used to convert analog audio signal into compressed digital signals and vice versa.

In this project, the OPNET simulator is used to implement the VoIP over WiMAX network. The network performance metrics such as throughput, MOS, jitter and delay have been used to evaluate the performance of VoIP codecs.

1.2 Problem Statements

The demand of users for real time services like VoIP is increasing tremendously. To achieve high quality of service for these applications, the new technologies like WiMAX must satisfy all the quality parameters required by the service.So, it is very important to study the WiMAX standard in order to evaluate the performance of VoIP and the degree of QoSsatisfaction.

1.3 Proposed Solution

This research work will evaluate the performance of VoIP with different codecs and variety network scenarios in order to determine the achievement ofQoS parameters such as jitter, throughput, MOS, delay and end-to-end delay.

1.4 Aim and Objective

The main aim of this project is to evaluate the performance of VoIP over WiMAX network. The main objectives are:

- To simulate the WiMAX network model, using different scenarios addressing QoS service classes, number of users and voice codecs in order to investigate VoIP performance
- To evaluate the performance of QoS in scenarios which include jitter, MOS, end to end delay ,delay and throughput.

1.5 Methodology

In this research, Optimized Network Engineering Tool (OPNET) is used to implement and design the network models. Different scenarios are designed to address several aspects of the WiMAX network that affect VoIP traffic, such as: network capacity, QoS classes and voice codec. The considered parameters are evaluated, discussed, and then compared.theQoS classes and parameters that used in this simulation are Unsolicited Grant Service (UGS), real-time Polling Service (rtPS) and Best Effort (BE) service classes, and fiveQoS parameters have been selected, jitter, MOS, Packet end to end delay, delay and throughput.

1.6 Research Layout

The research consists of five chapters, their outlines are of follow.

Chapter one is an introduction that gives a background about the project, its aim and objectives, the problem statement, purposed solution and brief description on how to achieve the goals of project in methodology

Chapter Two provides a theoretical overview of mobile WiMAX networks and VoIP technology

Chapter Three describes the methodology, modeling, simulation parameters, implementation and results.

In Chapter Four the simulation results are gathered, statistically analyzed and graphically presented. The chapter also provides further discussion and analysis of the obtained results.

Chapter Five gives a conclusion of the work done along with recommendations for possible future work.

Chapter Two

Literature Review

Chapter Two

Literature Review

2.1 Background

2.1.1 Cellular Communications from 1G to 3G

Mobile communication technologies are often divided into generations, with 1G being the analog mobile radio systems of the 1980s, 2G the first digital mobile systems, 3G the first mobile systems handling broadband data [4]and 4G which designed primary for data. This continuing race of increasingsequence numbers of mobile system generations is in fact just a matter of labels, what is important is the actual system capabilities and how they have evolved [4].

2.1.2 WiMAXoverview:

WorldwideInteroperability for Microwave Access (WiMAX) technology is one of the solutions of fourth generation (4G) wireless network which provides high data rates for IPnetworks that is capable of offering high Quality of Service (QoS).

WiMAX is the wireless standard which covers area up to 50 km and provide 30 to 40 mbps data rate. It is the replacement for Digital Subscriber Line(DSL) and cables. WiMAX can be used for various applications like rural area broadband services, wireless backhaul, Wi-Fi hotspot, voice over Internet Protocol(VoIP), and video conferencing. Some salient features of WiMAX are Fixed WiMAX (802.16-2004) and Mobile WiMAX (802.16-2005) are two versions of WiMAX which allows mobility without losing connections[5].

2.1.3 Background on IEEE 802.16 and WiMAX

The IEEE 802.16 group was formed in 1998 to develop an air-interface standard for wireless broadband. The group's initial focus was the development of a LOS-based point-to-multipoint wireless broadband system for operation in the 10GHz–66GHz millimeter wave band. Theresulting standard—the original 802.16 standard, completed in December 2001—was based on a single-carrier physical (PHY) layer with a burst time division multiplexed (TDM) MAC layer. Many of the concepts related to the MAC layer were adapted for wireless from the popular cable modem Data Over Cable Service Interface Specification) DOCSIS standard [5].The IEEE 802.16 group subsequently produced 802.16a ,802.16-2004 and 802.16e-2005 [6].

2.1.4 Types of WiMAX

The IEEE 802.16 standard with specific revisionsaddresses two usage models:Fixed WiMAX and MobileWiMAX.

- **FixedWiMAX**

The fixed WiMAX supports only fixed applications. It is very robust against multi-path propagation because it uses Orthogonal Frequency Division Multiplexing (OFDM). Fixed WiMAX can work as a point-to-multipoint with the transmission data rate of 1 Mbps to 75 Mbps at a transmission distance over 50km. It operates in 3.5 GHz and 5.8 GHz spectrum bands[7].

- **Mobile WiMAX**

The mobile WiMAX is an amendment to fixed WiMAX by adding mobility. The mobile WiMAX supports mobile, portable, fixed and nomadic applications. Mobile WiMAX uses the Scalable OFDMA (SOFDMA) for improving the multipath performance in non-line-of-sight. Mobile WiMAX works on the licensed spectrum allocated in the 2.3 GHz, 2.5 GHz and 3.5 GHz frequency bandwidth [7].

2.1.5 WiMAX Architecture

The WiMAX NWG has developed a network reference model to serve as an architecture framework for WiMAXdeployments and to ensure interoperability among various WiMAXequipment and operators. Figure 2.1 shows a simplified illustration of an IP-based WiMAX network architecture.

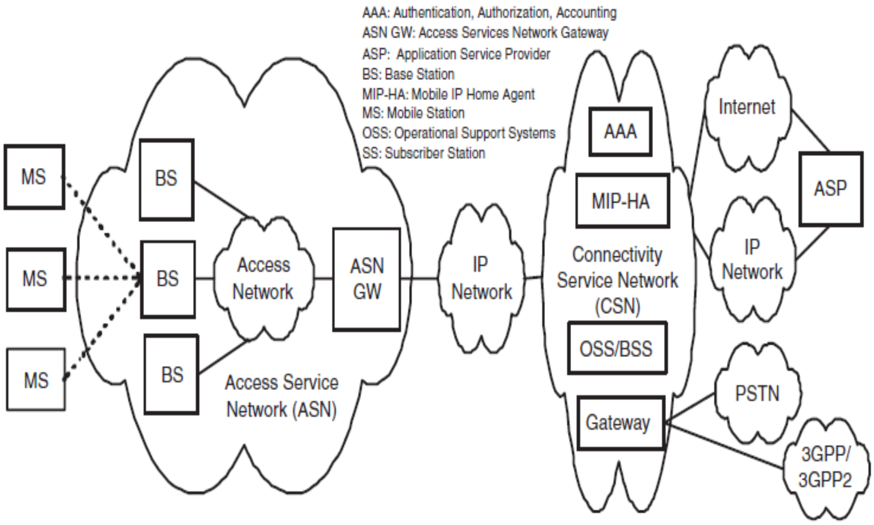

Figure 2-1: IP-Based WiMAX Network Architecture[6].

The WiMAX architecture consists of three logical entities:

Mobile Stations(**MSs**)

Used by the end user to access the network.

Access Service Network (ASN)

Comprises one or more Base Stations (BSs) and one or more ASN gateways (GWs) that form the radio access network (RAN) at the edge.

Connectivity Service Network (CSN)

Provides IP connectivity and all the IP core network functions[6].

Base station (BS)

Provides the air interface to the MS.In addition BS responsible for micro mobility management functions, radio resource management, QoSpolicy enforcement, traffic classification, Dynamic Host Control Protocol(DHCP) proxy, key management, session management, and multicast group management.

Access Service Network Gateway Acts as layer 2 traffic aggregation point within an ASN. In addition, ASN-GW performs AAA client functionality, establish and manage mobility tunnel with BSs, foreign agent functionality for mobile IP and outing towards selected Connectivity Service Network (CSN).

2.1.6 WiMAX Physical Layer

The WiMAXphysical layer provides air interface between the BS and the MS in different frequency bands for the entire range of WiMAX. The WiMAX physical layer is based on orthogonal frequency division multiplexing (OFDM) a scheme that offers good resistance to multipath, and allows WiMAX to operate in NLOS conditions.

2.1.7 Orthogonal frequency division multiplexing

OFDMtechnology provides operators with an efficient means to overcome the challenges of NLOS propagation. As presented in Figure 2-2 OFDM isbased on the traditional frequency division multiplexing(FDM), which enables simultaneous transmission ofmultiple signals by separating them into different frequencybands (subcarriers) and sending them in parallel[8].

OFDM is a multicarrier transporttechnology for high data rate communication system.The OFDM concept is based on spreading the high speed data to be transmitted over a large number of low rate carriers. The carriers are orthogonal to each other and frequency spacing between them are created by using the (FFT) [9]. The number of OFDM subcarriers can range from less than hundred to several thousand, with thesubcarrier spacing ranging from several hundred kHz down to a few kHz[4].

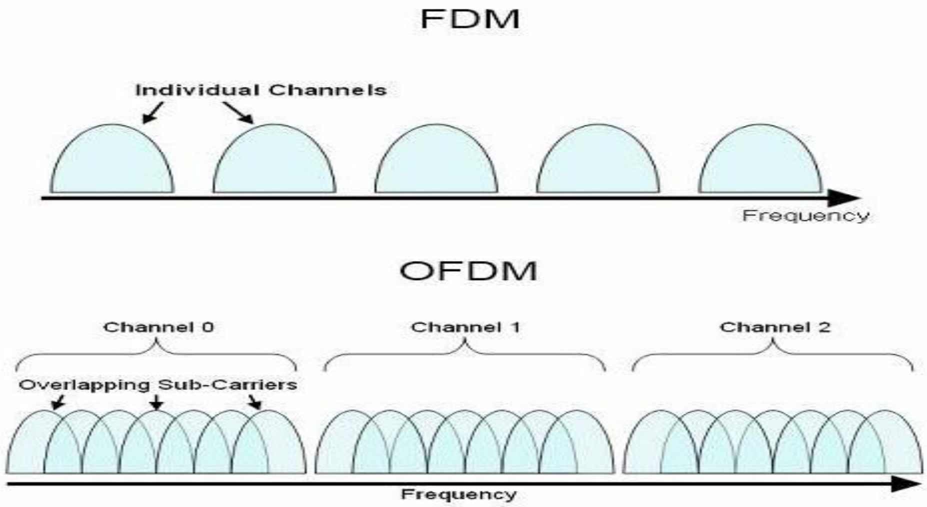

Figure 2-2: FDM and OFDM channels

OFDM is a spectrally efficient version of multicarrier modulation, where the subcarriers are selected such that they are all orthogonal to one another over the symbol duration, thereby avoiding the need to have non-overlapping subcarrier channels to eliminate inter-carrier interference[6].

In order to completely eliminate ISI, guard intervals are used between OFDM symbols. By making the guard interval larger than the expected multipath delay spread, ISI can be completely eliminated. Adding a guard interval, however, implies power wastage and a decrease in bandwidth efficiency. The amount of power wasted depends on how large a fraction of the OFDM symbol duration the guard time is [6].

Figure 2-3 illustrateOFDM Signal Representation in Frequency and Time Domain. Instead of transmitting all the information on the single RF carrier signal, the high data rate input stream is multiplexed into parallel combination of low data rate streams. The parallel streams are modulated onto separate subcarriers in the frequency domain through the use of inverse fast Fourier transform (IFFT) and transmitted through the channel. At the receiver, the signal is demodulated using Fast Fourier Transform (FFT) process to convert a time varying complex waveform back to its spectral components, recovering the initial subcarriers with their modulation and thus the original digital bits stream [10].

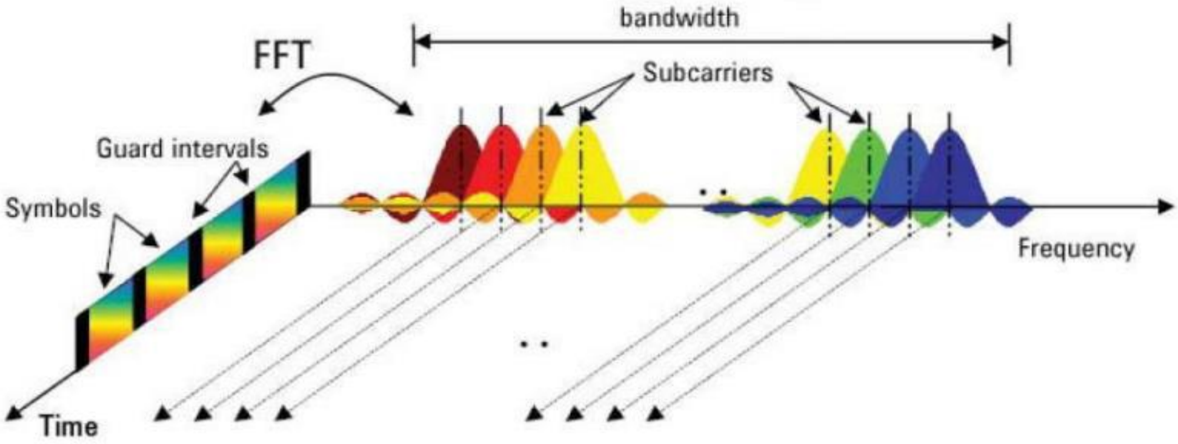

Figure 2-3: OFDM Signal Representation in Frequency and Time Domain[10]

2.1.8 Orthogonal Frequency Division multiple Access

Orthogonal Frequency Division multiple Access (OFDMA) is derived from Orthogonal Frequency Division Multiplexing (OFDM). In OFDMA subcarriers and time slots are allotted to different users. By using Time Division Multiple Access (TDMA) with basic Orthogonal Frequency Division Multiplexing (OFDM).

OFDMA is achieved thus allowing dynamic allocation of subcarriers among different users on thechannel. OFDMA provides a robust system with increased capacity and resistance to multipath fading.

The Figure 2-4 shows OFDMA transmitting a series of QPSK data symbols.In WiMAX, each subcarrier is modulated with a conventional modulation scheme depending on the channel condition. WiMAX uses BPSK,QPSK,16QAM or 64QAM for modulation at a low symbol rate. The FFT sizes of 128,256,512,1024 and 2048corresponding to WiMAX.In time domain, guard intervals known as cyclic prefix (CP) are inserted between each of the symbols to prevent inter-symbol interference at the receiver caused by multi-path delay spread in the radio channel. The CP is copy of the end of the symbol inserted at the beginning.

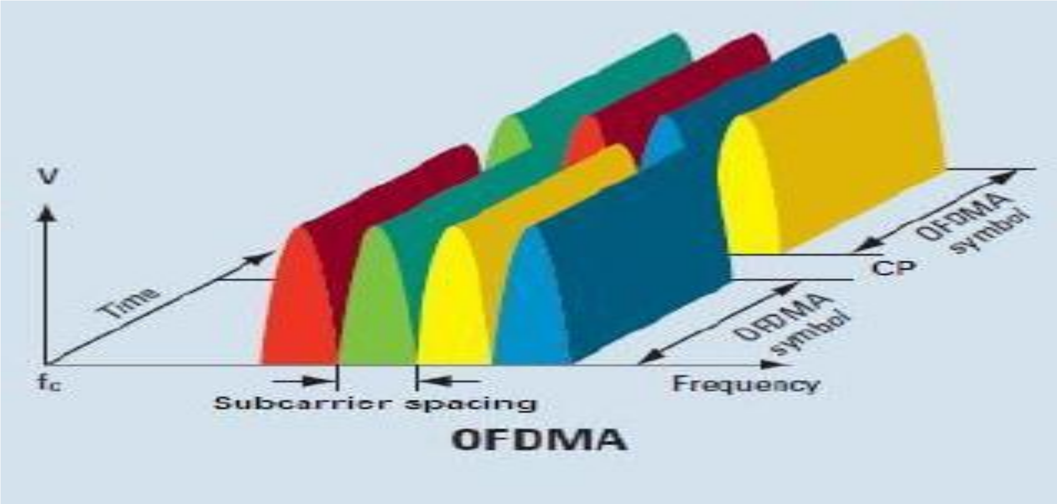

Figure 2-4: OFDMA Transmitting a Series of Data Symbols[10]

2.1.9 Adaptive and Modulation Coding AMC

WiMAX supports a number of modulation and forward error correction (FEC) coding schemes and allows the scheme to be changed on a per user and per frame basis, based on channel conditions. AMC is an effective mechanism to maximize throughput in a time-varying channel [6].

The basic idea is Transmit as high a data rate as possible when the channel is good, and transmit at a lower rate when the channel is poor, in order to avoid excessive dropped packets[6]. Lower data rates are achieved by using a small constellation, such as QPSK, and low-rate error-correcting codes, such as rate 1/2 convolutional. The higher data rates are achieved with large constellations, such as 64 QAM, and less robust errorcorrecting codes; for example, rate 3/4 convolutional. The Table 2-1 shows Modulation and Coding supported in WiMAX.

Table 2-1: Modulation and Coding Supported in WiMAX

	Downlink	**Uplink**
Modulation	BPSK, QPSK, 16 QAM, 64 QAM; BPSKoptional for OFDMA-PHY.	BPSK, QPSK, 16 QAM; 64 QAM optional.
Coding	Mandatory: convolutional codes at rate1/2, 2/3, 3/4, 5/6. Optional: convolutional turbo codes at rate1/2, 2/3, 3/4, 5/6; repetition codes at rate1/2, 1/3, 1/6, LDPC, RS-Codes forOFDM-PHY.	Mandatory: convolutional codes at rate 1/2, 2/3,3/4, 5/6 Optional: convolutional turbo codes at rate 1/2,2/3, 3/4, 5/6; repetition codes at rate 1/2, 1/3, 1/6, LDPC.

2.1.10 Quality of Service

QoS is a wide and unbound term that refers to the "collective effect of service," as obtained by the subscriber. QoS more hardly means meeting specific necessity basically, throughput, packet delivery ratio, packet error rate, network load, jitter, and delay related with a provided application. WiMAX networks must support a diversity of applications,i.e. voice, video, data and multimedia and each of these has various traffic patterns and QoS requirements. The QoS is granted based on application and service type under consideration[3].

2.1.10.1 Quality of Service Classes

To support a wide variety of applications, WiMAX defines five scheduling services(Table 2.2) that should be supported by the base station MAC scheduler for data transport over a connection.

- Unsolicited Grant Service UGSis design to support real time service flow which produces the static size data packet in a periodic way[11].
- Real time polling service (rtPS) is design to support real time facilities which normally creates variable size data packets in a periodic way.
- Non-Real time polling service (nrtPS) designed to support non-real-time and delay tolerant services that require variable size data grant burst types on a regular basis such as FTP [12].
- Extended Real Time Polling Services (ertPS)algorithm is supports features of UGS with variable-size data packets.
- Best Effort (BE)service supports data streams for which no minimum service level is required and which may therefore be handled on a space-available basis[13].

Table 2-2Service Flows Supported in WiMAX

Service Flow Designation	Defining QoS Parameters	Application Examples
Unsolicited grant services (UGS)	Maximum sustained rate Maximum latency tolerance Jitter tolerance	Voice over IP (VoIP) without silence suppression
Real-time Polling service (rtPS)	Minimum reserved rate Maximum sustained rate Maximum latency tolerance Traffic priority	Streaming audio and video, MPEG (Motion Picture Experts Group) encoded
Non-real-time Polling service (nrtPS)	Minimum reserved rate Maximum sustained rate Traffic priority	File Transfer Protocol (FTP)
Best-effort service (BE)	Maximum sustained rate Traffic priority	Web browsing, data transfer
Extended real-time Polling service (ErtPS)	Minimum reserved rate Maximum sustained rate Maximum latency	VoIP with silence suppression

2.1.10.2 Quality of Service parameters

The QoS is measured by performance metrics such as Mean Opinion Score (MOS), end-to-end delay, jitter and throughput.

- Jitter is the measure of the variation in arrival time of consecutive packets.It can be calculated using equation2-1:

$$\boldsymbol{jitter = Max1 \leq ii \leq n\{[t'(n) - t'(n-1)] - [t(n) - t(n-1)]\}} \quad \textbf{(2-1)}$$

- MOS provides a numerical measure of the quality of human speech in voice telecommunications, with value ranging from 1 to 5 where 1 is the worst quality and 5 is the best quality[13].
- Delayis the time taken by packet to go across the network.
- Packet End to End Delayis characterized as the amount of time it takes for speech to exit the speaker's mouth and reach the listener's ear. The total voice packet delay is computed as:

$$De2e = D_n + D_e + D_d + D_c + D_{de} \qquad \textbf{(2-2)}$$

  Where D_n, D_e, D_d, D_c, D_{de} show the network, encoding, decoding, compression and decompression delay, respectively[13].
- Throughputpresents the amount of successful data transferred from one location to another over a specific period of time.

$$Throughput = \frac{Total\ Bytes\ Received*8\ \ (bit)}{total\ duration\ of\ the\ simulation} \qquad \textbf{(2-3)}$$

2.1.11 VoIP

VoIP, known as IP Telephony, is the real-timetransmission of voice signals using the Internet Protocol(IP) over the public Internet or a private data network[14]. Itconverts the analog voice signal from a telephone or computer into a digital packetized signal that can be transmitted over the internet. Before transmitting the analog voice signals they are compressed and encoded into digital voice streams with the help of codecs. VoIP system divided into three indispensable components, namely codec, packetizer and the playout buffer[3]. The general overview of VoIParchitecture is shown as in Figure 2-5:

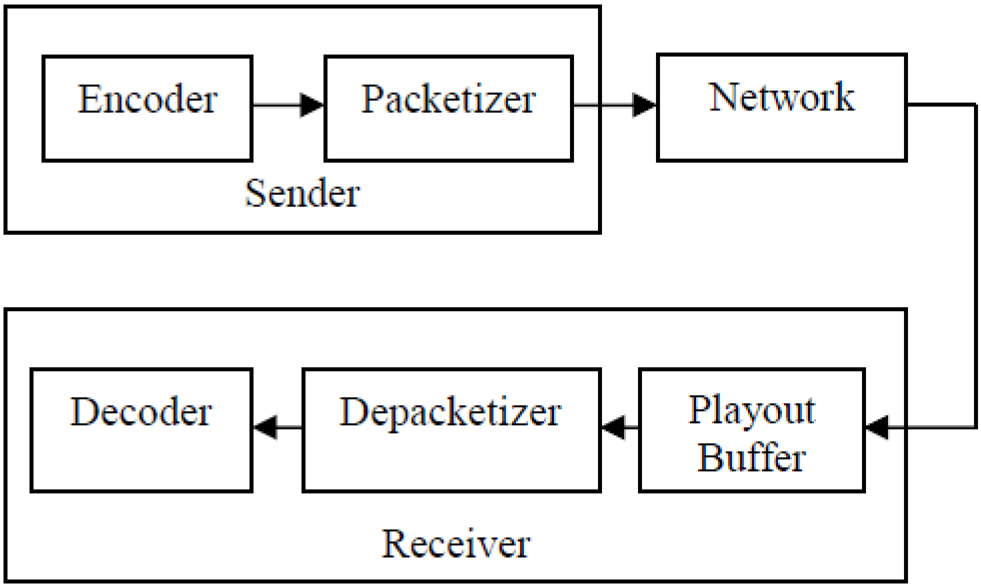

Figure 2-5: VoIP Architecture.

- **Codec**: The function of codec is to compress and encode the analog signals into digital voice signals. Codec provide good quality of voice even after compression, with minimum delay which is one of the main advantages of using codec.
- **Packetizer**: With the help of packetizer output digital streams are packed into constant bit rate voice packets.
- **Playoutbuffer**: A two way conversation is very sensitive to packet delay jitter so to eliminate the delay,playout buffer is used at receiver end [22].

2.1.11.1 Voice codecs

Codec is a voice/video encoding algorithm through which process of compression is carried out that permit the call transmission over the IP network [14].A Voice Codec used at the subscriber side to convert theanalogue voice waves to digital pulses and vice versa. There are different codecs types based on the selected sampling rate, data rate, and implemented compression algorithm. Most common codecs: G.711, G.723.1 and G.729A.

G.711 is a standard codec with high bit rate about 64 Kbps. This codec is used by the Public Switched Telephone Network (PSTN) and Integrated

Services Digital Network (ISDN) lines and it gives better voice quality when used under VoIP application. G.711 employs logarithmic compression that compresses each 16 bit sample to 8 bit samples.

G.723 is one of the most efficient and licensed codec with highest compression ratio. This codec is widely used in applications such as audio, video, fax, telephony, speech and also used in VoIP applications.

G.729 This codec has low bandwidth requirements but provides good audio quality. The applications of this codec are fax, telephony, VoIP, Fax over Internet Protocol (FoIP), voice compression, imaging and video software for various applications. [7]

2.2 Related works

In[15]distinguished the state of heterogeneity access within a city area by testing both conditions: LOS and NLOS. So by evaluate the quality of experience and network QoS parameters, and collecting the obtained results display the correct QoS service classes' management on the numerous well served users of VOIP.

In [16] distinguished that the capability of voice in WIMAX Networks is filling up with very good latency and jitter and MOS scale with 4.0. His work also express that bandwidth does not affect the jitter in the network. Also expressed that both latency and jitter values for WIMAX was not high in compared to broadband and network.In [17]analyzed quality of service modules mapping for two of the most widely used cases of hybrid networks WiFi+WiMAX and WiFi+LTE. Typically users or network providers will presume that real time polling service (rtPS) quality of service will give the very best throughput or performance compared with the Best Effort (BE). This is due to the theoretical characteristic of the rtPSQoS which is designed to support real-time service flow. In contrast, BE QoS is designed for non-real-time applications where no service guarantees is provided and therefore control services on a best available basis.

Chapter Three

Simulation Setup

Chapter Three

Simulation Setup

In this project, OPNET Modeler 14.5 is used to simulate 5 different WiMAX network topologies (light network size with background application, light network size without background application, high network size with background application and high network size with background application). Any topology has contained three scenarios (UGS scenario, ertPS scenario and BE scenario). The WiMAX model and its corresponding object palette are used. The parameters that will be tested: jitter, throughput, MOS and delay of UGS, ertPS and BE service classes to VoIP system over WiMAX traffics.

The best service class is tested using G.711 and G.729 voice codec's.

3.1 OPNET

OPNET simulator is a tool to simulate the behavior and performance of any type of network. The main difference with other simulators lies in its power and versatility. This simulator makes possible working with OSI model, from layer 7 to the modification of the most essential physical parameters.

OPNET also includes features such as comprehensive library of network protocols and models, user friendly GUI (Graphical User Interface), Web report is feature allows you to organize and distribute the results of your simulations in form graphical results and statistics [7]

The OPNET MODELLER provides:

- Better understanding of the component of the network.
- Live behavior analysis of the specific network.

- Continuous monitoring and testing of the network.
- Visualization of traffic in real time.

3.2 WiMAX Network Model

The WiMAX network model consists of one cell and an IP backbone. Table 3-1 presents the general parameters of the WiMAX network model.

Table 3-1:General Simulation Parameters

Cell Radius	3 kilometers
Number of cells	1
Base Station Power	20 Watt
Mobile Station Power	0.2 Watt
Number of MSs	7 & 25
QoS classes	UGS, rtPs and BE
Voice codec	G.711, G.723.1 and G.729A
Simulation time	600 sec

3.2.1 Light Network size

Figure 3-1 shows the light network size which is contained seven Subscriber Stations (SSs), one BSs, router, and server.

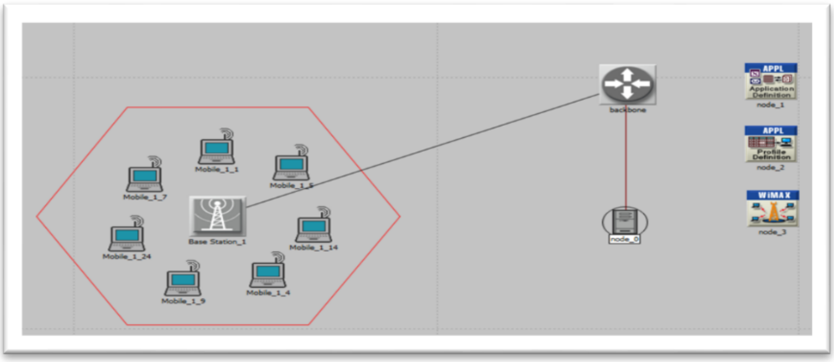

Figure 3-1: WiMAX Light Network Size

3.2.2 High Network Size

Figure 3-2 shows the high network size which is contained 25 Subscriber Stations (SSs), one BSs, router, and server.

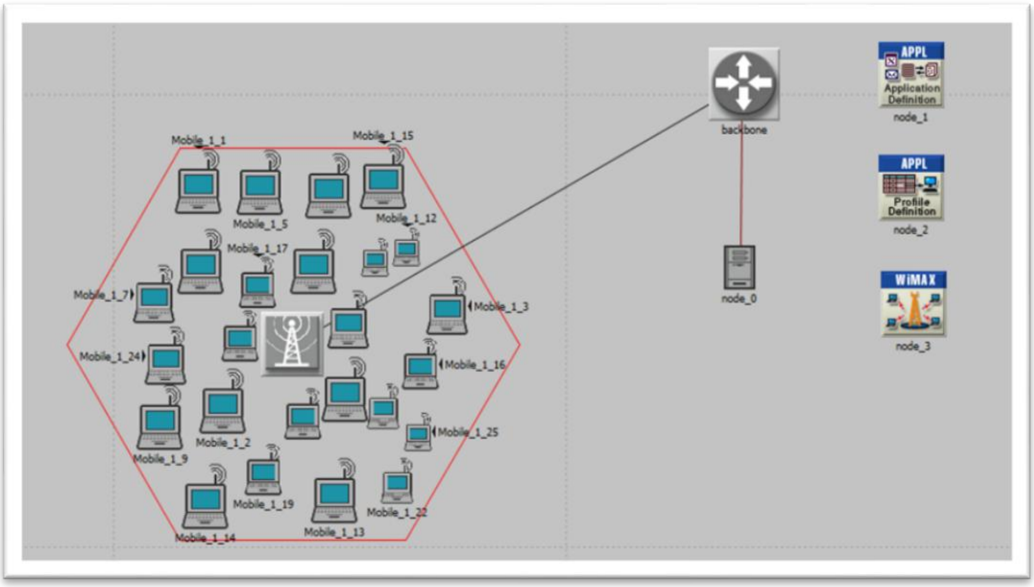

Figure 3-2: WiMAX high Network Size

3.3 Traffics Model

For all scenarios, a VoIP traffic model had to be created, the Application, and Profile had to be configured.

3.3.1 Application configurations:

Application configuration object is an object used to define and configure all Applications in the network according to the user requirements. OPNET has most common applications like: HTTP, E-mail, video, File transfer, Voice, database [18].

In the first case, one application named VoIP was created in the Application definition node. In the second case, two applications named VoIP and Video were created in the Application definition node.

The settings are shown in Figure 3-3.

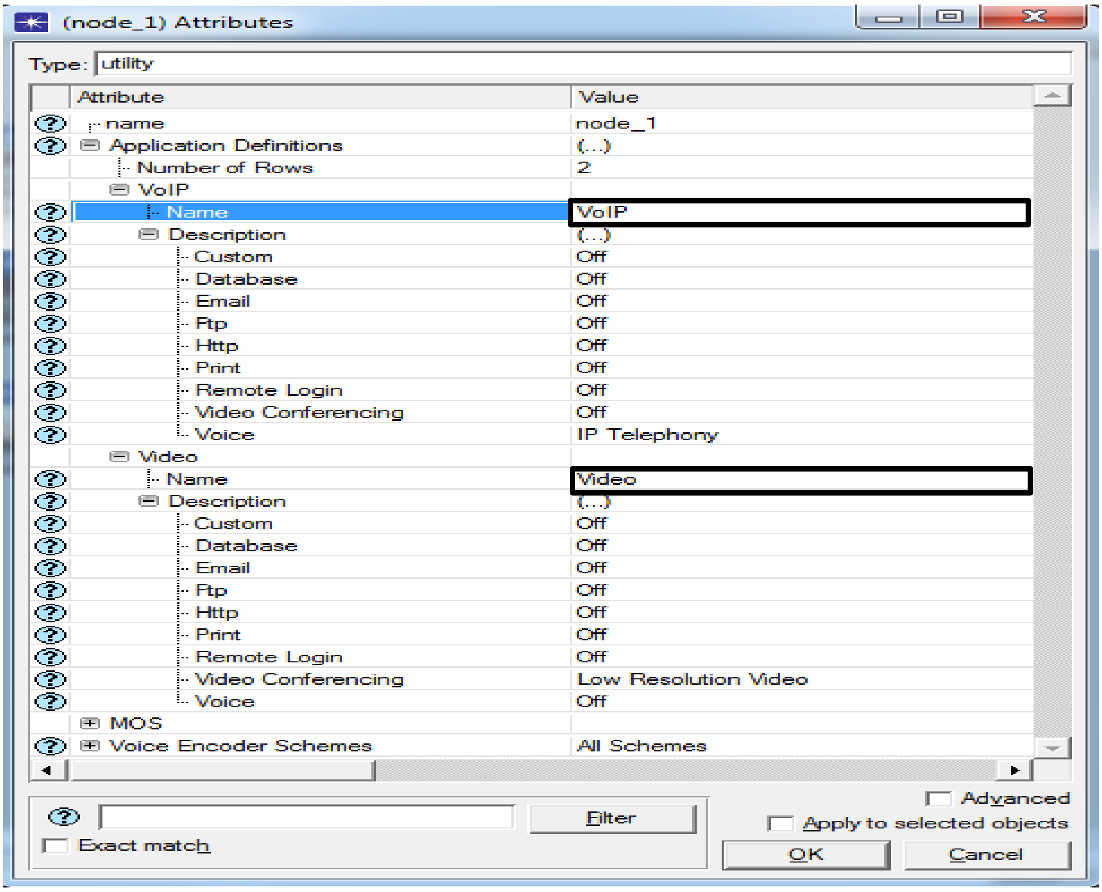

Figure 3-3: Application Attributes

3.3.2 Profile Configurations

Profile Configuration Objects an object that can be used to create users profiles, profile can contains one or more applications and each application can be configured by the starting time and ending time [18]. In case (1), one profile called VoIP was created in the Profile definition node. In case (2), Profiles called VoIP PRO and Video PRO were created in the Profile definition node. Figure 3-4 shows the Profile Configuration menu.

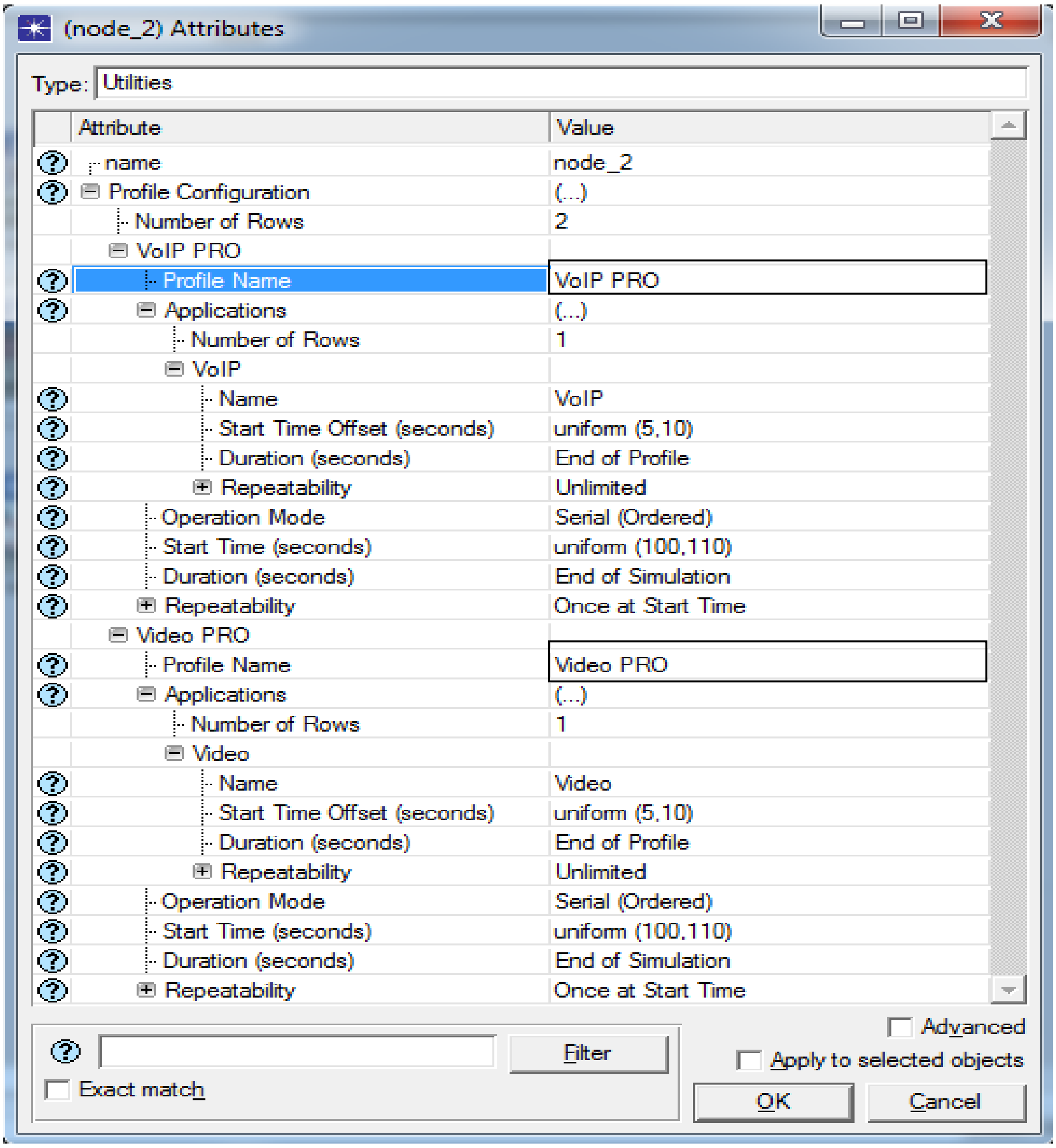

Figure 3-4: Profile Attributes

3.4 WiMAX Configuration

The parameters that need to be modified are the MAC Service Class Definitions and the Efficiency Mode. The MAC Service Class Definitions allows to configure parameters with the Quality of Service (QoS) requirements. The Efficiency Mode is set to efficiency enabled in order to study and evaluate the QoS parameters. The configuration can be seen in Figure 3-5.

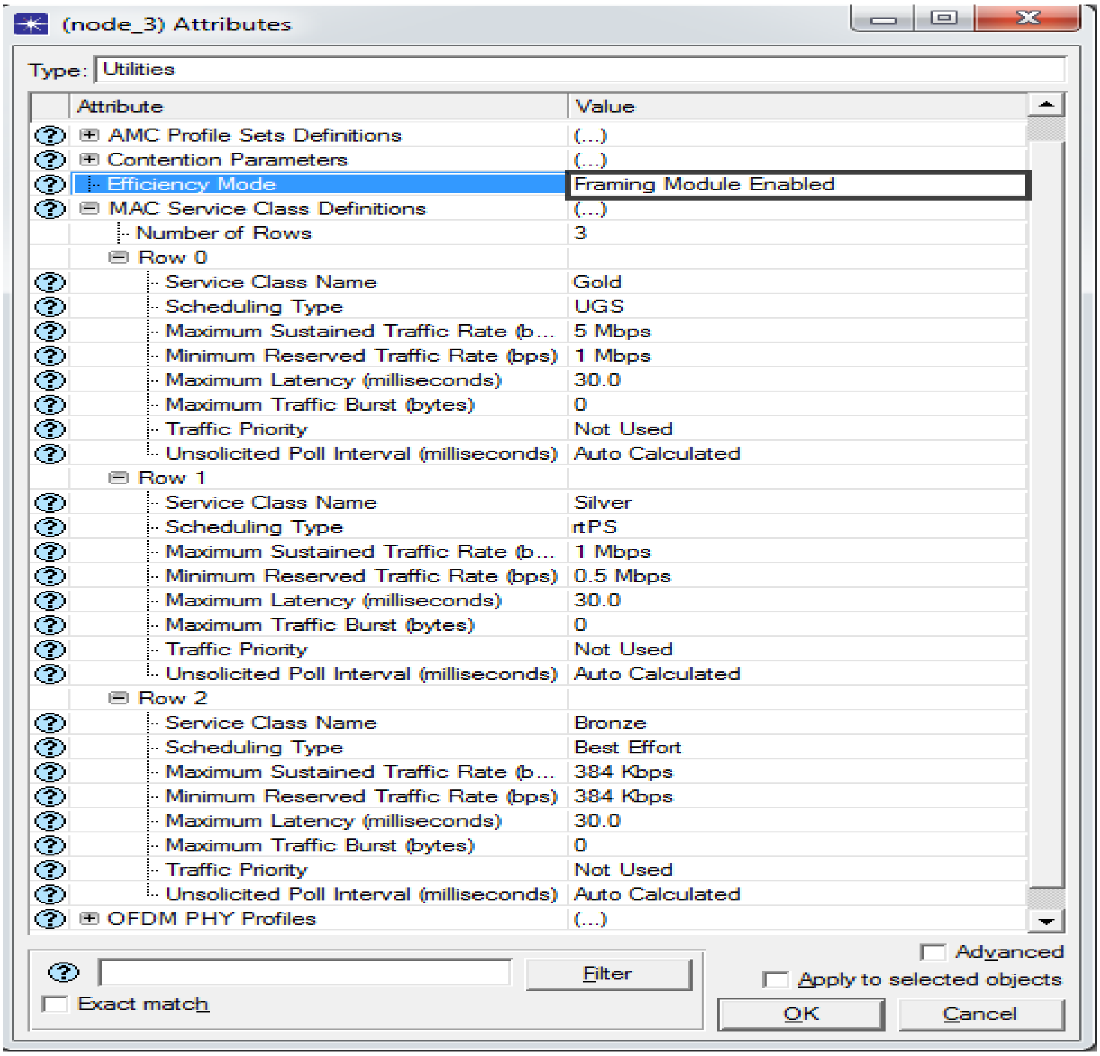

Figure 3-5: WiMAX Configuration

3.4.1 WiMAX Base Station

There are typical values that are commonly used in real life situations. In case (1), one row is added. In case (2), Two rows are added in to represent main and background Traffic (i.e. VoIP, video) as shown in Figure 3-6 Interactive Voice, Background respectively. Service Class name for VoIP is set to Gold. After that BS and MS will be adjusted.

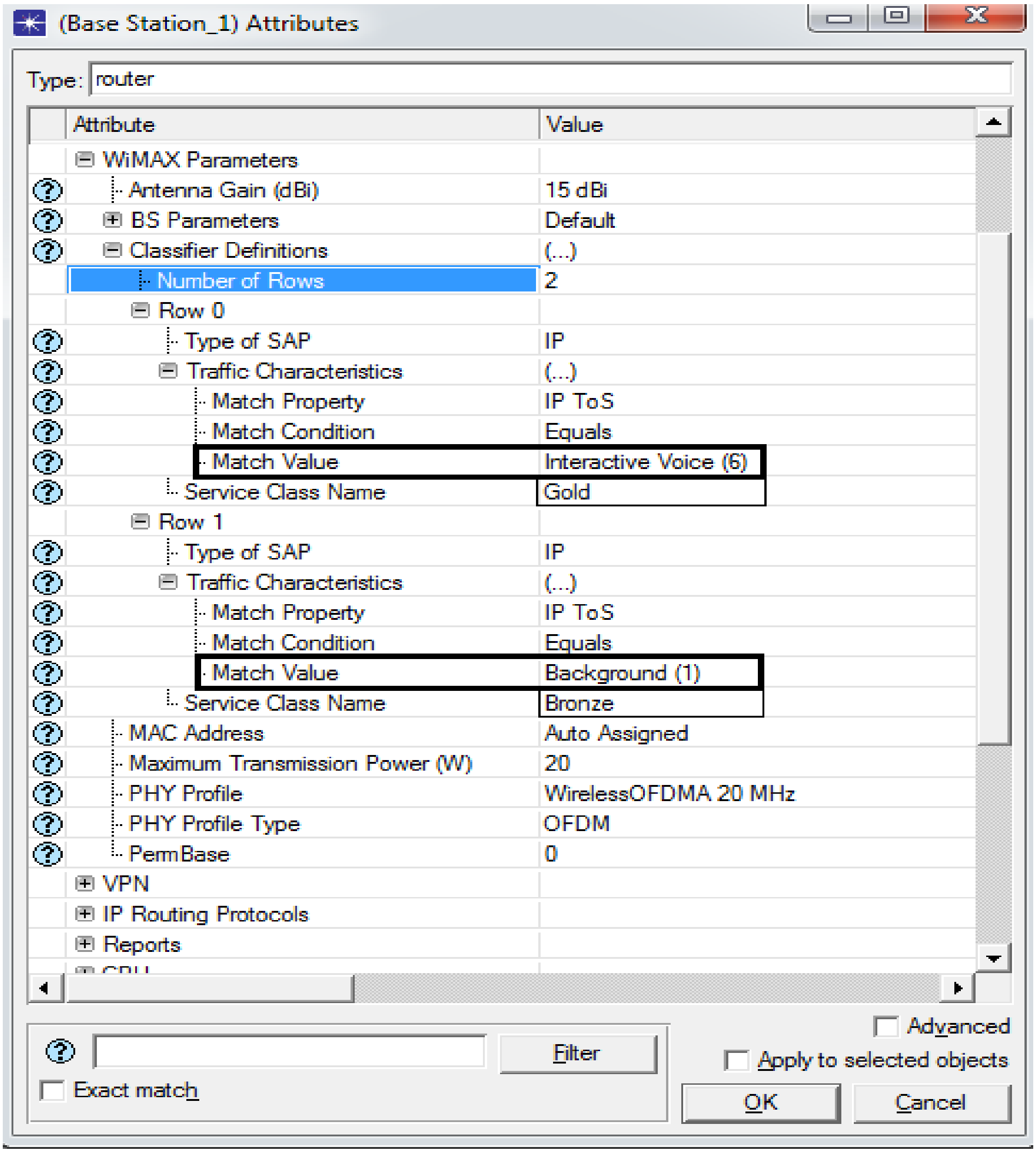

Figure 3-6: WiMAX Base Station (BS) Parameters Configuration

3.4.2 Mobile Node

The key parameters that were modified are WiMAX parameters and applications. In case (1) one row is added to WiMAX parameters. In case(2) 2 rows are added to represent main and background Traffic (i.e.

VoIP, Video) as shown in Figure 3-7 Interactive Voice, Background respectively. Service class name for VoIP is set VoIP Gold.

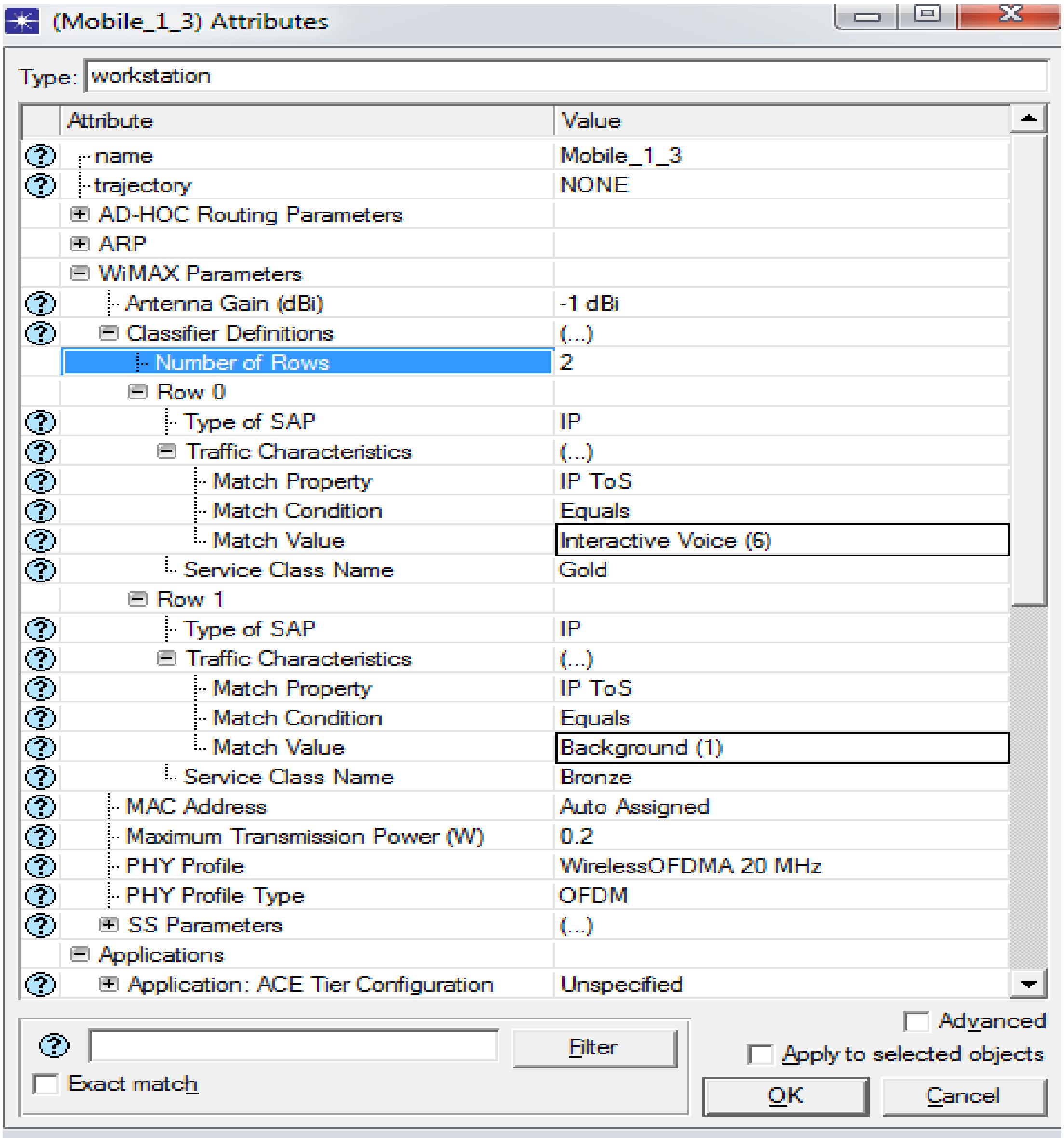

Figure 3-7: WiMAX Mobile Node - WiMAX Parameters Configuration

Figure 3-8 shows the sitting of WiMAX Application Parameters, VoIP and video profiles are added to the supported profiles section and applications are added to the supported services section

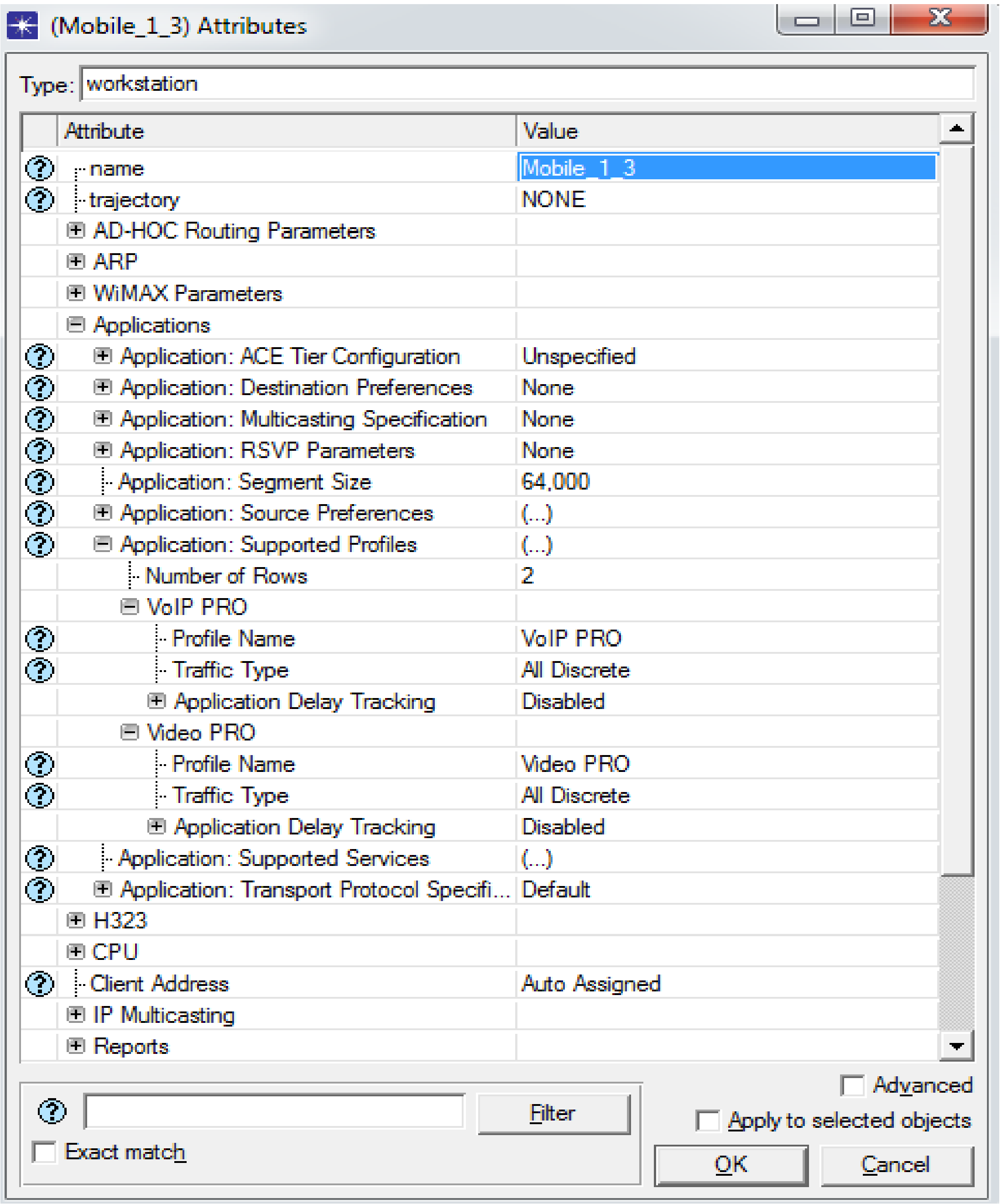

Figure 3-8WiMAX Mobile Node: Application Parameters Configurations

Finally, the simulation is run for 10 minutes.

Chapter Four

Results and Discussion

Chapter Four

Results and Discussion

As explained in the previous chapter, 4 different network topologies have been deployed to study the performance of VoIP over mobile WiMAX networks. Several scenarios have been performed for each network topology. The simulation results are: throughput, delay, end_to_end delay, jitter and MOS. Results are studied, graphically presented and discussed in the following sections of this chapter.

4.1 Network Topology without background application

This topology has 7 SSs, it was simulated to study the effect of different QoS classes on VoIP services over WiMAX network. The QoS classes used for the investigation include UGS, rtPS and BE.

4.1.1 Jitter of topology without background application

Jitter is one of the main parameters for indicating the perceived quality of voice as it reaches the destination node. Figure 4-1 shows the average jitter for all the three service classes. UGS service class has the lowest value of jitter of (-0.000008) sec. BE service class has the highest jitter of (-0.000021) sec. It is also observed that rtPS has low jitter values and falls very close to UGS. The negative value of jitter means that the time difference between the packets at the destination is less than that at the source.

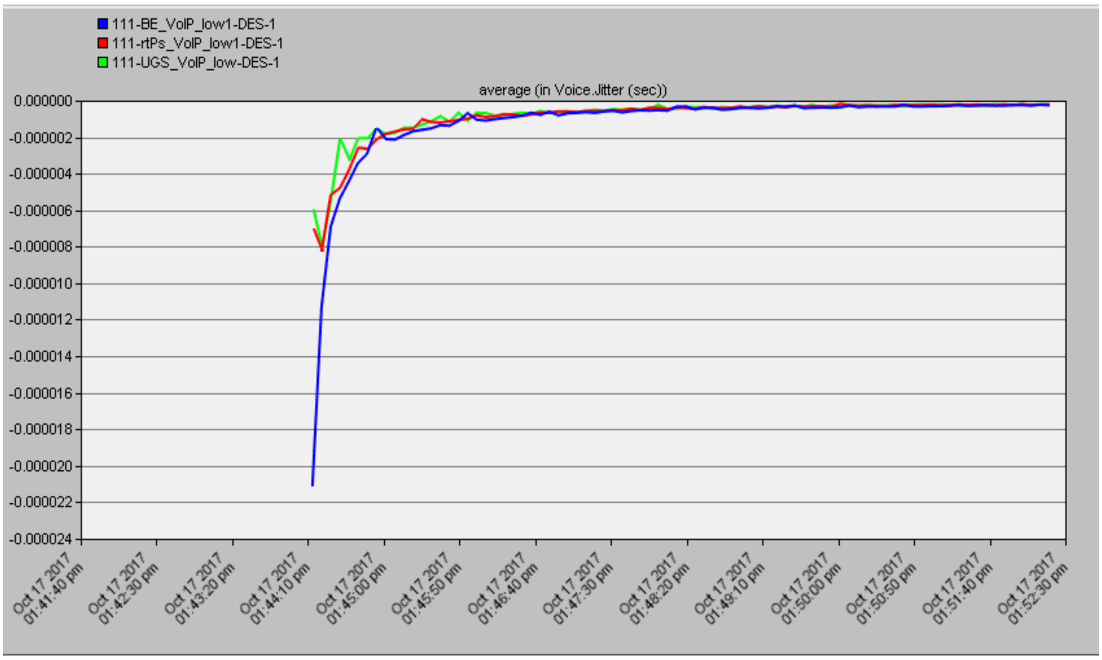

Figure 4-1: jitter of topology without background application

4.1.2 MOS of topology without background application

The MOS values that shown in Figurer 4-2 have the same values for all the service flows.

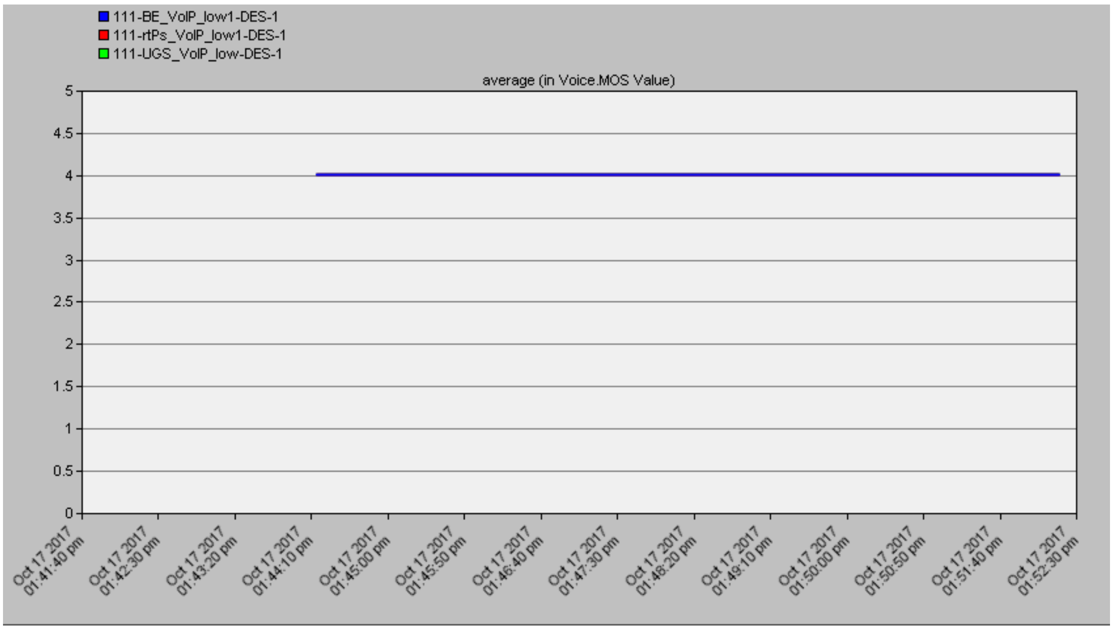

Figure 4-2: MOS of topology without background

4.1.3 End_to_end delay of topology without background application

Figure 4-3 shows the end-to-end delay for the three service classes. The values for UGS and BE service classes fall very close to each other and didn't exceed 0.09s. On the other hand, the rtPs service class has the lowest delay.

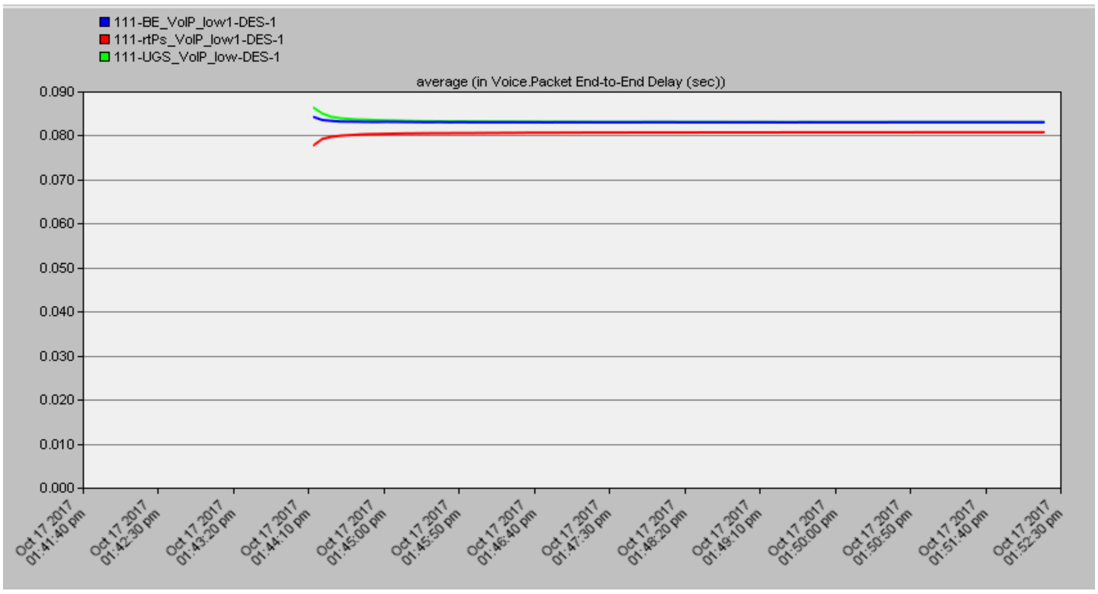

Figure 4-3: packet end to end delay for different service classes

4.1.4 Delay of topology without background application

Figure 4-4 Compares delay when UGS, rtPS and BE are applied. UGS class has the lower delay of approximately 0.0131 sec and rtPS with the higher value of 0.015 sec and BE with 0.0138 sec.

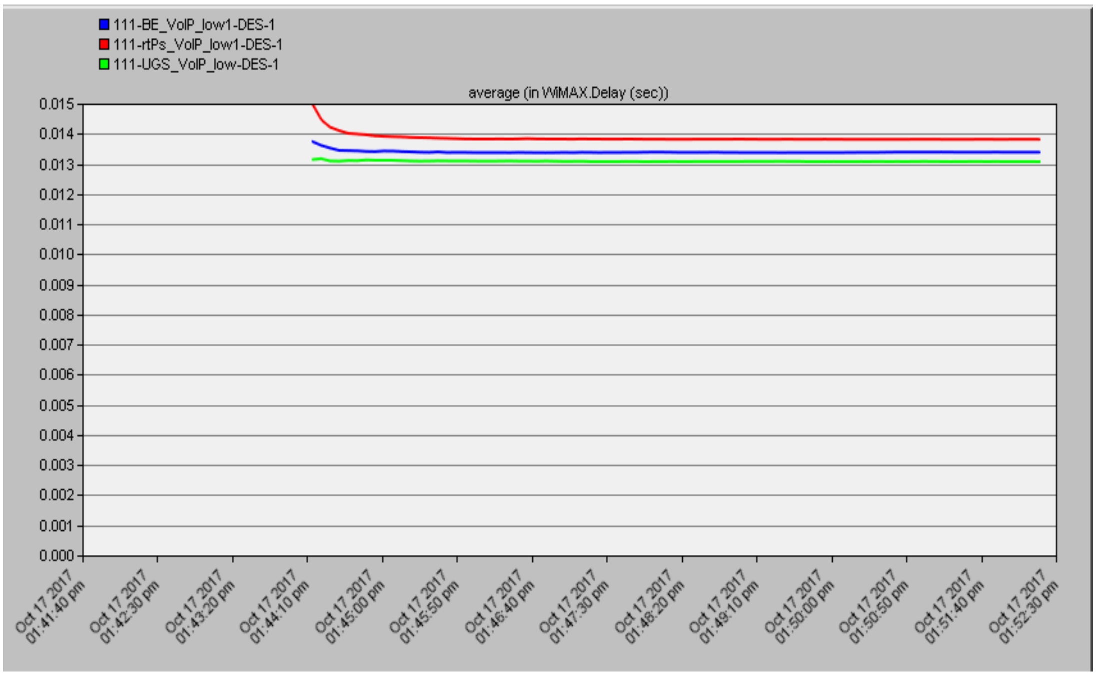

Figure 4-4: Delay of topology without background application

4.1.5 Throughput of topology without background application

Figure 4-5 presents the throughput for all three service classes. The UGS class has the highest throughput. This due to constant bit rate traffic of UGS. The throughput for rtPS and BE service classes fall very close to each other.

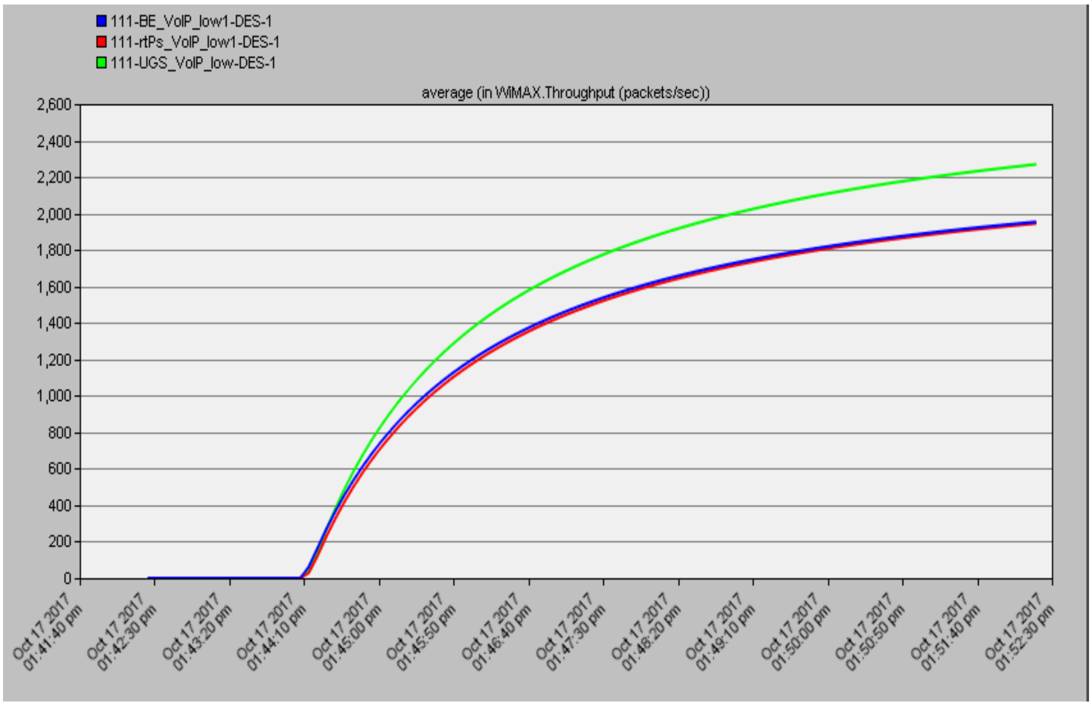

Figure 4-5: Throughput without background application

4.2 Network Topology with background application: light size

This topology has 7 MSs, it was simulated to study the effect of different QoS classes on VoIP services over WiMAX network. The QoS classes used for the investigation include UGS, rtPS and BE. The amount of video background traffic is sent, and later VoIP results obtained from the simulation is shown below. Results obtained from the simulation is shown below.

4.2.1 Jitter of topology with background application: light size

Figure 4-6 shows the average jitter the three service classes. UGS service class has the lowest jitter. BE service class has the highest jitter. It is also observed that rtPS has low jitter value and falls very close to UGS.

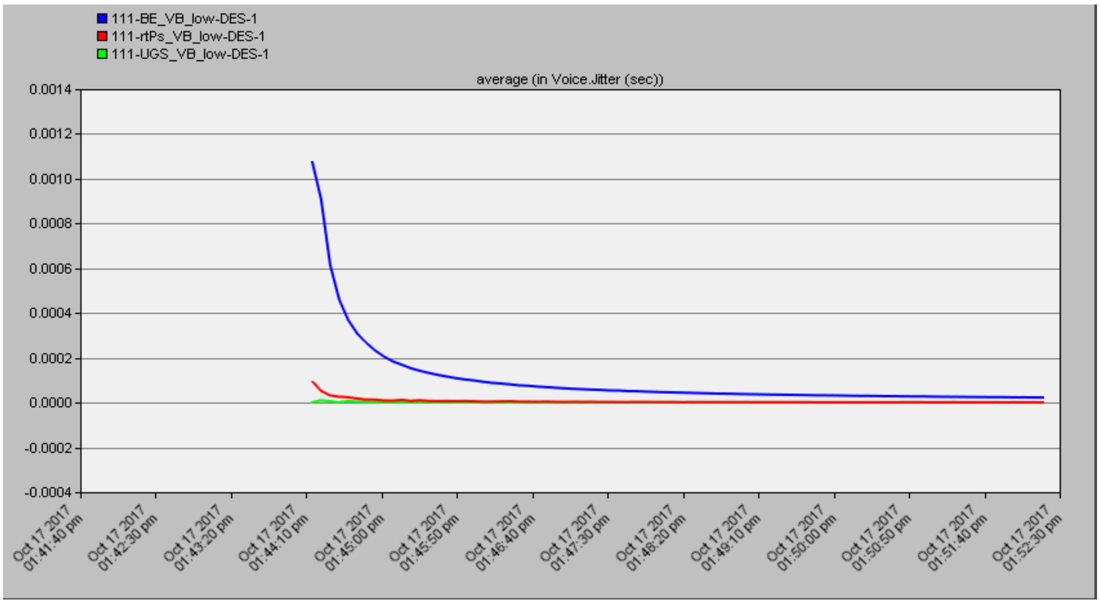

Figure 4-6: jitter with background application: light size

4.2.2 MOS of topology with background application: light size

Figure 4-7 shows the MOS values. UGS and rtPs scored the highest values and they are the same and the MOS values for BE.

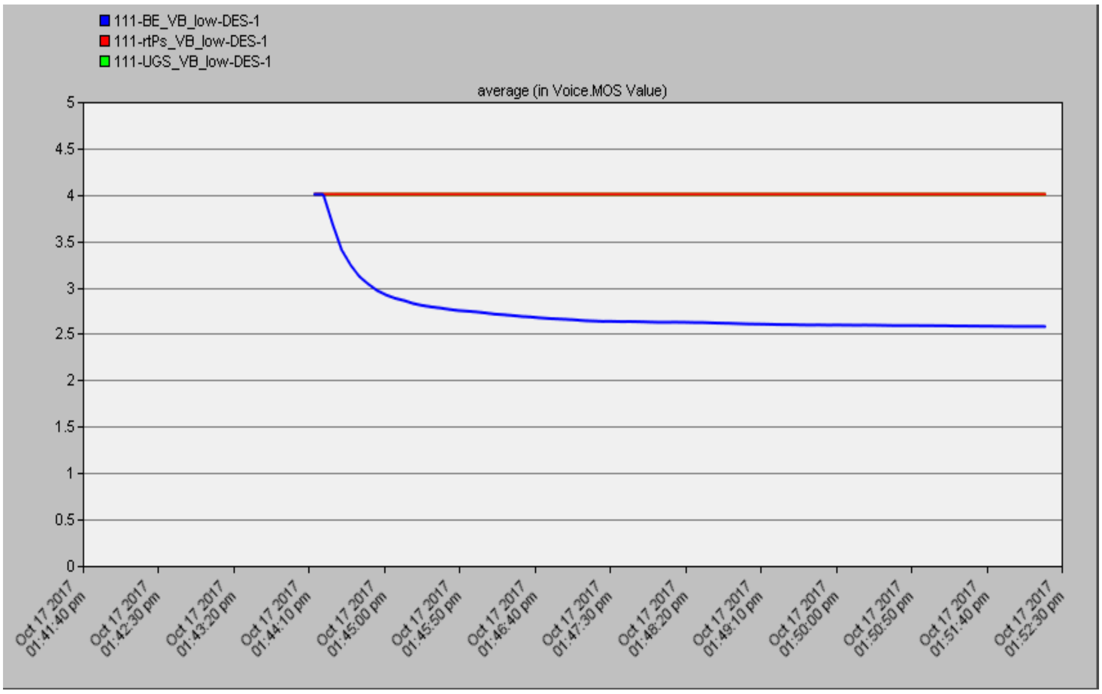

Figure 4-7: MOS with background application: light size

4.2.3 End-to-end delay of topology with background application: light size

Figure 4-8 shows the end-to-end delay. UGS has the best end to end delay that doesn't exceed 0.19s. BE has the highest delay.

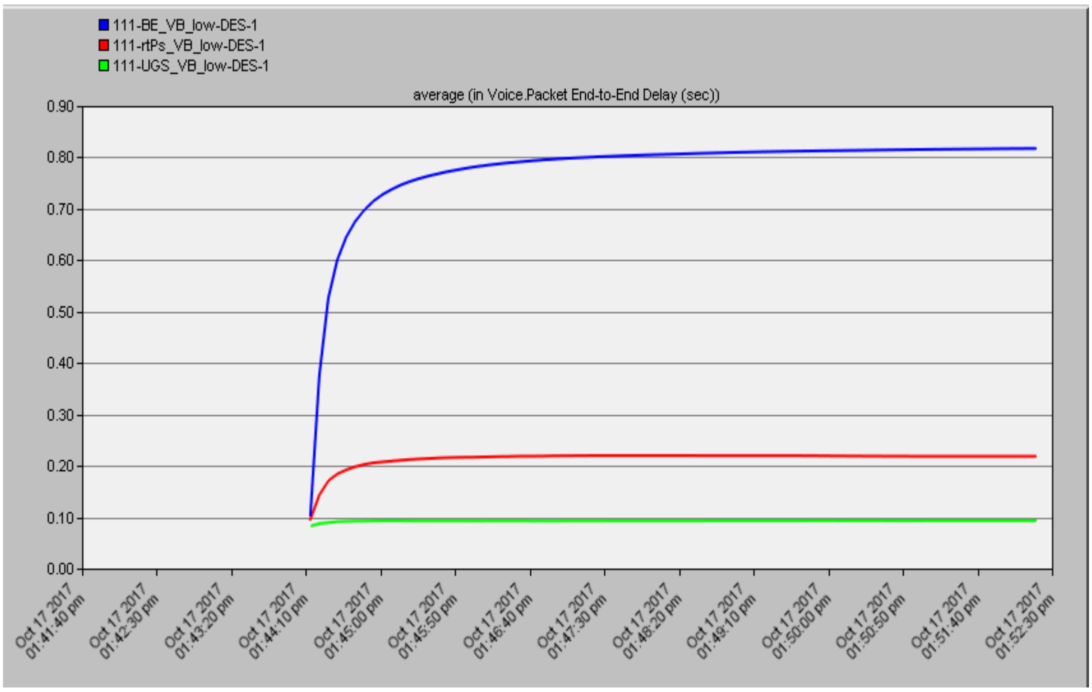

Figure 4-8: packet end_to_end delay with background application: light size

4.2.4 Delay of topology with background application: light size

Figure 4-9 Compares delays. UGS class returned the lower delay of approximately 0.025 sec, rtPs has the higher value of 0.11 sec and BE with the highest value which approximately 0.65 sec.

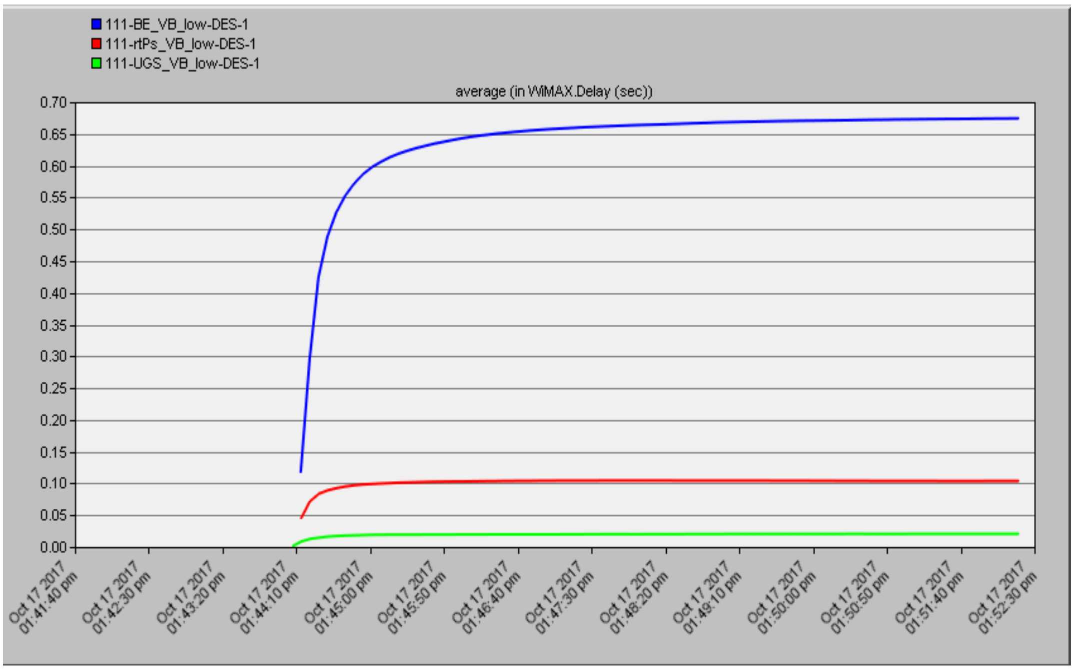

Figure 4-9: delay with background application: light size

4.2.5 Throughput of topology with background application: light size

Figure 4-10 shows the throughput. The throughput of UGS flow is the highest among the three. The reason for this is that UGS service class is designed with constant bit rate traffic. The throughput for rtPS is lower than UGS, and BE has the lowest value.

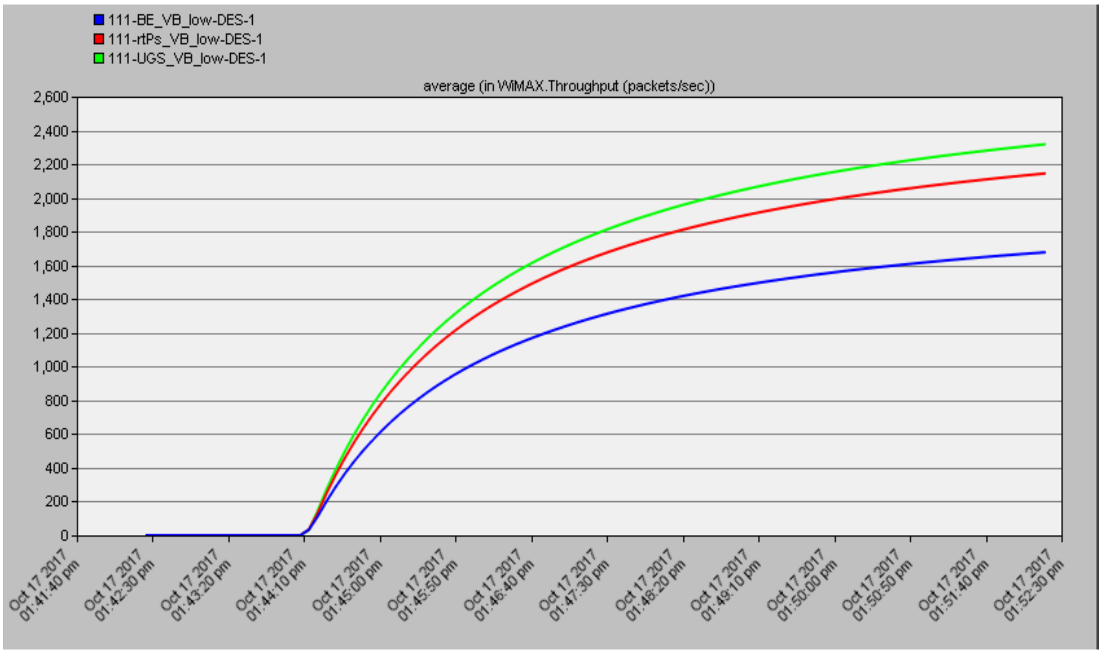

Figure 4-10: Throughput with background application: light size

4.3 Network Topology with background application: High size

This topology has 25 MSs and one BS, it was simulated to study the effect of different voice codecs over WiMAX networks. The voice codecs that used in this simulation are G.711, G.723.1 and G.729A. And amount of video background traffic is sent.

4.3.1 Jitter of Topology with background application: High size

Figure 4-11 shows the average jitter. UGS service class has the lowest jitter which is < 0.25 ms. And BE service class has the highest jitter.

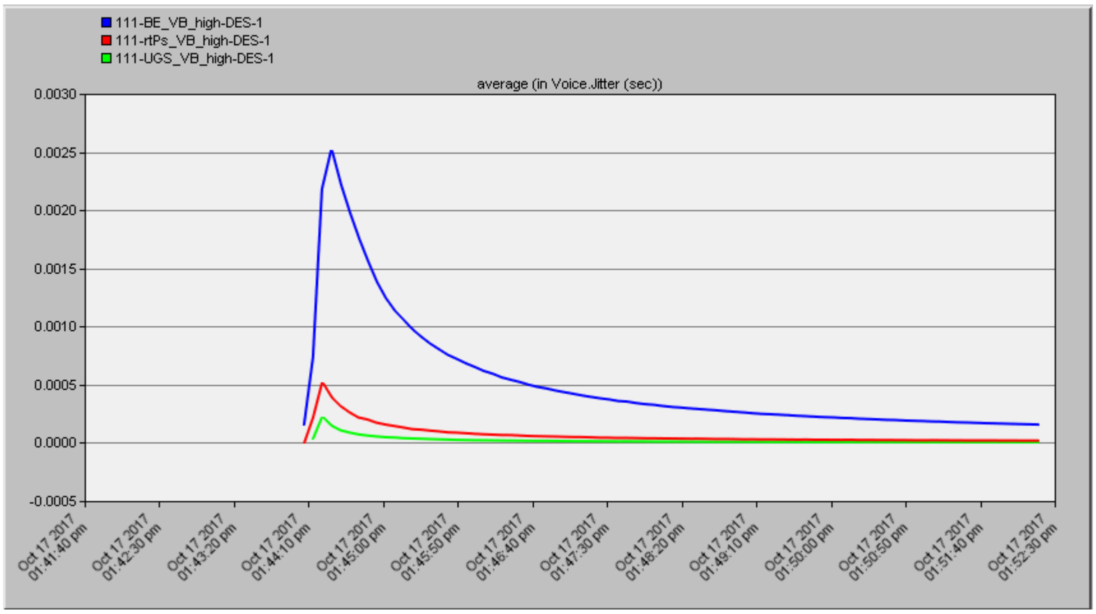

Figure 4-11: jitter for different service classes

4.3.2 MOS of Topology with background application: High size

Figure 4-12 plots the MOS values for the service classes UGS, rtPS and BE. A major observation is that the average MOS almost the same for the three service classes.

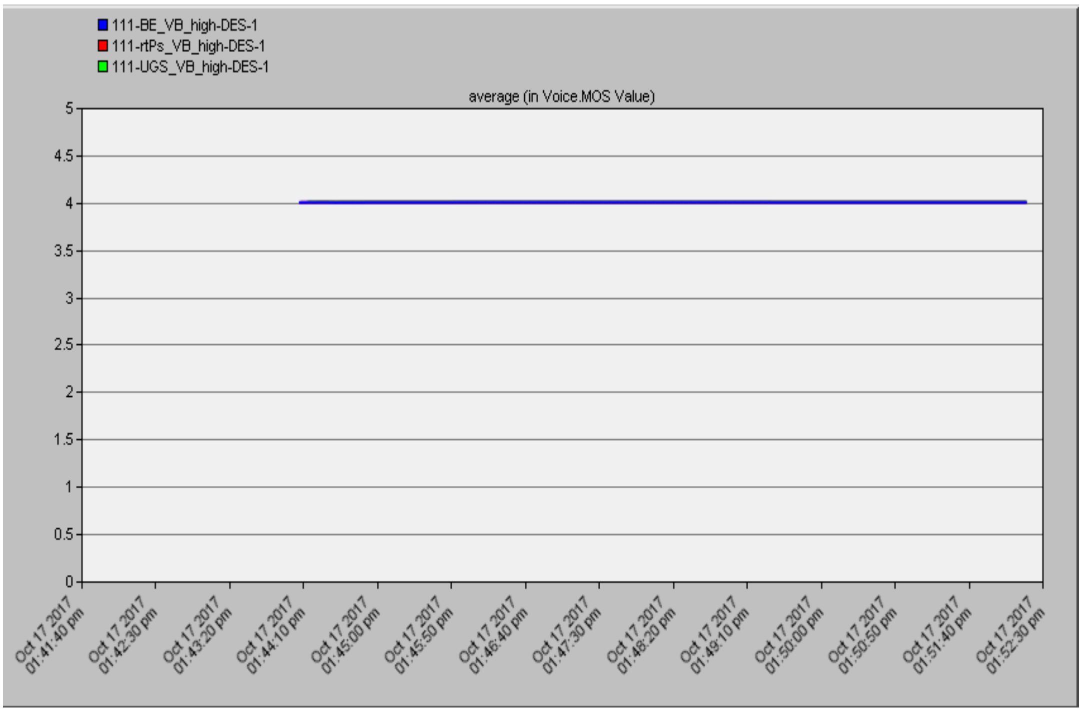

Figure 4-12: MOS for different service classes

4.3.3 End to end delay of Topology with background application: High size

Figure 4-13 show the average packet end-to-end delay, It can be seen from the figure UGS has lowest delay that doesn't exceed 0.35s.

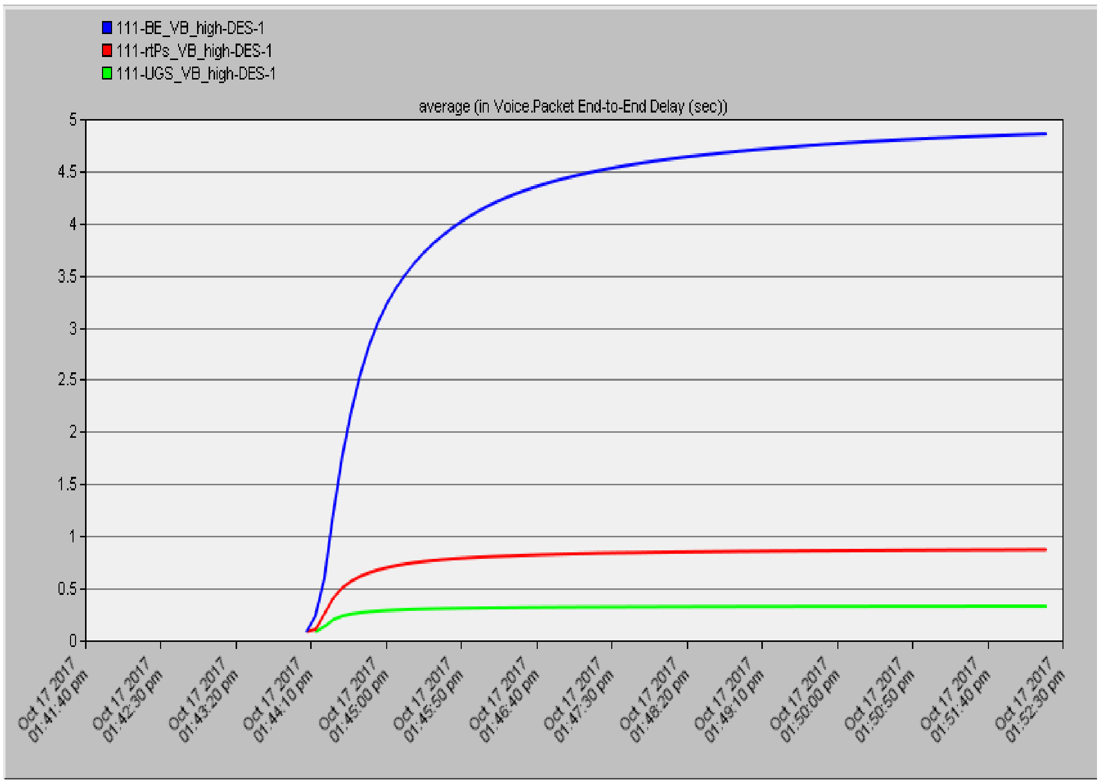

Figure 4-13: packet end_to_end delay for different service classes

4.3.4 Delay of Topology with background application: High size

Figure 4-14 Compares delays. UGS class has the lower delay of approximately 0.147 sec and rtPS with the higher value of 0.42 sec and BE with the highest value which approximately 2.5 sec.

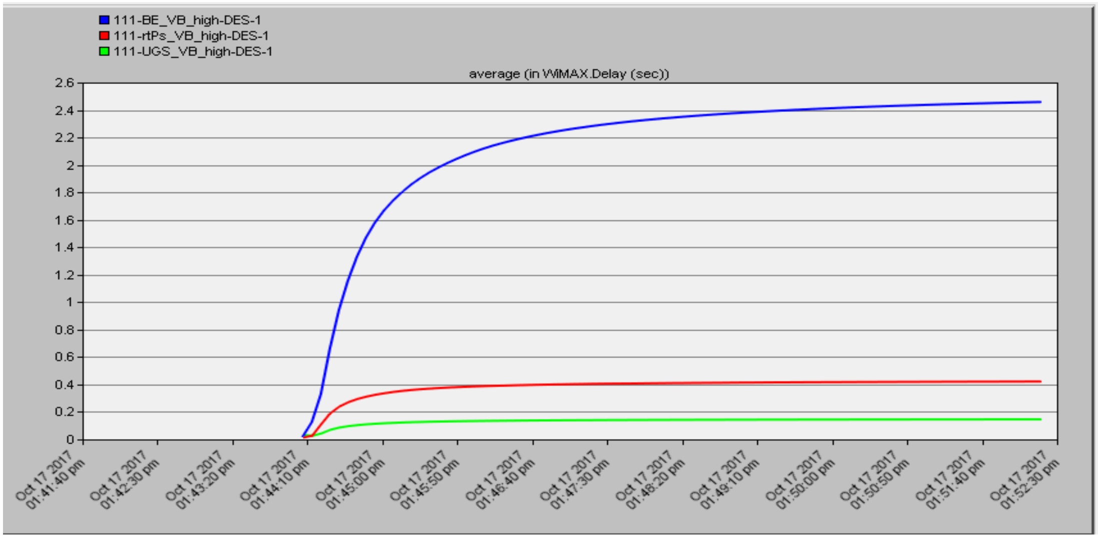

Figure 4-14: delay for different service classes

4.3.5 Throughput of Topology with background application: High size

Figure 4-15 shows the throughput. The throughput of rtPs flow is the highest among the three.

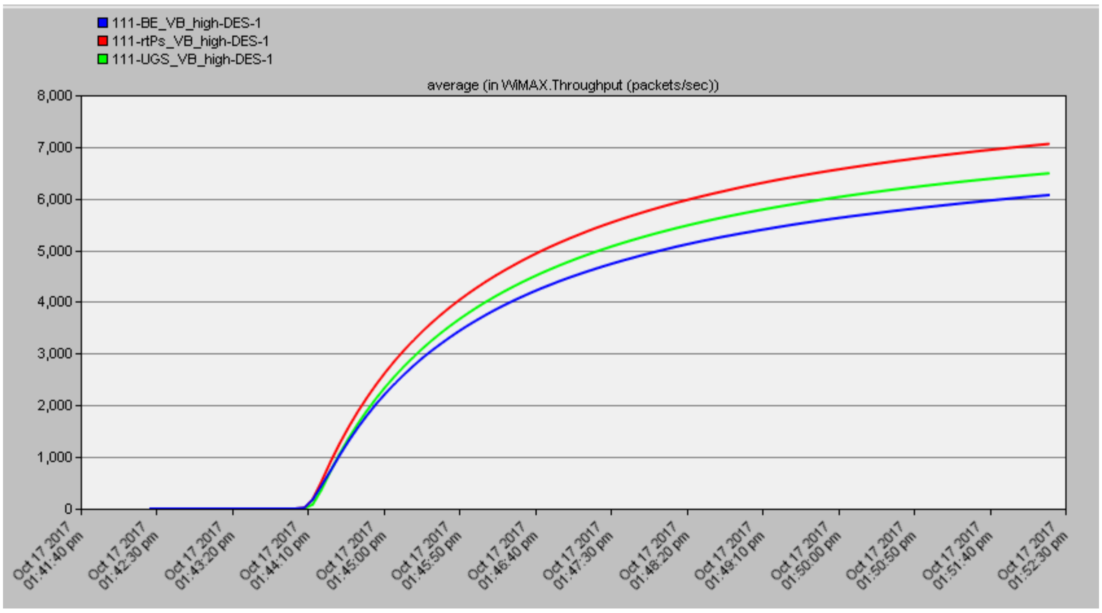

Figure 4-15: Throughput for different service classes

4.4 Network Topology with codecs

This topology contain of one BS and 7 MSs, it was simulated to evaluate VoIP performance bytesting G.711, G.723.1 and 00G.729 codecs.

4.4.1 Jitter Topology with codecs

Figure 4-16 shows the comparative results of voice jitter for the codecs that are used. It can be seen from the figure that the G.723.1 codec scheme has the largest value of jitter variation of 0.000026 sec. The voice jitter value for G.711 is (-0.000012) sec. The voice jitter for G.723 is close to G.711, with a value of (-0.00008) sec.

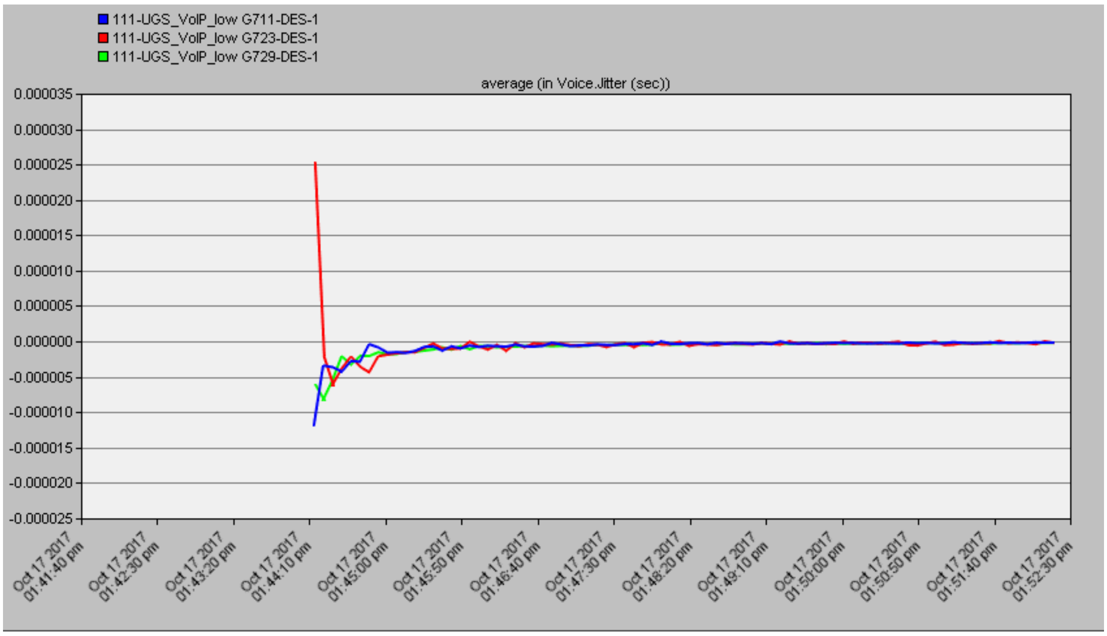

Figure 4-16: jitter for different voice codec

4.4.2 MOS of Topology with codecs

The average MOS value for the three codec's is shown in Figure 4-17. Codec G. 711 achieved the best MOS value of 4.35 followed by G.723 and G.729 with MOS values of 3.9 and 4.0, respectively.

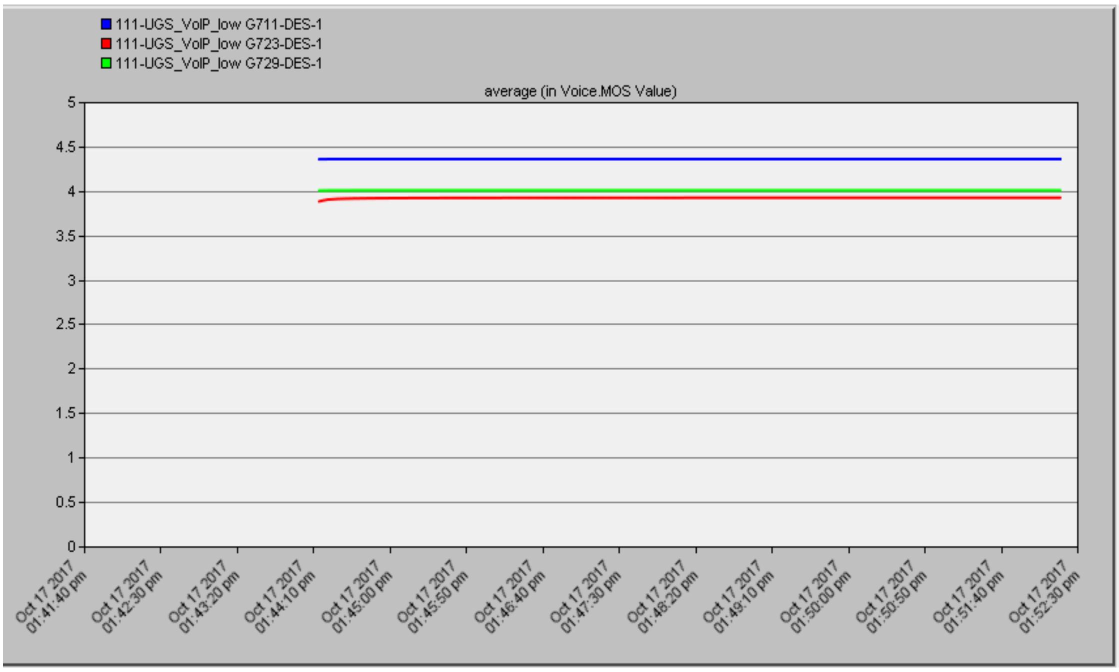

Figure 4-17: MOS for different voice codec

4.4.3 Packet end to end delay of Topology with codecs

All codecs have average end-to-end delays less than 140 ms as shown in Figure 4-18. They are in the range of a good voice connection.

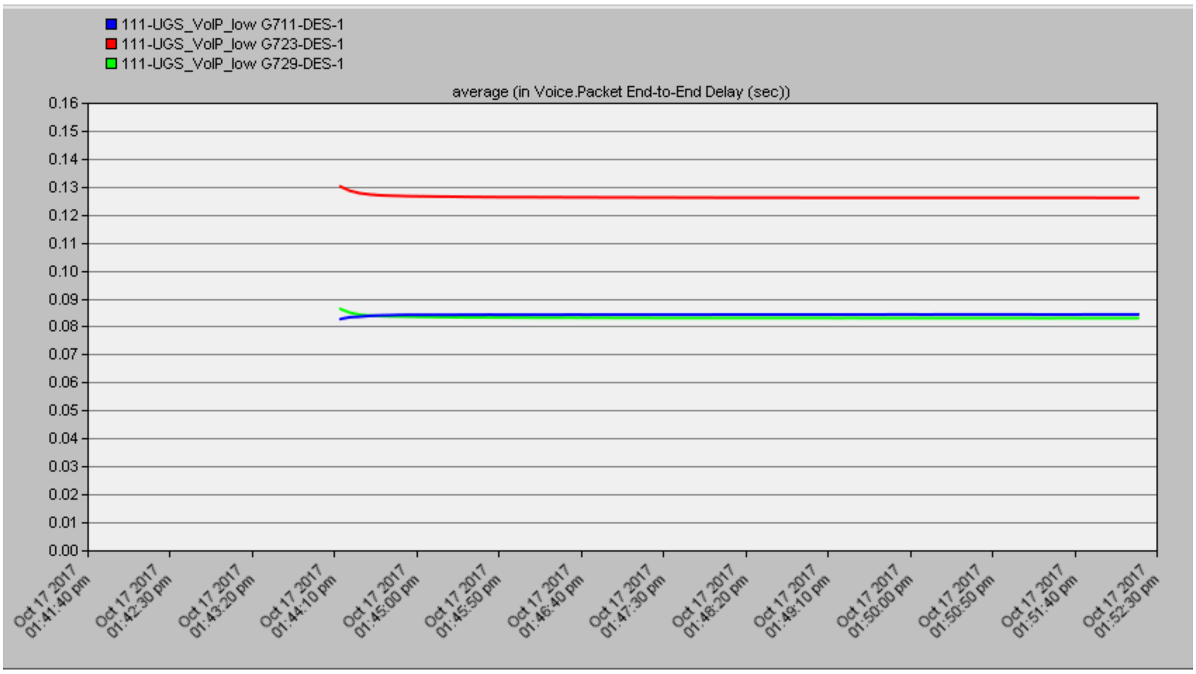

Figure 4-18: packet end_to_end delay for different voice codec

4.4.4 Delay of Topology with codecs

Figure 4-19 presented the average WiMAX network delay. In the Figure, G.729A codec had the least WiMAX network delay of 13.1 ms while G.723.1 recorded the highest network delay of 15 ms. But, all delays are in acceptable rang.

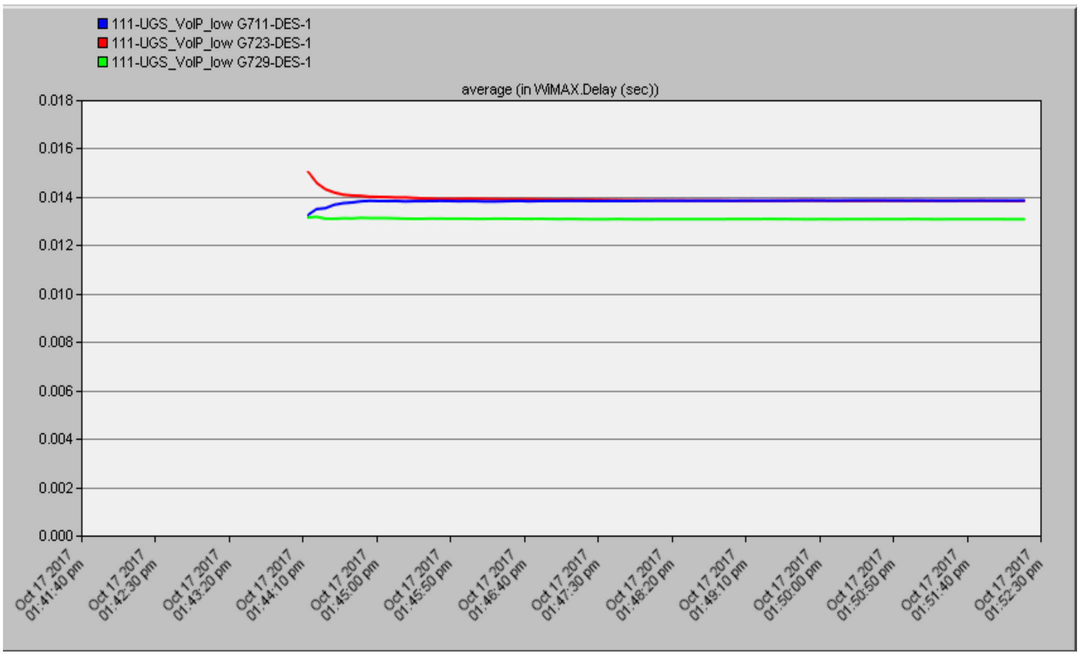

Figure 4-19: delay for different voice codec

4.4.5 Throughput of Topology with codecs

The Figure 4-20 explain the throughput of G.11, G.723.1 and G.729A. G.711 codec has the highest network throughput and this is due to its high bandwidth consumption. The next higher throughput is for G.729A. The G723.1 codec has the lowest throughput.

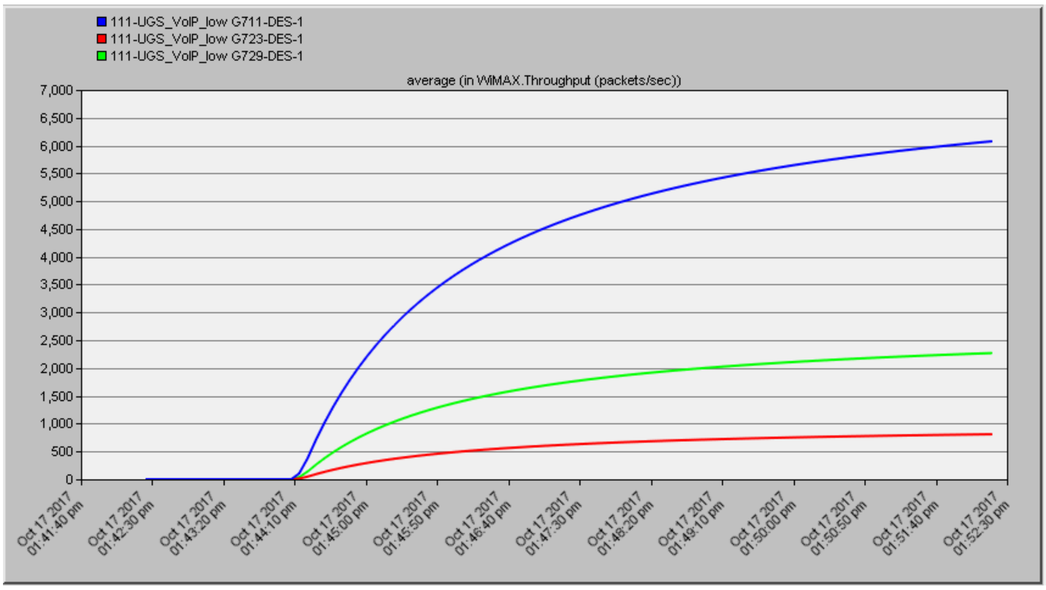

Figure 4-20: Throughput for different voice codec

4.5 Result

All the tested QoS service classes is performed good with low capacity and without background applications. When capacity increase and background application is added, the result shown distinctness. And the efficiency of Qos classes appears. In this scenario, the best QoS is UGS which has lowest delay, best throughput, good MOS and low jitter. The codec G.711 has low delay, best throughput, good MOS and low jitter.

Chapter Five

Conclusion and Recommendations

Chapter Five

Conclusion and Recommendations

5.1 Conclusion

In this research, we have conducted simulation study to evaluate the performance of VoIP over mobile WiMAX networks. We have achieved this through the investigation of WiMAXQoS service classes (UGS, rtPS& BE) and VoIP codecs (G.711, G.729 & G.723.1). The performance ofVoIP over WiMAX was measured and evaluated in terms ofmany QoS parameters such as Jitter, MOS, end-to-end delay, delay andthroughput.

OPNET modeler 14.5 network simulation tool is used to achieve that. Simulation results show that UGS service class has the good performance parameters serving VoIPin different network sizes, Light, High. It is also observed that the G.711 is better than codecs G.723.1 and G.729A because it has acceptable delay,high MOS and high throughput.

5.2 Recommendations

For future work, performance could be enhanced by the following suggestions:

- VoIP can also be simulated in LTE as 4G network, and then compared the simulation results with that on WiMAX network.
- Study of ertPS for VoIP over WiMAX network.

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