Sudan University of Science and Technology
College of Graduate Studies

Performance Evaluation of VoIP QoS in WiMAX Networks

تقييم أداء جودة الخدمة للصوت عبر بروتوكول الإنترنت في شبكات الوايماكس

A Thesis Submitted in Partial Fulfillment of Requirements for the Master Degree in Electronics Engineering (Computer and Network Engineering)

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استهلال

قال تعالى:

أعوذ بالله من الشيطان الرجيم

﴿ وَمَا تَوْفِيقِي إِلَّا بِاللَّهِ عَلَيْهِ تَوَكَّلْتُ وَإِلَيْهِ أُنِيبُ ﴾

[سورة هود - 88]

صدق الله العظيم

Dedication

To my father, mother, brothers, sisters and my friends I dedicate this humble work.

Acknowledgement

Piously my gratitude and prayers to ALMIGHTY Allah for the mercy which followed me during the long path of this research.

Though only my name appears on the cover of this dissertation, a great many people have contributed to its production. I owe my gratitude to everyone who directly or indirectly helped me in this work.

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Eng. Mohammed Makkawi I deeply appreciate your assistance. You were a good friend and always willing to help me. I will be forever thankful to you.

Special thanks for a special friend, Yusra Alhaj. She was patient with my proverbial absence most of the time. She also gave me many precious memories and made a place in my heart only she will ever occupy. Yusra has played the part of friend, confidant, conscience, caring, love and soul etc. In her I have a lifelong friend. I am eternally grateful. Thank you Yusra for being persistent and encouraging, and for believing in me.

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My family thank you for everything. I warmly thank and appreciate my parents and my old brother for their material and spiritual support in all aspects of my life

Abstract

Worldwide Interoperability for Microwave Access WiMAX is a new emerging access technology and one of 4G broadband access wireless technologies with large coverage area, low cost of deployment and high speed data rates. In parallel, transportation of voice over Internet Protocol based networks (VoIP) is a very rapid evolving communication technology. As real-time application VoIP requires packet delivery with low latency, jitter, packet loss, and sufficient bandwidth. WiMAX technology defines 5 different data delivery service classes that can be used in order to satisfy QoS requirements of different applications, such as VoIP, FTP, Web, etc. This research evaluates the performance of VoIP traffic over WiMAX networks. In particular, the study compares the performance obtained using two different QoS service classes that supported VoIP traffic, Unsolicited Grant Service (UGS) and Extended real-time Polling Service (ertPS). Through different network simulation experiments using realistic network scenarios in OPNET environment, this research provides an insight into the VoIP performance in the WiMAX networks. Parameters that indicate the Quality of Service such as end to end delay, packets delay variation, and throughput are obtained and analyzed in these scenarios. In a result of simulation, the general network QoS parameters of the two service classes that used are in the acceptable range. However, UGS has enhanced in range 4.4% to 7.7% for the delay and 21% to 47% for the throughput opposed to ertPS.

المستخلص

تعد وايماكس تقنية جديدة ناشئة للنفاذ، وأحد تكنولوجيا الجيل الرابع من الاتصالات اللاسلكية ذات النطاق العريض مع توفير منطقة التغطية الكبيرة وانخفاض تكلفة النشر ومعدلات البيانات عالية السرعة. وبالتوازي مع ذلك، نقل الصوت عبر بروتوكول الإنترنت هو تطور سريع جدا لتكنولوجيا الاتصالات .يتطلب تطبيق الصوت عبر بروتوكول الإنترنت كمثال علي تطبيقات الوقت الحقيقي تسليم الحزمة ب تأخير منخفض ومعدل التغيير في التأخير منخفض ،مع إنخفاض في فقدان الحزمة وعرض نطاق ترددي كافي. تعرف تقنية الوايماكس خمس فئات مختلفة لخدمة توصيل البيانات التي يمكن استخدامها من أجل تلبية متطلبات جودة الخدمة في التطبيقات المختلفة مثل الاتصالات عبر بروتوكول الإنترنت وبروتوكول نقل الملفات والمتصفح وما إلى ذلك. يقيم هذا البحث أداء حركة الصوت عبر بروتوكول الإنترنت داخل شبكات وايماكس. وعلى وجه الخصوص، تقارن الدراسة بين الأدائين الذين تم الحصول عليهما باستخدام نوعين مختلفين من فئات جودة الخدمة التي تدعم حركة االصوت عبر بروتوكول الإنترنت خدمة المنح غير المرغوبة وخدمة الاقتراع الموسعة في الوقت الحقيقي .من خلال تجارب محاكاة مختلفة باستخدام سيناريوهات شبكات في بيئة برنامج المحاكاة أوبنيت، يوفر هذا البحث نظرة ثاقبة لأداء الصوت عبر بروتوكول الإنترنت في شبكات وايماكس. ويتم الحصول على المعلمات التي تشير إلى نوعية الخدمة مثل التأخر من طرف إلى طرف وتغير تأخر الحزم والإنتاجية وتحليلها في هذه السيناريوهات. وفي نتيجة المحاكاة، ظهرت معلمات نوعية الخدمة في الشبكة العامة لفئتي الخدمة المستعملتين في المدى المقبول. ومع ذلك، فقد تحسن الأداء مع خدمة المنح غير المرغوبة في نطاق 4.4٪ إلى 7.7٪ بالنسبة للتأخير و 21٪ إلى 47٪ بالنسبة للإنتاجية مقارنة بخدمة الاقتراع الموسعة في الوقت الحقيقي.

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Abbreviations

1G	First Generation
2G	Second Generation
3G	Third Generation
3GPP	Third Generation Partnership Project
4G	Fourth Generation
AAA	Authentication, Authorization, and Accounting
AMPS	Advanced Mobile Phone System
ASN	Access Service Network
ASP	Application Service Provider
BE	Best Effort
BPSK	Binary Phase Shift Keying
BS	Base Station
BWA	Broadband Wireless Access
CBR	Constant Bit Rate
CDMA	Code Division Multiple Access
CP	cyclic prefix
CSN	Connectivity Service Network
DHCP	Dynamic Host Control Protocol
DL	Download ratio

EDGE	Enhanced Data for Global Evolution
ertPS	extended real-time Polling Service
ETE delay	End-to-End delay
FDD	Frequency Division Duplex
FFT	Fast Fourier Transform
FTP	File Transfer Protocol
GPRS	General Packet Radio Service
GSM	Global System for Mobile
GW	Gateway
IEEE	Institute of Electrical and Electronics Engineers
IFFT	Inverse Fast Fourier Transform
IMT	International Mobile Telecommunications
IP	Internet Protocol
ITU-R	International Telecommunications Union Radio Communications sector
LOS	line of sight
LTE	Long Term Evolution
MAN	Metropolitan Area Network
MIMO	Multi-Input Multi-Output
MIP-HA	Mobile IP Home Agent
MOS	Mean opinion score
MRT	Maximum Ratio Transmission
MS	Mobile Station
NLOS	Non-line-of-sight
NRM	Network Reference Model
nrtPS	Non-Real-Time Polling Service

NSP	Network Service Provider
NWG	Network Working Group
OFDM	Orthogonal Frequency-Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
OPNET	Optimized Network Engineering Tool
OSS	Operation Supports Systems
PMP	Point-to-Multipoint
PSTN	Public Switched Telephone Network
PTP	Point-to-Point
QAM	Quadrature Amplitude Modulation
QoS	Quality of service
QPSK	Quadrature Phase-Shift Keying
RAN	Radio Access Network
RF	Radio frequency
RP	Reference Point
RTP	Real-Time Transport Protocol
rtPS	Real-Time Polling Service
SMS	Short Message service
SS	Subscriber Stations
STBC	Space Time Block Coding
TCP	Transport Control Protocol
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
UDP	User Datagram Protocol
UGS	Unsolicited Grant Service
UL	Upload ratio

UMTS	Universal Mobile Telecommunications System
VMS	Voice Mail Service
VoD	Video on Demand
VoIP	Voice over Internet Protocol
WIMAX	Worldwide Interoperability for Microwave Access
WMAN	Wireless Metropolitan Area Network

Chapter One

Introduction

1.1 Preface

Voice over Internet Protocol (VoIP) practices is potentially mounting day by day resulting in the demand of rapid improvements in the networks. There is a demand of decreasing the difference between the qualities of voice and increasing the available bandwidth to provide the best VoIP services comparative to the traditional circuit switched telephony.[1]

VoIP technology turns analog voice into digital data packets that can be stored, searched, manipulated, copied, combined with other data, and distributed to virtually any device that connects to the IP network. This capability virtually made it possible to achieve maximum flexibility in the transport or transmission of voice that has been transformed into data. Making use of the internet infrastructure or in transmitting digitized voice the same way data packets are transmitted across great distances have virtually made VoIP the answer to the availability of infrastructure. [2]

The most important factor that has moved VoIP into the mainstream is improved quality of service (QoS). Providing the required QoS is vital to deliver a good user experience over Internet. The notion of QoS is becoming even more important in fourth-generation wireless technologies such as Worldwide Interoperability for Microwave Access (WIMAX) is designed to support current and future QoS needs.

An attractive wireless technology for VoIP is WiMAX specified by IEEE 802.16 standard aimed at providing wireless access over long distances in a variety of ways from point-to-point communication to mobile cellular access.

WiMAX provides wide coverage area with lower cost of network deployment. The coverage area of a single WiMAX cell is around 30 to 50 km, and its speed is up to 40 Mbps. Moreover, WiMAX supports Quality of Service (QoS) by providing different service classes for both real-time and non-real-time traffic. [3]

WiMAX is broadband wireless access (BWA) network technology, aiming to play a crucial role in 4G and beyond “all-IP” broadband wireless access technologies progress. It was developed to support higher number of users with higher data rates, coverage and availability. [4]

1.2 Problem Statement

Since VoIP services have yet to establish the credibility with regards to the quality of voice, it is needed to identify the factors that can affect the voice quality such as end to end delay, jitter, and throughput.

1.3 Proposed Solution

This research is to find out to what extent the QoS of the VoIP traffic varies while traveling through the latest generation of networks. An implementation of OPNET is needed to study and evaluate the QoS over WiMAX, and to investigate how well this network cope with voice application. The performance comparison of VoIP on WIMAX is focused and analyzed End-to-end delay, Packet Delay Variation (jitter), and Throughput.

1.4 Aim and Objectives

The focus of this research is directed to the WIMAX 4G Network QoS for VoIP.

- To simulate WiMAX model and evaluate the performance of QoS implemented for voice traffic in various network scenarios.
- To analyze and compare the QoS parameters performance of the proposed network. End-to-end delay, Packet delay variation, and throughput are considered for the comparison.

1.5 Methodology

The analysis that has been taken out is to evaluate the WiMAX network environment in which VoIP communication is being conducted. Several QoS classes and parameters can be used, in this analysis, Unsolicited Grant Service (UGS) and Extended real-time Polling Service (ertPS) service classes are used, and three QoS parameters have been selected, system end to end delay, packet delay variation, and throughput. The main justification for this selection is such that the study focuses on the performance of VoIP communication over different networking conditions and environment, and hence the time and reliability would be the major concerns for evaluation.

Educational version of Optimized Network Engineering Tool (OPNET) is used to implement and design the network models. The OPNET modular enables us to build an experimental 4G networks, WIMAX topology, simulate it and collect the required results for analysis and evaluation of QoS performance in this network. The considered parameters are evaluated, discussed, and then compared.

1.6 Thesis Outlines

In general the thesis will be divided into five chapters. Each chapter will discuss on different issues related to the project. The following are the issues discussed.

Chapter One states the problem, proposed solutions and methodology.

Chapter Two describes the background required to understand the proposed system and some examples of related works.

Chapter Three define tools and program sittings that used to apply the design

Chapter Four analyzing the results for each scenario, which have been created to make environment study and notice the change in performance parameters in every scenario.

Chapter Five outlines conclusion that have reached by the experience and recommendations for future works**.**

Chapter Two

Literature Review

2.1 Background

In this section, the background information of WiMAX network will be demonstrated.

2.1.1 Evolution of Network Generations

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, and 3G the first mobile systems handling broadband data. This continuing race of increasing sequence 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.

2.1.1.1 First Generation (1G)

The first generation cellular networks were invented in the 1980s. The key idea behind 1G was that the geographical area is divided into cells (typically 10-25km), each served by a “base station.” Cells are small so that frequency reuse can be exploited in nearby (but not adjacent) cells. This allows many more users to be supported in a given area. All 1G systems were analog systems popularly known as early cellular phone technology working in the frequency band of 150 MHz. [5]

The technologies were used in the 1st Generation of Wireless Telecommunication are Advance Mobile Phone Service (AMPS), Nordic Mobile Telephone (NMT), Total Access Communication System (TACS).

There is need for proper band width in this generation for a large number of users. The major disadvantage of 1G is the quality of voice, there was no clarity of noise and a constant disturbance from background noise [6].

2.1.1.2 Second Generation (2G)

The Second Generation (2G) cellular telecom networks were commercially launched on the Global System for Mobile (GSM) standard in Finland by *Radiolinja* in 1991. It used digital signals for voice transmission and had a speed up to 64 kbps. 2G Technology came up with many data services for mobile. Voice Mail Service (VMS) was also proud and value added service in 2G. A new feature Short Message service (SMS) was an added in 2G. Furthermore, the system use Band width range of 30-200KHZ. [5, 6]

Also, 2G mostly based on circuit switched technology, which are digital and expand the range of applications to more advanced voice services. 2G wireless technologies can handle some data capabilities such as fax and short message service at the data rate of up to 9.6 kbps, but it is not suitable for web browsing and multimedia applications.[7]

Some key benefits of 2G Network over its predecessors was that, Digital Encryption was supported by 2G systems which had higher penetration efficiency thereby being more efficient on network spectrum.

Furthermore, 2G was upgraded to 2.5G. This is a technology which was introduced in 1990's. It uses a technology known as General Packet Radio Service (GPRS) stand. In this technique delivering packet switched data capabilities to already existing Global System for Mobile (GSM) networks. A add on feature of sending Graphics data as packets is available in this technology packet switching made its impact with increasing Internet and Internet protocol. Enhanced Data for Global Evolution (EDGE) network is an example of 2.5G.[6]

2.1.1.3 Third Generation (3G)

International Mobile Telecommunications-2000 (IMT-- 2000), also known as 3G, is a generation of standards for mobile phones and mobile telecommunications services fulfilling the International Telecommunication Union. It uses Wide Brand Wireless Network with which clarity is increased. The data are sent through the technology called Packet Switching. Voice calls are interpreted through Circuit Switching. Along with verbal communication it includes data services, access to television/video, new services like Global Roaming. It operates at a range of 2100MHz and has a bandwidth of 15-20MHz used for High-speed internet service, video chatting.3G uses Wide Band Voice Channel that is by this the world has been contracted to a little village because a person can contact with other person located in any part of the world and can even send messages too [5, 8]

Although the usage price of 3G technology has greatly reduced since its inception due to wider adoption, it still is very costly as compared to 2G technologies. Due to high bandwidth transmission of 3G technologies, power consumption greatly increases which results in reduced device battery life.

2.1.1.4 Fourth Generation (4G)

Third generation (3G) mobile networks faces a new rival; so called 4G. An astonishingly new network may be even more profitable. The goal of 4G is to replace the current proliferation of core cellular networks, with a single worldwide cellular core network based on standard IP for control and media.[9]

The International Mobile Telecommunications Advanced (IMT-Advanced) specification sets the peak speed requirements for 4G service, 100 megabits per second (Mbit/s) for high mobility communication and 1 gigabit per second (Gbit/s) for low mobility communication. A 4G system not only provides voice and other 3G services but also provides ultra-broadband network access to mobile devices. Applications vary from IP telephony, HD Mobile Television, video conferencing to gaming services and cloud computing. One of the initial devices to access 4G network was USB wireless modem which was later followed by cellular phones with WiMAX and Long Term Evolution (LTE) technology.[5]

In spite of different approaches, each resulting from different visions of the future platform currently under investigation, the main objectives of 4G networks can be stated in the following properties:

Ubiquity means that mobile networks must be available to the user, anytime, anywhere. To accomplish this objective, services and technologies must be standardized in a worldwide scale. Furthermore, the services to be implemented should be available not only to humans as have been the rule in previous systems, but also to everything that needs to communicate such as M2M communications.

A multi-service platform is an essential property of the new mobile generation, not only because it is the main reason for user transition, but also because it will give telecommunication operators access to new levels of traffic. Voice will loose its weight in the overall user bill with the raise of more and more data services.

Low-bit cost is an essential requirement in a scenario where high volumes of data are being transmitted over the mobile network. With the actual price per bit, the market for the new high demanding applications, which transmit high volumes of data (e.g. video), is not possible to be established. [7, 10]

To achieve the proposed goals, a very flexible network that aggregates various radio access technologies, must be created. This network must provide high bandwidth, permit fast handoffs, and efficient delivery system over the different wireless technologies available. Also what is necessary a QoS framework that enables fair and efficient medium sharing among users with different QoS requirements, supporting the different priorities of the services to be deployed. The network should also offer sufficient reliability by implementing a fault-tolerant architecture and failure recovering protocols.[11]

2.1.2 WiMAX Overview

The WiMAX is an evolving IEEE standard and is also known as IEEE 802.16. WiMAX, like 2G/3G networks, can provide service on the scale of Metropolitan Area Network (MAN) with high bandwidth. The WiMAX wireless technology is called the last-mile solution for wireless broadband access. It can also act like a hot-spot. WiMAX has benefits in terms of spectral efficiency, wider coverage, easy deployment and frequency re-use. IEEE standard just provides the WiMAX technology. A large organization called WiMAX forum made of network operators, academics and telecommunication members' work on the compatibility, technicality, regulatory, and marketing aspects of the WiMAX.[12]

Worldwide Interoperability for Microwave Access (WiMAX) technology, also known as the IEEE 802.16 standard, is based on Wireless Metropolitan Area Network (WMAN). It provides data rates up to 75 Mbps over the distance of 50 km. WiMAX uses frequency bands of 10-66 GHz, covering long geographical areas using licensed or unlicensed spectrum. WiMAX uses Orthogonal Frequency Division Multiple Access (OFDMA) as multiplexing technique in uplink and downlink directions. The mode of operation used for communication between multiple subscriber stations and base station is Point-to-Multipoint (PMP), whereas the mode of operation used between two base stations is Point-to-Point (PTP).

Other versions of WiMAX include IEEE 802.16-2004 and IEEE 802.16-2005. IEEE 802.16-2004 is known as fixed WiMAX, has no mobility and is used for fixed and nomadic access. Since fixed WiMAX has no mobility it does not support handovers. IEEE 802.16-2005 is known as mobile WiMAX, which is an extension of fixed WiMAX, introducing many new features to support enhanced Quality of Service (QoS) to provide high mobility. The mobile WiMAX supports data rate of up to 75 Mbps.[13]

2.1.3 System Architecture

WiMAX system architectures is optimized for packet data in order to support the increasing amounts of data traffic. Compared with existing wide area cellular system's architectures, they provide improved respond times and both of these are all-IP backbone. These new system architectures benefit not only the subscribers but also the mobile wireless operators.

2.1.3.1 WiMAX Architecture

The WiMAX architecture is based on a network reference model to define end-to-end WiMAX network.

Network Reference Model (NRM)

The network reference model for WiMAX was developed by the WiMAX Network Working Group (NWG). The model defines the entire WiMAX network. The NRM ensures interoperability between various WiMAX enabled devices and operators. The network architecture is based on IP services and it can be logically divided into three parts; Mobile Station, Access Service Network and Connectivity Service Network. The network reference model is described in Figure 2-1.

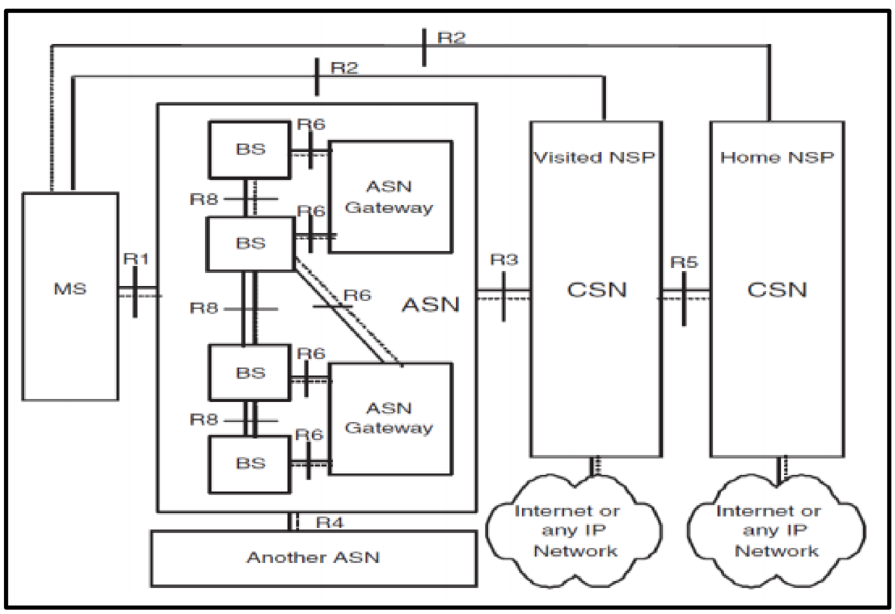

Figure 2-1: Network Reference Model for WiMAX [14]

a. **Mobile Station (MS):** Used to access the network.
b. **Access Service Network (ASN):** Comprised of ASN Gateways (GWs) and base stations (BSs) to form Radio Access Network (RAN) at the edge.
c. **Base Station:** Provides air interface to MS. In addition, BS is responsible for handoff triggering, radio resource management, enforcement of QoS policy, Dynamic Host Control Protocol (DHCP) proxy, session management, key management and multicast group management.
d. **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). [12]
e. **Connectivity Service Network:** Provides IP connectivity to internet, Public Switched Telephone Network (PSTN), Application Service Provider (ASP) and corporate networks. In addition, it provides core IP functions. CSN is owned by the Network Service Provider (NSP), and is comprised of AAA servers, Mobile IP Home Agent (MIP-HA), Operation Supports Systems (OSS) and gateways. AAA servers are used to authenticate devices, users and specific services. CSN has following responsibilities:
    - IP address Management.
    - Mobility, roaming and location management between ASN's.
    - Roaming between NSPs by Inter-CSN tunneling.

The logical link that connects two functional groups is called Reference Point (RP). The NRM shown in figure 2.1 has 8 RPs ranges from R1 to R8. The description of RPs is given in table 2-1. [12, 14]

2.1.3.2 Air interface

WiMAX network support Frequency Division Duplex (FDD) and Time Division Duplex (TDD) mode in radio access modes, although the earlier version of WiMAX only support TDD mode. Besides, cellular operators prefer to choose the FDD mode since most of current cellular systems basing on FDD mode so that it is easier to migrate to LTE network. That is also one of the reasons why the following version of WiMAX supports the FDD mode. Besides, a combination of various modulation schemes is used in this system in order to adapt different demand, such as high throughput. Moreover, the Multi-Input Multi-Output (MIMO) technology is used. WiMAX network defines 2x2 MIMO. Six aspects of air interface standard are listed in table 2-2. [15]

Table 2-1: Description of Reference Points

Reference Points	**Description**
R1	Connect Mobile Station (MS) and ASN
R2	Connect MS and CSN
R3	Connect ASN and CSN
R4	Connect two ASNs
R5	Connect two CSN
R6	Connect BS and ASN- GW
R7	Represents the internal communication within the gateway.
R8	Connect two Base Stations (BSs)

Table 2-2: Comparison of Air Interface Standard

Aspects	WiMAX
Frequency bands	2.5 to 11 GHz (fixed WiMAX) 2 to 6 GHz (mobile WiMAX)
Radio access modes	FDD TDD (earlier version only supports TDD)
Modulation	DL: BPSK (optional for OFDMA-PHY) QPSK 16QAM 64QAM UL: BPSK QPSK 16QAM 64QAM(optional)
Peak data rates	DL: 75 Mbps UL: 25 Mbps
Multiple access technology	DL: OFDMA UL: OFDMA
Multiple antenna techniques	DL: 2×2 MIMO UL: 2×2 MIMO

2.1.4 Multiple Access Technology

Downlink and uplink transmission WiMAX are based on multiple access technologies. A technology called Orthogonal Frequency Division Multiple access technology (OFDM) is used for uplink and downlink transmission of WiMAX. OFDMA is a physical layer technology.

2.1.4.1 OFDMA (WiMAX Uplink/Downlink)

OFDMA is derived from Orthogonal Frequency Division Multiplexing (OFDM), a digital multi-carrier modulation scheme which uses the principle that information can be transmitted on radio channel through variations of carrier signals frequency, phase or magnitude. 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.[16]

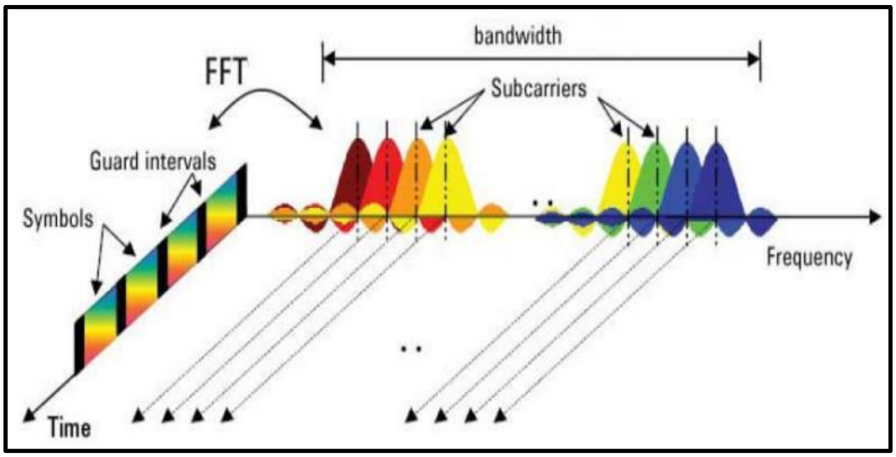

Figure 2-2: OFDM Signal Representation in Frequency and Time Domain[16]

In OFDM, the subcarriers are spaced closely together without any guard bands in frequency domain and use the FFT to convert the digital signals from time domain into a spectrum of frequency domain signals that are mathematically orthogonal to each other. The frequency domain null of one subcarrier corresponds to the maximum value of adjacent subcarrier which allows subcarriers to overlap without interference and thus conserve bandwidth. 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 the channel. OFDMA provides a robust system with increased capacity and resistance to multipath fading.

In WiMAX, subcarrier is modulated with a conventional modulation scheme depending on the channel condition. WiMAX uses BPSK, QPSK, 16QAM, 64QAM for modulation at a low symbol rate. The FFT sizes of 128, 256, 512, 1024, and 2048, corresponding 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 normal CP for WiMAX is 1/8 the length of OFDMA symbol time, typically 11.43 µs for OFDMA symbol duration of 102.86 µs. The CP is copy of the end of the symbol inserted at the beginning. The figure 2-3 shows OFDMA transmitting a series of QPSK data symbols. [16]

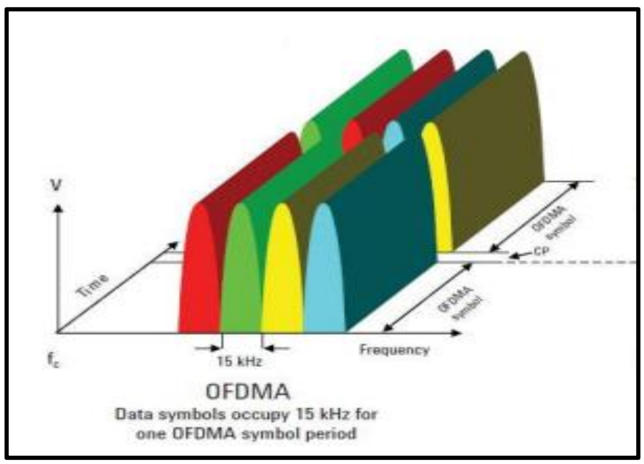

Figure 2-3: OFDMA Transmitting a Series of Data Symbols[16]

2.1.4.2 Multiple Input Multiple Output (MIMO)

MIMO refers to a system having minimum two antennas at the base station as well as at the mobile station. MIMO system enhances the performance of WiMAX including spatial multiplexing, diversity and interference reduction. WiMAX supports two forms of MIMO systems, Open loop MIMO and Closed loop MIMO systems. A general MIMO system is shown in Figure 2-4.

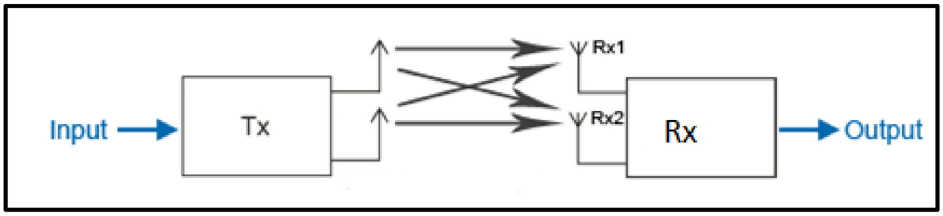

Figure 2-4: General MIMO System

a. Open loop MIMO System

Open loop MIMO techniques are subdivided into Matrix A and Matrix B. Open loop MIMO does not utilize the information of the channel. Matrix A refers to the Space Time Block Coding (STBC) whereas Matrix B refers to the spatial multiplexing in WiMAX. Open loop techniques increase the range and capacity of WiMAX.

b. Closed loop MIMO System

The transmitter collects information about the propagation channel in the closed loop MIMO to further enhance coverage and capacity of WiMAX. Closed loop MIMO utilizes the beamforming or Maximum Ratio Transmission (MRT) [14]. The Multiple antenna organization chart for WiMAX is shown in Figure 2-5.

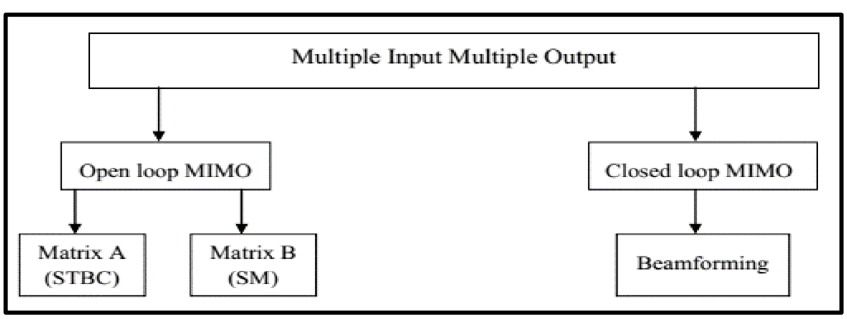

Figure 2-5: WiMAX Multiple Antenna Implementation Organization Chart [14]

2.1.5 QoS in WiMAX

WiMAX networks must support a diversity of applications, such as video, voice, multimedia and data and each of these has different traffic patterns and QoS necessity. The QoS is granted on the basis of type of application and service under consideration. For example, a user sending an email needs no real-time data stream like another user having a Voice over IP (VoIP) application. To provide the service parameters respectively, the traffic management is necessary.

2.1.5.1 QoS Service Classes in WiMAX

There are four main service classes named as UGS, rtPS, nrtPS, BE but there is a fifth type QoS service class which is added in 802.16e standard, named as: extended real-time Polling Service (ertPS). These services are prioritized in decreasing order. Within all these classes of services resources are allocated to manage and satisfy the QoS of higher priority services. In general, IEEE 802.16 has five QoS classes Table 2-3 broadly classifies various service classes defined in WiMAX and its applications.[17, 18]

1. **Unsolicited Grant Service (UGS)**

   Supports real-time data streams for delay constraint traffic which require optimal throughput. UGS supports jitter tolerant, maximum latency tolerant (5-40 ms latency over the air and 100ms latency over an IP backbone) and maximum sustained rate applications. UGS supports application with Constant Bit Rate (CBR) service, such as VoIP for which achieving low latency is extremely critical. In WiMAX, UGS flows are buffered separately from the other service classes, such as nrtPS and Best Effort (BE), so they get higher priority over other trivial applications, such as SMS. During the upstream, the system uses UGS to bypass the normal request-grant mechanism for upstream traffic by allowing the base station to give automatic grant to a UGS flow.[17, 18]

2. Real-time Polling Service (rtPS)

Supports real-time data streams. It is used for real-time services, such as streaming video that generates the data packets of variable sizes with variable bit rates, a guaranteed minimum rate and a guaranteed delay. The rtPS has more request overhead than UGS, but it supports variable grant sizes for data transport efficiency. Unlike UGS, there is a polling overhead which can sometime reach up to 60 percent. The rtPS supports periodic, high priority, maximum latency tolerance, maximum reserved rate and maximum sustained rate applications. A drawback of this QoS type is that it has a significant impact on the overall throughput. [17, 19]

3. Non-real-time Polling Service (nrtPS)

Supports delay tolerant data with variable packet sizes. The nrtPS service class supports non-real-time services that require variable size data packets, and a minimum data rate with higher latency, such as file transfer protocol (FTP). This is done by using unicast polls on a regular basis, which ensures that the service flow receives requests even during network congestion. Priority is given to UGS and rtPS applications over nrtPS.[19]

4. Best Effort (BE)

Supports data streams where no minimum data rate is required, and packets are handled based on available bandwidth. Unicast polling requests are not guaranteed in this case, requiring contention requests to be used. BE packets may take long time to transmit during network congestion.[17]

5. Extended real-time Polling Service (ertPS)

This type of QoS is used for scheduling algorithms for VoIP service with variable data rates and silence suppression. It has been newly introduced to support real- time service flows that generate variable sized data packets on a periodic basis with minimum reserved rate, maximum sustained rate, maximum latency tolerance, jitter tolerance and traffic priority. The ertPS service class enables silence suppression mechanism and makes better use of header compression. VoIP is an example of ertPS class application. [12, 19]

Table 2-3: QoS Classes in WiMAX

Service classes	Description	Applications
Unsolicited Grant service (UGS)	For constant Bit rate and delay dependent applications	VOIP
Real Time Polling Service (rtPS)	For variable rate and delay dependent applications	Streaming audio , video
Extended Real time Service (ertPS)	For variable rate and delay dependent applications	VOIP and Silence Suppression
Non real time polling service (nrtPS)	Variable and non-real time applications	FTP
Best Effort (BE)	Best effort	Email, Web, Traffic

2.1.5.2 QoS Parameters

QoS requirements become very important in WiMAX technology to guarantee their performance, in the presence of various types of connections, such as current calls, new calls and the handoff connection.

Throughput is measure of number of packets successfully delivered in a network. It is measured in terms of packets/second. The throughput value should be high or low it affects every service class defined in WiMAX.

Jitter is the variation in the delay time of packets arriving at their destination. VoIP packets sent at regular intervals from the sender to the receiver, but network latency of interval between packets can vary at the destination.

Delay or latency could be defined as the time taken by the packets to reach from source to destination. The main sources of delay can be categorized into: propagation delay, processing of source delay, network delay and destination delay. End to end delay is a measure of elapsed time taken during modulation of the signal and the time taken by the packets to reach from source to destination. Table 2-4 summarize and clarify QoS parameter constrains for VoIP application. [20]

Table 2-4: Respective Quality of Service (QoS) Requirements

Parameter	Definition	Measurement Units	QoS Requirements
Throughput	Total data transferred from one node to another	Bytes/second	n/a
Percent Data Loss	$\frac{\text{Packet sent} - \text{Packet recived}}{\text{Packet sent}}$	Percent	≤1%
End-to-End Delay	Total delay for data transfer to occur.	Milliseconds	≤140 ms
Jitter	Variation in packet arrival.	Milliseconds	≤0.5 ms

2.1.6 Voice over Internet Protocol

VoIP application typically works as follows. First, a voice signal is sampled, digitized, and encoded using a given algorithm/coder. The encoded data (called frames) is packetized and transmitted using RTP/UDP/IP. At the receiver's side, data is de-packetized and forwarded to a playout buffer, which smoothes out the delay incurred in the network. Finally, the data is decoded and the voice signal is reconstructed. "Measuring Data and VoIP Traffic in WiMAX Networks". Figure 2-6 gives us an idea about the transmission flow from the sender to receiver with the network transmission consisting of uplink, backbone and then downlink transmission.[21]

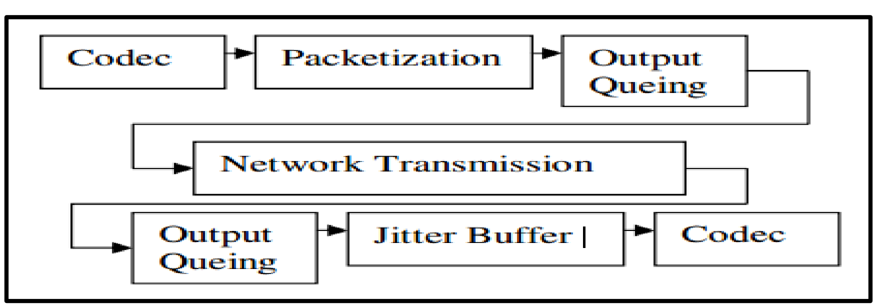

Figure 2-6: VoIP Components [21]

2.1.6.1 VoIP Components

VoIP have been widely accepted for its cost effectiveness and easy implementation. A VoIP system is divided into three components, namely codec, packetizer, and playout buffer.

- **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 advantage of using codec.
- **Packetizer**: With the help of packetizer output digital streams are packed into constant bit rate voice packets.
- **Playout buffer**: A two way conversation is very sensitive to packet delay jitter so to eliminate the delay, playout buffer is used at receiver end. [22]

2.1.6.2 VoIP Transport System

VoIP uses a combination of protocols for delivering phone data over networks. Various. Signaling protocols are used like SIP and H.323.These can be regarded as the enabler protocols for voice over IP (VoIP) services. VoIP communications require these signaling systems to setup, control, initiate a session and facilitate real-time data transfer in order to provide clear communications. SIP and H.323 works in conjunction with the Real Time Transport Protocol (RTP) and the User Datagram Protocol (UDP) to transfer the voice stream. Voice data is put in data packets using the RTP protocol. The RTP packets, enclosed inside the UDP packets, are then transferred to the receiver.

2.1.6.3 VoIP Codec's

RTP and UDP protocols are the logical choice to carry voice. The transport control protocol (TCP) favours reliability over timeliness. Voice signals are digitally encoded. This means that each voice signal is converted from digital to analog and back. The analog signal is firstly sampled based on a sampling rate of 8 KHz, 8 bits per sample is the most frequently cases. Next, the output is encoded according to many factors: the compression rate and the framing time or the frames length.

Finally, one or more of these frames are encapsulated into an RTP/UDP/IP packet for transmission over the network. All these practices are achieved by one of various audio codec's, each of which vary in the sound quality, the bandwidth required, the computational requirements, encoding algorithm and coding delay. They are as discussed below:

G.711 is the default standard for all vendors and manufacturers and provides very low processor requirements. This standard digitizes voice into 64 Kbps and does not compress the voice. It performs excellent in local networks where we have ample amount of available bandwidth.

G.729 is supported by many vendors for compressed voice operating at data rate of 8 Kbps. Excellent bandwidth utilization and Error tolerant with quality just below that of G.711.

G.723.1 was once the recommended compression standard. It operates at data rate 6.3Kbps and 5.3 Kbps. High compression with high quality audio. Although this standard decreases bandwidth consumption, voice is much poorer than with G.729 and is not very popular for VoIP traffic.[23]

2.1.6.4 VoIP Signaling Protocol

The mechanism for carrying a VoIP connection generally involves a series of signaling transactions between the endpoints. There are several protocols in existence to handle this.

- **H.323**

H.323 was developed by the ITU in May of 1996 as a means to transmit voice, video, data, and fax communications across an IP-based network while maintaining connectivity with the PSTN. H.323 has not enjoyed much success among users and enterprises, although it is still the most widely used VoIP protocol among carriers. The standard is interoperable and has both point to point and multipoint capabilities. H.323 uses a number of other sub protocols for the various functions.

- **H.255.0:** Registration, Admission, Status, Call Signaling, Control.
- **H.245**: Terminal Capability Exchange, Media Description, Control of Logical Channel.

H.323 supports Secure Real-Time protocol (SRTP) for media confidentiality, and Multimedia Internet Keying (MIKEY) for key exchange. [24, 25]

2.1.6.5 Session Initiation Protocol (SIP)

SIP is a more recent standard for multimedia conferencing over IP. The standard was defined by the Internet Engineering Task Force (IETF).

SIP is used for creating, modifying and terminating sessions between endpoints. It is based on the existing protocols like SMTP and HTTP, and uses a text based syntax that is comparable to HTTP uses in web addresses. A web address is comparable to a telephone number in a SIP network. SIP also provides a mobility function to the users. It also supports multiple media sessions during a single call hence users can - share a game, use instant message (IM), and talk at the same time. SIP works with most protocols like RTP, Session Description Protocol (SDP), and Session Announcement Protocol (SAP). SIP works on a client server architecture, where the clients are referred to as User Agents (UA). Although most VoIP implementations today use the H.323 protocol for IP services, SIP is gaining more acceptance in the network telephony market due partly to its flexibility and lower implementation costs. It is possible to use each protocol alone or both protocols within the same network in order to provide universal connectivity. [24, 25]

2.2 Related Works

Improving, optimizing and examining voice traffic over data networks have been an important issue to developers and researchers, various schemes have been suggested depending on analyses from simulated traffic and real word.

A. Durantini, M. Petracca, and F. Ananasso evaluate the WiMAX performance and capabilities in [26] with a real fixed WiMAX test-bed when transmitting Video on Demand (VoD), video streaming and web traffic with different QoS service cl asses(i.e. rtPS, ertPS, UGS..etc.), within different line of sight (LoS) and Non-line-of-sight (NLoS) conditions in fixed and nomadic scenarios. Additionally, it also assesses the difference between different modulation schemes. The results shows that higher throughputs is achieved with more complex modulation schemes and that the rtPS is the most suitable service class for video transmissions. However, these measurements focus only in QoS metrics, such as delay and throughput. This work don't mention VoIP traffic, it just evaluate video and web traffic.

Some researches consider the large deployment of WiMAX and Universal Mobile Telecommunications System (UMTS) networks and the promising integration of the two, they study their QoS differences and possible ways to resolve the differences between QoS models. The researchers, *S. Jadhav, H. Zhang, and Z. Huang* focused on UMTS networks based on wideband Code Division Multiple Access (CDMA) technology – a third generation telecommunication system that contains improved performance and quality of service that contribute to the development of human communication. In a compare between these network and WIMAX using OPNET simulator tool by analyzing various important critical parameters such as mean opinion score (MOS), end to end delay, jitter and packet delay variation. A conclusion is reached that WIMAX is a better technology to support VoIP applications compared to UMTS. [3]

Nevertheless this evaluation is faced with low number of simultaneous user's limitation and the main focus is on the comparison between WiMAX and UMTS, leaving behind the comparison between different QoS service classes.

I. Aldmour in research [27] aimed to compare LTE with WiMAX. It also discusses the factors and QoS parameters such as spectrum allocation, inter-carrier spacing which led to LTE winning as the technology for near future public networks. Also foresees future directions of both technologies and the alternatives that WiMAX technology has. The draw conclusion is broad with no precise QoS models compare also with no specific application (i.e. data, voice...etc.).

In the same track *D. Zvikhachevskiy, J. M. Sultan, and K. Dimyati* 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 rtPS QoS 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. [28]

If there is a large number of WiFi users, they have to be connected to the LTE BE QoS. This unique hybrid network can support up to more than 30 WiFi users where at this point the WiFi BE user's throughput is better than that of the WiFi rtPS user.

Hence, it can be concluded that although BE QoS is the cheapest pricing or probably the most unwanted QoS model, it still possesses satisfying network accomplishment.

Chapter Three

Simulation Setup

In this project, OPNET Modeler 14.5 is used to simulate 3 different network topologies (light network size, medium network size, and high network size). Any topology has contained two scenarios (UGS scenario and ertPS scenario) as categorized in the following sections. The WiMAX model and its corresponding object palette are used. These sets are to study the average throughput, average delay and average jitter of ertPS, and UGS service classes to VoIP system over WiMAX traffics.

3.1 An Overview of the OPNET

OPNET (Optimized Network Engineering Tool) is a network simulator that provides virtual environment.

3.1.1 Why OPNET modeler?

- Good for performance study of existing systems based on users conditions.
- Helpful in evaluation of designs for new network models and architectures
- Pre-defined network models and design exists for user education and development purpose.

3.2 Network Topologies

The WiMAX access network model is made up of seven cells and an Internet Protocol (IP) backbone. Table 3-1 illustrate the parameters that are used for WiMAX model.

Table 3-1:Simulation Parameters

Cell Radius	30 kilometers
No. of Base Stations	One per cell
Base Station Power	10 Watt
Mobility of Nodes	Mobile
Encoder Scheme	G.711
Simulation Time	30 minutes

In this simulation setup, VoIP service traffic and FTP background traffic are considered and three experiments are performed as the following:

3.2.1 Light Network Size

Figure 3-1 shows the light network size which is contained Three Subscriber Stations (SS) in each cell. The total number of SS is **21** mobile nodes.

3.2.2 Medium Network Size

Figure 3-2 shows the medium network size which is contained seven Subscriber Stations (SS) in each cell. The total number of SS is **49** mobile nodes.

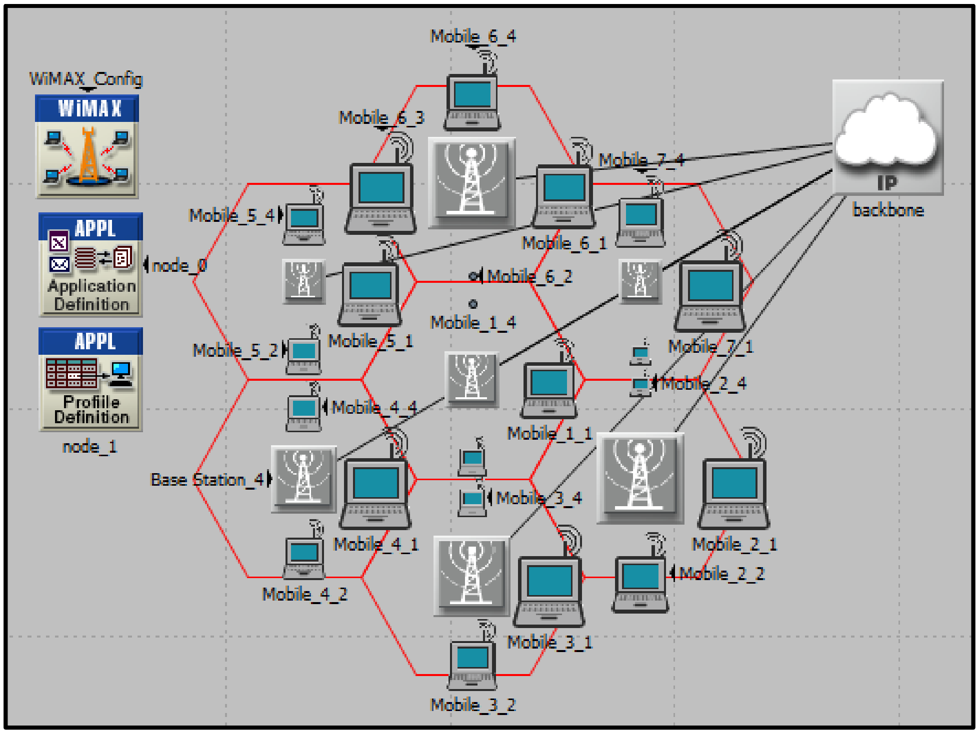

Figure 3-1: WiMAX Light Network Size

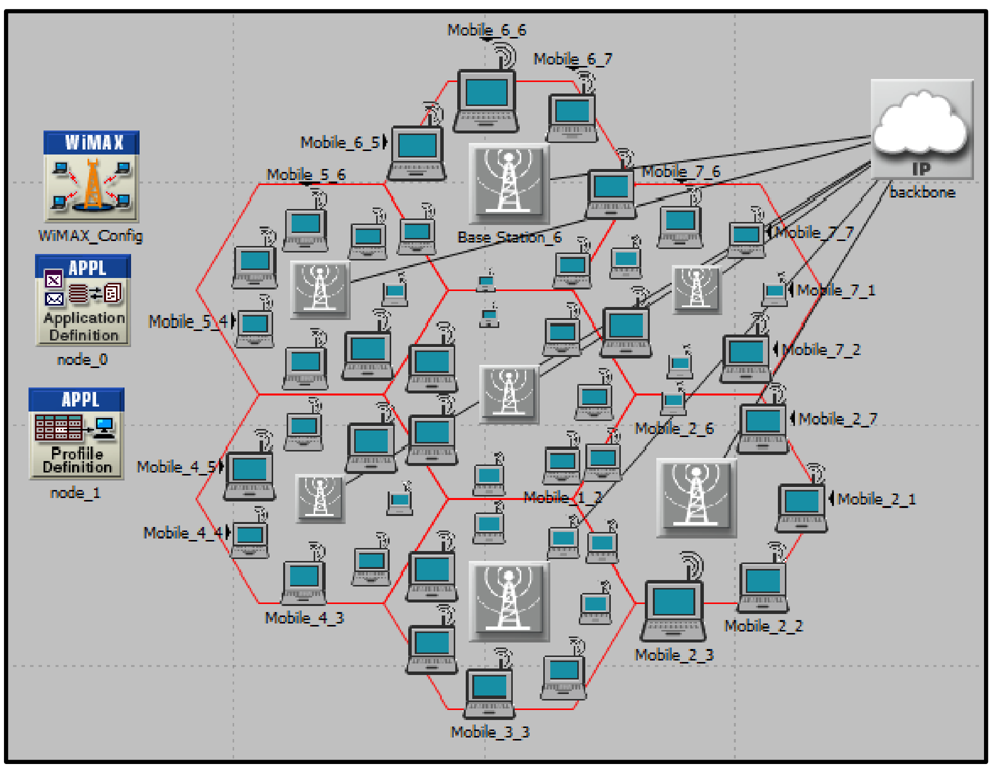

Figure 3-2: WiMAX Medium Network Size

3.2.3 High Network Size

Figure 3-3 shows the high network size which is contained twelve Subscriber Stations (SS) in each cell. The total number of SS is **84** mobile nodes.

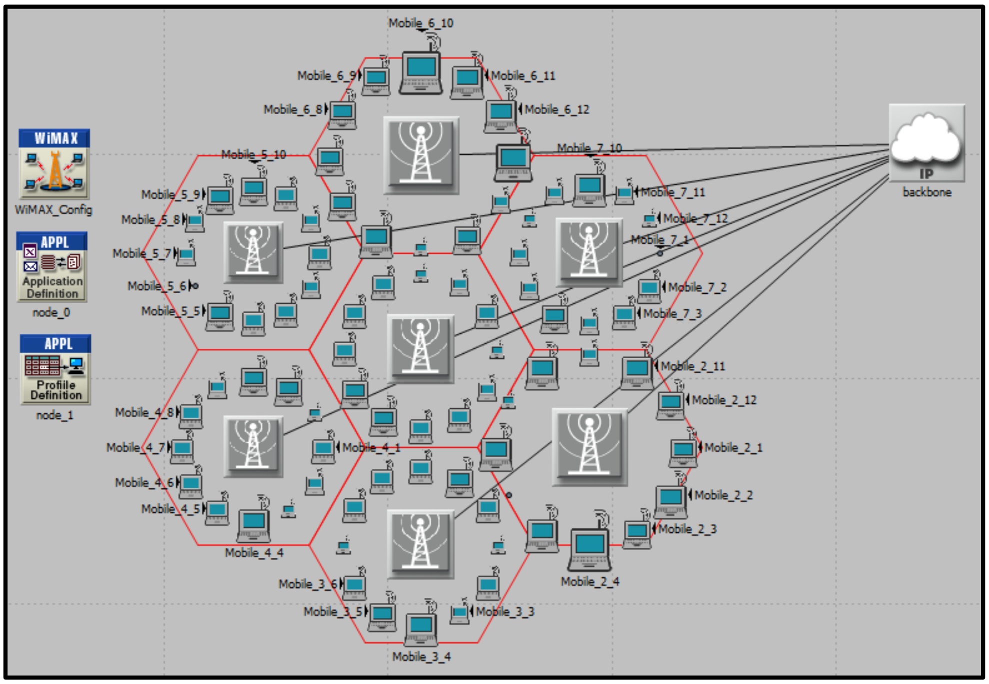

Figure 3-3: WiMAX high Network Size

3.3 Traffics Model

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

3.3.1 Application configurations

Two applications named VoIP-app and Ftp-app were created in the Application definition node. The settings are shown in figure 3-4.

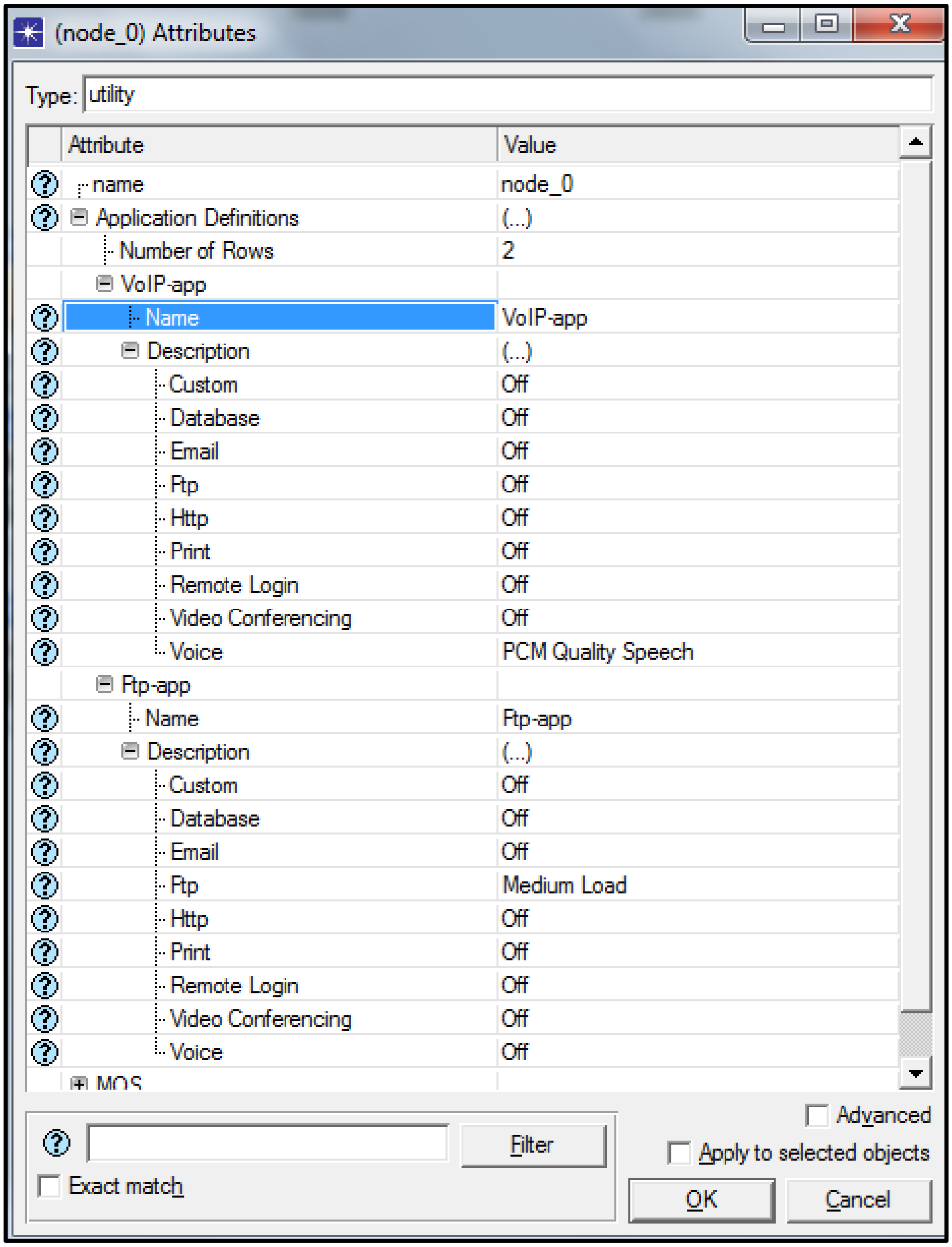

Figure 3-4: Application Attributes

3.3.2 Profile Configurations

Profiles called VoIP-pro and Ftp-pro were created in the Profile definition node. This profiles are used by the user nodes in order to generate traffic. The profile was defined to use the VoIP application and it was set to begin at the start of the simulation and continue until the simulation is completed. Figure 3-5 shows the profile configuration menu.

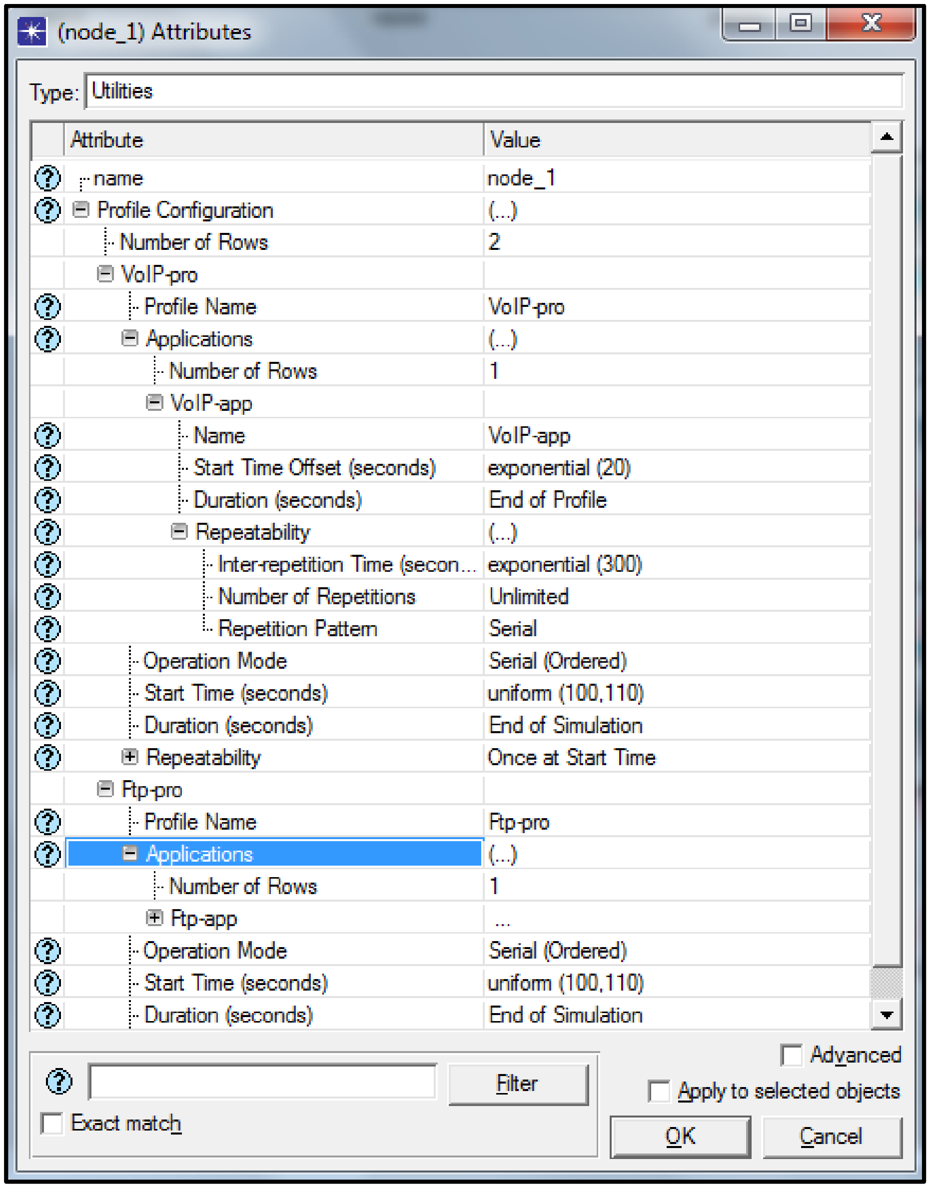

Figure 3-5: Profile Attributes

3.4 WiMAX Nodes Configurations

Given that, any experiment size is conducted two sets of simulations scenarios, the aim of each one of these sets is to study the average throughput, average delay and average jitter of ertPS, and UGS service classes to VoIP system over WiMAX traffics. Configurations of WiMAX_Config node, base stations, and mobile nodes are needed.

3.4.1 WiMAX_Config node

The WiMAX_Config node is configured. 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 3-6 and figure 3-7.

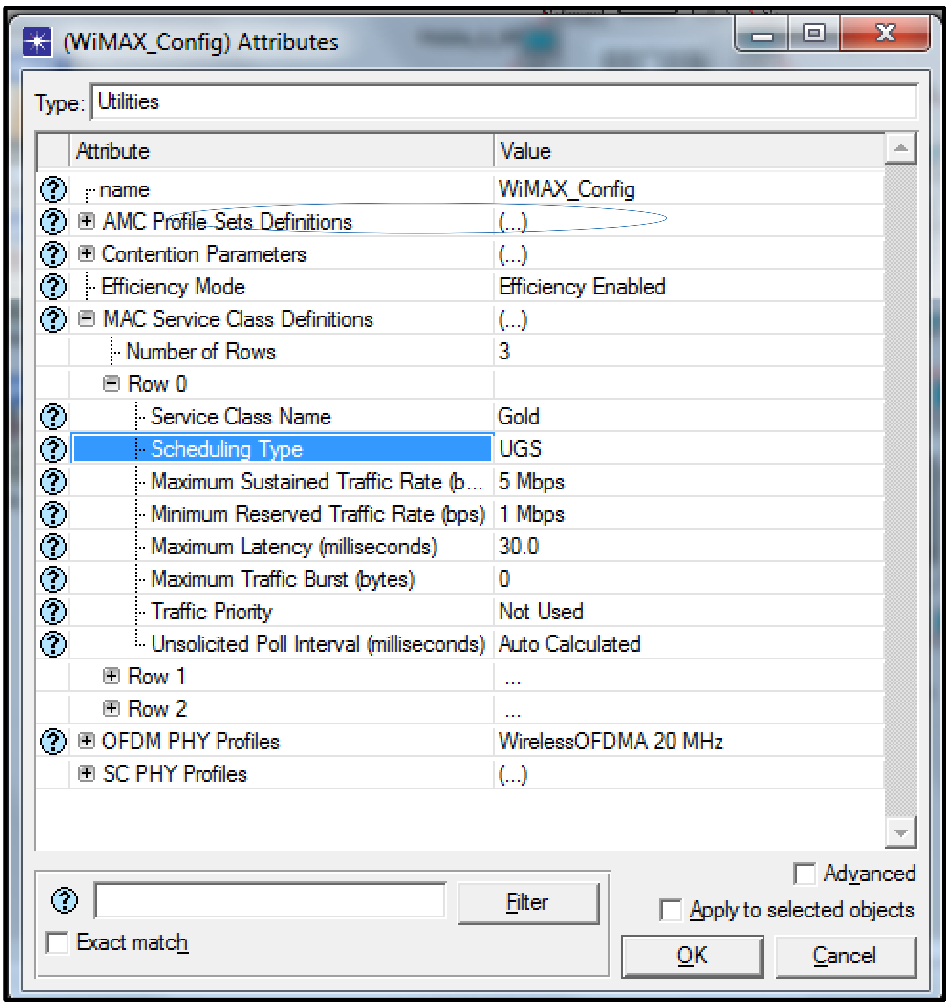

Figure 3-6: WiMAX Configuration: UGS Scheduling Type

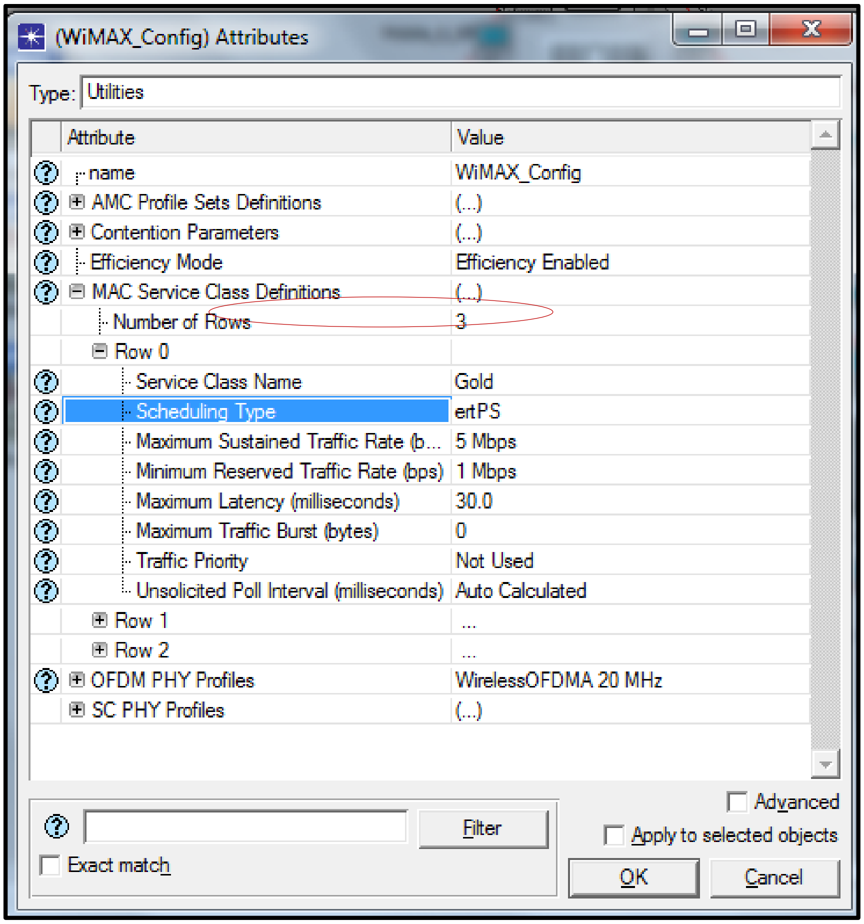

Figure 3-7: WiMAX Configuration: ertPS Scheduling Type

3.4.2 WiMAX Base Station

There are typical values that are commonly used in real life situations. Two rows are added in to represent main and background Traffic (i.e. VoIP, FTP) as shown in figure 3-8 Interactive Voice, Background respectively. Service Class Name for VoIP is set to Gold.

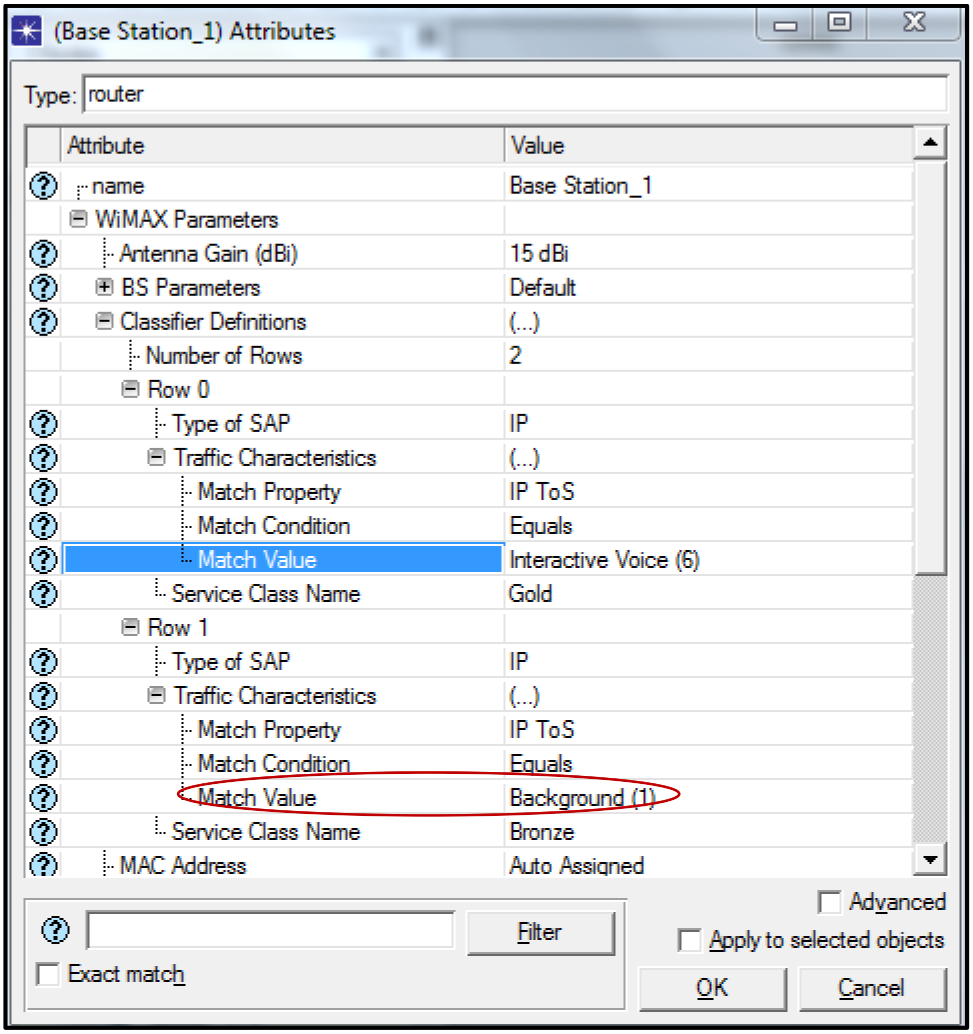

Figure 3-8: WiMAX Base Station (BS) Parameters Configuration

3.4.3 Mobile Node

The key parameters that were modified are trajectory, WiMAX parameters, and applications. For SS parameters, 2 rows are added to represent main and background Traffic (i.e. VoIP, FTP) as shown in figure 3-9 Interactive Voice, Background respectively. Service Class Name for VoIP is set to Gold.

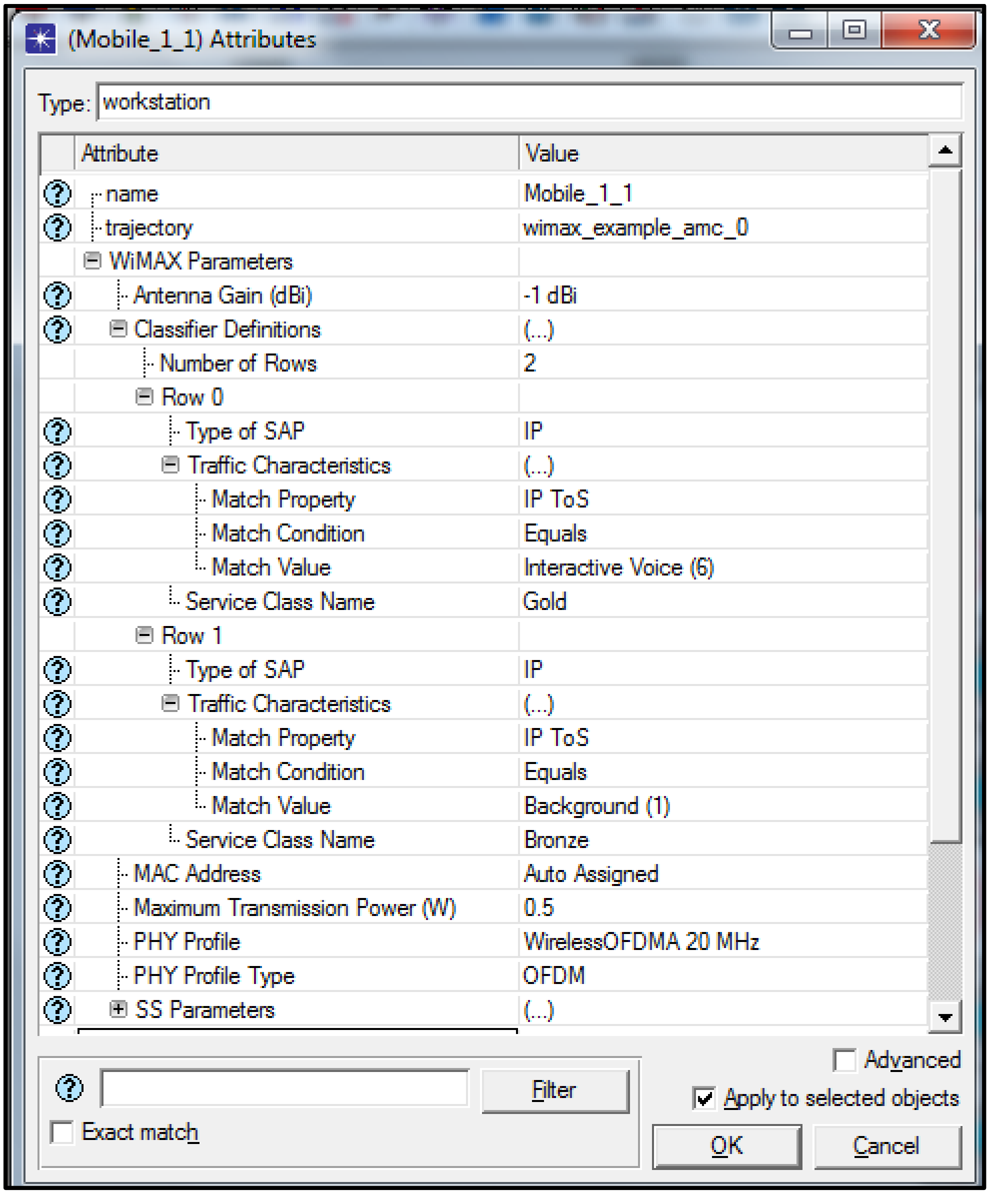

Figure 3-9: WiMAX Mobile Node: WiMAX Parameters Configuration

Figure3-10 shows the sitting of WiMAX Application Parameters, VoIP and ftp profiles are added to the supported profiles section and applications are added to the supported services section

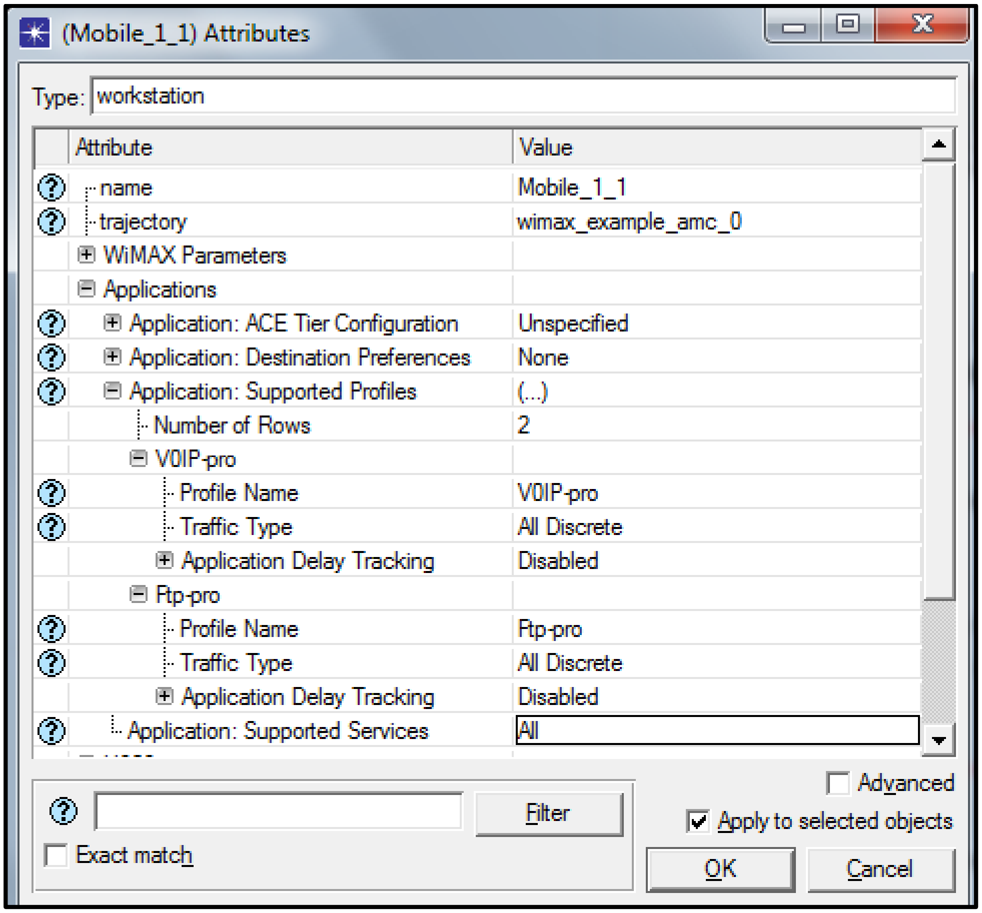

Figure 3-10: WiMAX Mobile Node: Application Parameters Configurations

Finally, the simulation is run for 30 minutes.

Chapter Four

Simulation Results and Analysis

Here a detailed comparative analysis of the modeled networks performance of voice over IP using WiMAX network via extensive network research simulation methodology were carried out. The results of the OPNET simulation experiments are presented in the following sections.

4.1 Topology One: Light Network

This topology scenario was simulated to study the effect of different QoS classes on VoIP services over WiMAX light networks. The QoS classes used for the investigation include UGS, and ertPS. The amount of ftp background traffic sent is shown in figure 4-1 and later VoIP results obtained from the simulation is shown below.

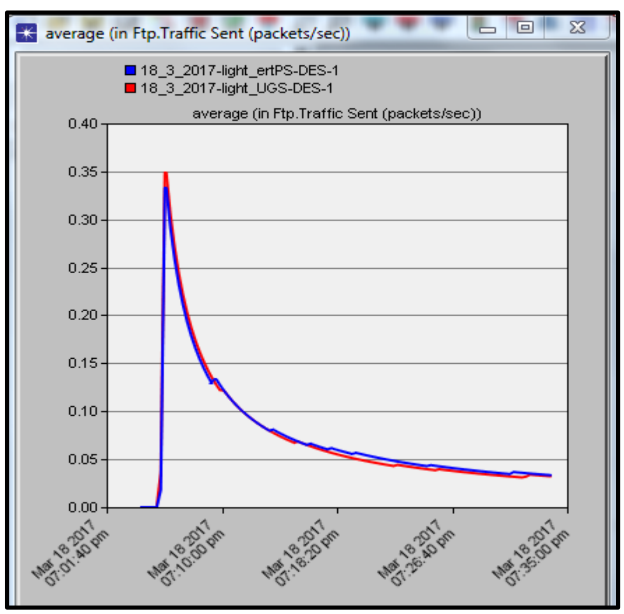

Figure 4-1: FTP Background Traffic (Light Network)

4.1.1 VoIP End to End Delay of Light Network

The VoIP traffic suffered packet end to end delay as clearly witnessed in Figure 4-2, ertPS class produced the higher packet end to end delay of 0.0998 sec and UGS class yielding the lower value of 0.0952 sec.

4.1.2 VoIP packet delay variation of Light Network

Figure 4-3 Compares voice packet delay variation (jitter) levels when UGS and ertPS are applied. While the number of nodes is 21, UGS class returned the lower voice jitter of approximately 0.000067 sec and ertPS with the higher value of 0.000077 sec.

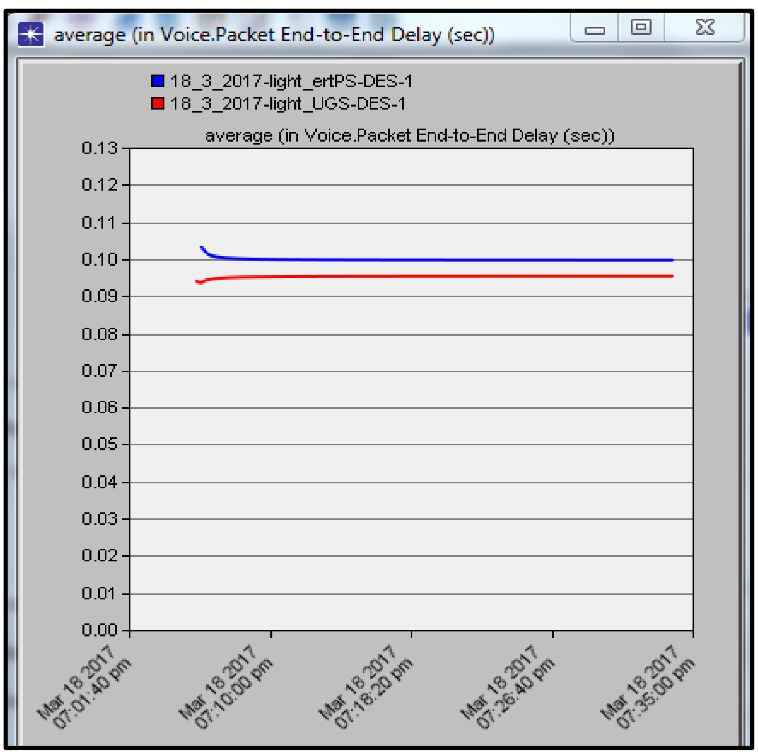

Figure 4-2: Average Voice End to End Delay (Light Network)

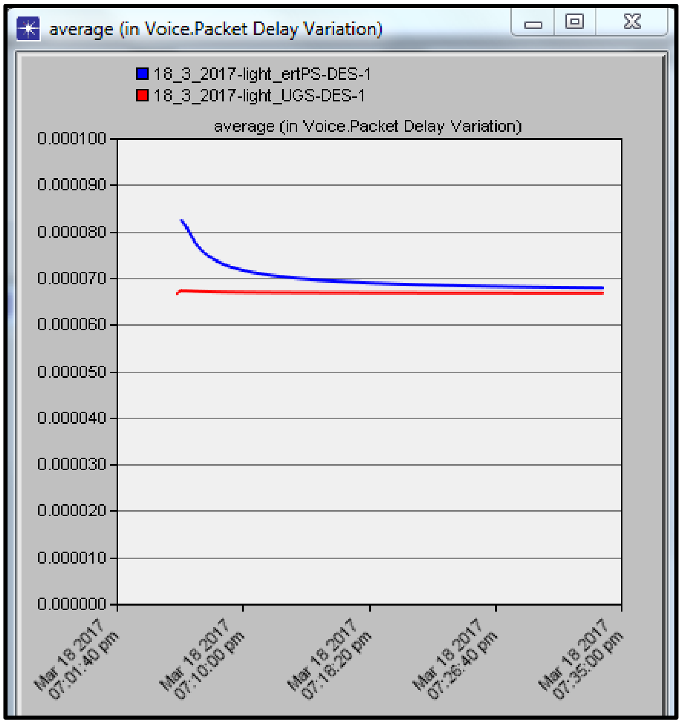

Figure 4-3: Average Voice Packet Delay Variation (Light Network)

4.1.3 Throughput of Light Network

The overall throughput of UGS service flow is higher than ertPS service flow as shown in figure 4-4. For 21 nodes, the value of throughput for UGS service flow is 462.3542 packets/sec while 381.1340 packets/sec for ertPS service class.

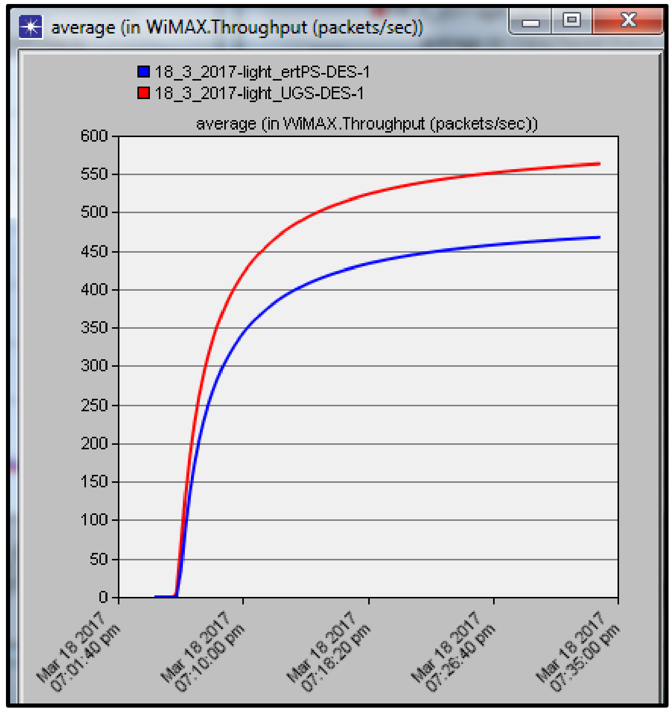

Figure 4-4: Average Network Throughput (Light Network)

4.2 Topology Two: Medium Network

In This topology scenarios, the number of nodes is increased to 49 nodes to study the effect UGS and ertPS QoS classes on VoIP services over WiMAX Medium networks. Figure 4-5 represents the ftp background traffic sent .VoIP results obtained from the simulation are shown later on.

4.2.1 VoIP End to End Delay of Medium Network

The average voice end to end delay presented in Figure 4-6, UGS had the lower end to end delay of 0.094 sec while ertPS recorded higher delay of 0.105 sec.

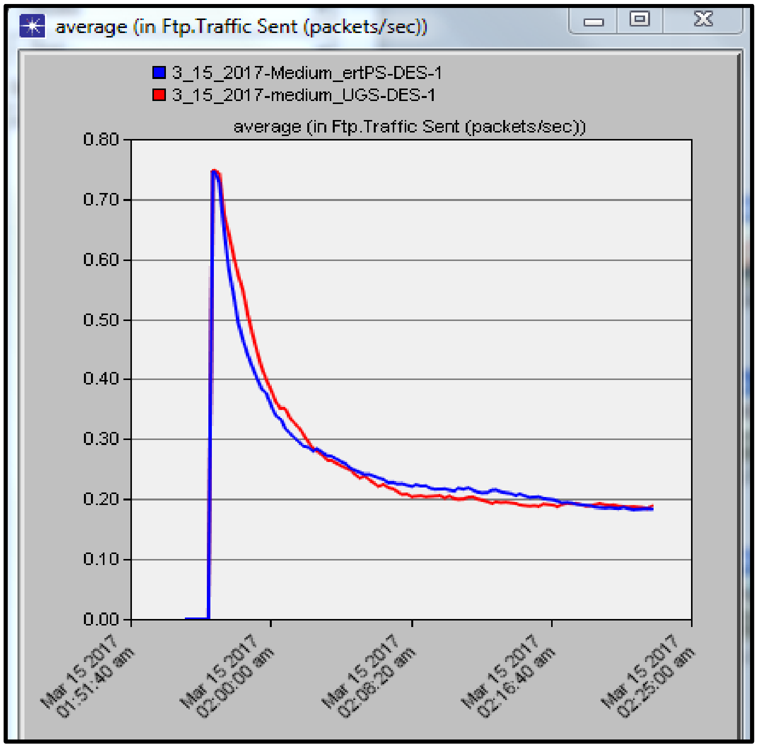

Figure 4-5: FTP Background Traffic Sent (Medium Network)

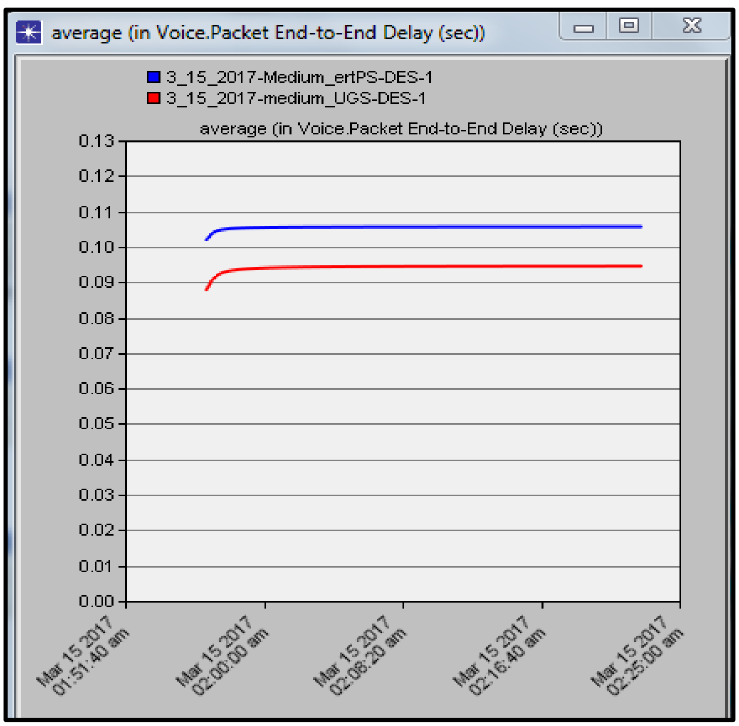

Figure 4-6: Average Voice End to End Delay (Medium Network)

4.2.2 VoIP packet delay variation of Medium Network

In Figure 4-7 where the number of nodes is 47, the average jitter does not vary as much as the number of nodes increases. In addition to that, the value is very small.

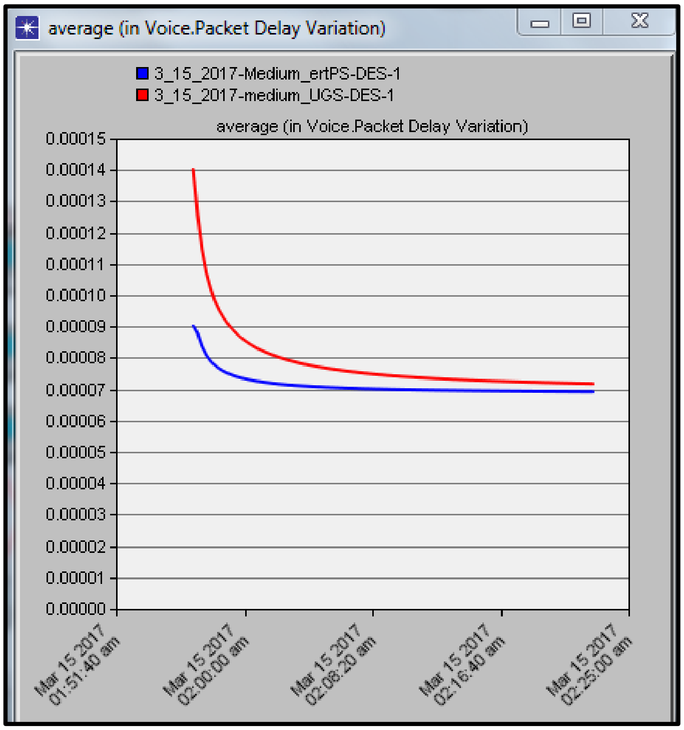

Figure 4-7: Average Voice Delay Variation (Medium Network)

4.2.3 Throughput of Medium Network

UGS service class yields the higher network throughput than ertPS as seen in figure 4-8, total throughput rates of 827.88 packets/sec and 756.80packet/sec respectively.

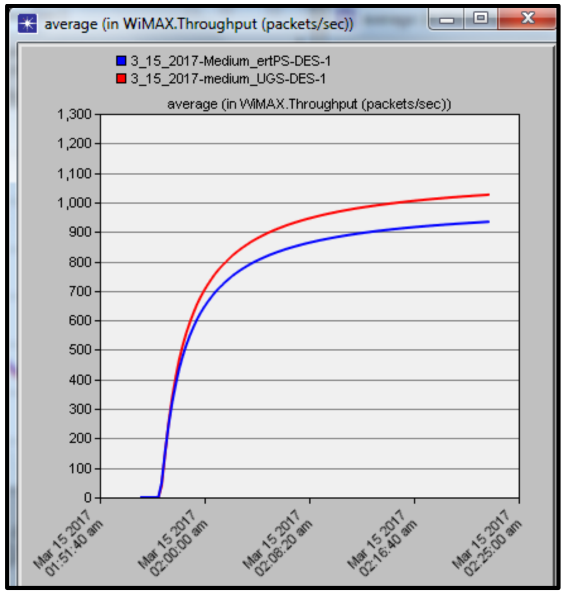

Figure 4-8: Average Network Throughput (Medium Network)

4.3 Topology Three: High Network

This topology have scenarios that shows the average parameters for ertPS and UGS classes when the number of nodes increase to 84. The amount of ftp background traffic sent is shown in figure 4-9 and later VoIP results obtained from the simulation is shown below.

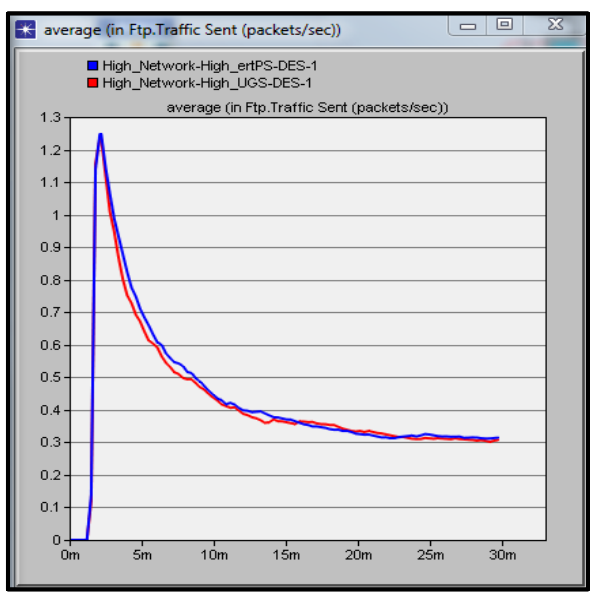

Figure 4-9: FTP Background Traffic Sent (High Network)

4.3.1 VoIP End to End Delay of High Network

The VoIP traffic suffered packet end to end delay as clearly witnessed in Figure 4-10, ertPS class produced the higher packet end to end delay of 0.116 sec and UGS class yielding the lower value of 0.107 sec

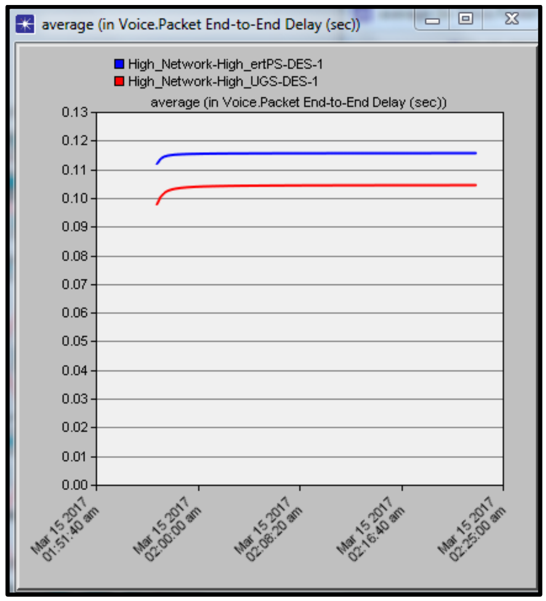

Figure 4-10: Average Voice End to End Delay (High Network)

4.3.2 VoIP packet delay variation of High Network

The average packet delay variation does not vary as much as the number of nodes increases and very small values approximately 0.0 sec as explained earlier. Figure 4-11 shows average VoIP variation when the number of nodes is raised to 84.

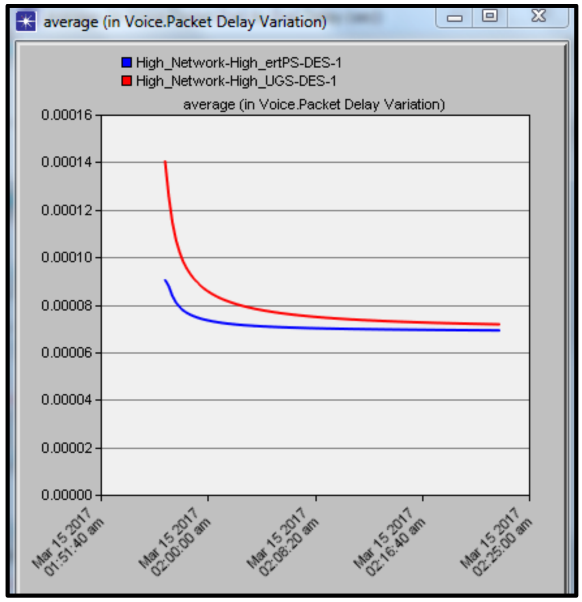

Figure 4-11: Average Voice Delay Variation (High Network)

4.3.3 Throughput of High Network

The throughput of UGS is still higher than ertPS class 1.284.3110 packets/sec and 870.547 packets/sec respectively.

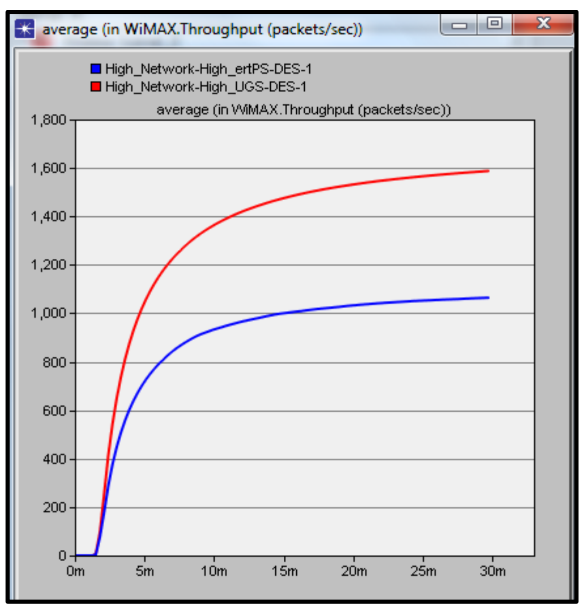

Figure 4-12: Average Network Throughput (High Network)

4.4 Summary of Results

In all topologies applied and studied, the general network QoS parameters are in the acceptable range values refer to table 2-4. The voice parameters values are within the range of acceptable voice jitter threshold ($\leq$ 0.5 ms), acceptable packet end to end delay threshold ($\leq$ 140 ms), and good throughput.

Regard to the previous figures, it’s observed that UGS service class has the better throughput, lowest average jitter and lowest end to end delay. When the number of nodes is 21, end to end delay and throughput are enhanced by 4.6% and 21.3% respectively in UGS compare to ertPS. These values are increased to 7.7% for delay and 47.5% for the throughput when the nodes are rise to be 84.

Actually, UGS service class is dedicated to handle real-time service flows. The packets are generated in fixed sizes at regular interval, which is the case for VoIP. The bandwidth can be periodically requested in the ertPS service class instead of fixed.

Chapter Five

Conclusion and Recommendations

5.1 Conclusion

In this research, OPNET modeler 14.5 network simulation tool is used to investigate and study VoIP applications performance over 4G network. Several network performance parameters including voice packet ETE delay, voice delay variation (jitter), and WiMAX network throughput, are used to determine the quality of VoIP calls that can be guaranteed with good QoS over WiMAX network.

The research study investigated VoIP system performance under two different QoS classes and three different network sizes. UGS and ertPS classes are studied using simulation methodology in order to determine the best suitable class that will yield optimal voice quality for VoIP systems over WiMAX network.

The results of network research simulation exercise proved that when the number of nodes are raised between 21 to 84 nodes, UGS has enhanced in range 4.4% to 7.7% for the delay and 21% to 47% for the throughput opposed to ertPS. It is concluded that, in different network sizes, Light, Medium, and High, the service class UGS provide better VoIP quality.

5.2 Recommendations

VoIP was considered in the current analysis. Further analysis could be done for other applications including video streaming, video telephony which combines video traffic and VoIP traffic, HTTP traffic etc.

Furthermore, VoIP and FTP work can also be simulated in LTE as 4G network, and then compared the simulation results with that on WiMAX network. In addition, combine various applications together on LTE network and analyze the effect to the performance.

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