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Performance Evaluation of Quality of Service Classes in WiMAX Networks

تقييم أداء فئات جودة الخدمة في شبكات الوايماكس

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

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استهلال

قال تعالى:

أعوذ بالله من الشيطان الرجيم

((قل هل يستوي الذين يعلمون والذين لا يعلمون))

[سورةالزمر - 9]

صدق الله العظيم

Dedication

I would like to dedicate my research to my family, my friends and to all persons whose help and support me.

Acknowledgement

I am thankful to my supervisor Dr. Fath Earthman Ismael for all the support and help he gave me .I appreciate him forthe valuable advice, guidance and assistance in practical writing of my dissertation. Without his guidance and constant feedback this Research would not have been achievable.

Also I am really grateful that this research cannot be completed without the effort and help from the people who’s around me, Special thanks for a special friends

My family thanksfor everything. I warmly thank and appreciate my parents and my husband for their great support in all aspects of my life.

I would like to say thank god for the gift you that gave me,Haga Fatima Abo Alreesh thank you for the great effort, thewords only cannot be express my gratitude for you

Abstract

Nowadays, the desire of internet access has influenced quite a large number of users to access high quality video application. In wireless medium, video applications still have high resource requirements, for example, bandwidth and traffic priority. For this challenges, The IEEE 802.16e defined five different Quality of Service (QoS) classes including UGS, ertPS ,rtPS, nrtPS and BE. It is well known that UGS is designated to support QoS for voice without silence suppression and video conference applications and ertps is designed also to support real time application but with silence suppression, as for rtps is designed to support video and audio streaming. The other two classes have different applications .In this research we investigate the performance of UGS, ertps and rtPS QoS classes in terms of multimedia applications such as videoconferencing and voip over WiMAX network scenarios .There were six scenarios have been implemented at the same topology but different traffic type, three of them used video conferencing application and others used voip application, also a certain combination of users that are allocated a QoS and a selected application. These selected application had priority of allocating resource by QOS classes for each scenario .The OPNET modeler 14.5 simulator is employed for simulation purposes in order to evaluate the performance of UGS, ertps, and rtPS of video conferencing and voip applications. The results show that the user with UGS QoS class could provide higher throughput which 37% and 33% more than rtPS class , lower end to end delay which is 66% and 52% more than rtPS class ,finally lower packet delay variation which are 80% and 68% more than rtps class for video and VoIP respectively. So UGS offered the highest performance.

المستخلص

في الوقت الحاضر ، تؤثر رغبة الوصول إلى الإنترنت على عدد كبير من المستخدمين للوصول إلى تطبيق الفيديو عالي الجودة. في الوسائط اللاسلكية ، لا تزال تطبيقات الفيديو تحتوي على متطلبات عالية الموارد ، على سبيل المثال ، عرض النطاق الترددي وأولوية حركة المرور ,بسبب هذه التحديات.حددت منظمة معهد مهندسي الكهرباء والاكترونيات خمسة فئات مختلفة لجودة الخدمةوهي خدمة المنحة الغير مطلوبة,وخدمة الانتخاب للوقت الحقيقي, وخدمة الانتخاب للوقت الغير حقيقي,خدمة الانتخاب الممتدة للوقت الحقيقي وخدمة أفضل جهد. من المعروف أن خدمة المنحة الغير مطلوبة مصممة لدعم جودة الخدمة من أجل الصوت بدون خاصيه كبح صمتوتطبيقات مؤتمر الفيديو وتم تصميم خدمة الانتخاب الممتدة للوقت الحقيقي أيضًا لدعم التطبيقات في الوقت الحقيقي ولكن مع قمع الصمت ، كما تم تصميمخدمة الانتخاب للوقت الحقيقي لدعم تدفق الفيديو والصوت. والتصنيفان الأخران لهما تطبيقات مختلفة.في هذا البحث ، نقوم بالتحري عن أداء فئات جودة الخدمة لكل نوع من هذه الفئات من حيث تطبيقات الوسائط المتعددة مثل مؤتمرات الفيديو و الصوت عبر بروتوكول الانترنت عبر سيناريوهات شبكة الوايماكس. لقد تم تنفيذ ستة سيناريوهات في نفس الهيكلية مع نوعان مختلفان تدفق المرور ، ثلاثة منهم يستخدم تطبيق مؤتمرات الفيديو والمتبقية تستخدم تطبيق الاتصالات عبر بروتوكول الإنترنت ، وأيضا مجموعة معينة من المستخدمين الذين يتم تخصيص لهم جوده خدمة وتطبيق محدد. هذه التطبيقات المختارة لها أولوية في تخصيص الموارد حسب نوع الفئات الممنوحة لكل سيناريو. يتم استخدام محاكاة نموذج اداء هندسة الشبكة الافضل الذي يدعى بأوبنيت لأغراض المحاكاة من أجل تقييم أداء كل من الفئات من مؤتمرات الفيديو وتطبيقاتالصوت عير بروتوكول الانترنت. تُظهر النتائج أن المستخدم المزود بفئة الخدمة الغير المطلوبة يمكن أن يوفر أعلى معدل انتاجية والذي يزيد عن ما توفره فئة خدمة الانتخاب الممتدة ب 37٪ و 33٪ لكل من الفيديووالصوت عبر بروتوكول الانترنت على التوالي وكذلك أقل تأجيل ينخفض إلى 66٪ و 52٪ لكل من للفيديو والصوت عبر بروتوكول الانترنت على التوالي ، وأخيرًا يكون أقل اختلاف تأخر في الرزمة الذي ينخفض الي 80 ٪ و 68 ٪لكل من الفيديووالصوت عبر بروتوكول الانترنت على التوالي . لذلك قدمت فئة الخدمة الغير المطلوبة أعلى مستوى أداء .

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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
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
PMP	Point-to-Multipoint
PTP	Point-to-Point
QAM	Quadrature Amplitude Modulation
QoS	Quality of service
QPSK	Quadrature Phase-Shift Keying
RAN	Radio Access Network
UGS	Unsolicited Grantee Service

Chapter One

Introduction

1.1 Preface

Mobile wireless communication, because of its adaptability and flexibility has now become a necessity. A user not only wants the basic voice functionality of mobile wireless devices, but also multimedia and data communication applications. Over the years, with the improvement of technology, the bandwidth over the wireless network has been increasing, at a measured pace. Over the past few decades, different modulation schemes have been developed to increase the data rate supported in wireless networks thus supporting a higher data rate. Technologies like Wi-MAX and other 3G technologies can theoretically support up to 74Mbps per channel[1].Real-time video sessions, such as IPTV and video streaming, will be among the most important applications in future networking systems. The delivery of those sessions based on Quality of Service (QoS) techniques assures packet differentiation and indicate the impact of multimedia traffic on the network performance, but do not reflect the user perception[2].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].Network is that it provides data rates up to 100 Mbps for downlinks and 50 Mbps for uplinks, and throughput of 50-144 Mbps [4].WiMAX and LTE are both broadband wireless access (BWA) network technologies, aiming to play a crucial role in 4G and beyond “all-IP” broadband wireless access technologies progress. LTE is an evolved 3GPP technology, and as WiMAX, was developed to support higher number of users with higher data rates, coverage and availability[5].

1.2 Problem Statement

There are many factors that affect the quality of services such as QOS classes that determine the priority of allocating resources to each service. There are five QOS classes in WIMAX need to be evaluate in various cases (used different applications VoIP and video conferencing).

1.3 Proposed Solution

This thesis evaluate the performance of three QOS classes such as UGS,ertps and rtps using a simulation in OPNET .the performance of video conferencing and voip are evaluated considering End to End delay ,Packet delay variation and Throughput under the same topology .

1.4 Aim and Objectives

The general aim of this research is to simulate WiMAX model and evaluate the performance of QoS classes implemented for video conference and voip traffic in various network scenarios which are video_UGS ,video_ertps,video_rtps,voip_UGS,voip_.ertps andvoip_.rtps. also aim 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 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 is an introduction that states the problem statement, proposed solutions and the objectives of the research .states the problem and proposed solutions .
- Chapter Two is description of background to understand the proposed system and some examples of related works.
- Chapter Three explain the methodology that followed to get concern result.

- Chapter Four define the 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 is 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

Technologies of mobile communication are often divided into generations, with 1G being the analog mobile radio systems, 2G the first digital mobile systems, and 3G the first mobile systems are handling broadband data. This continuing race of increasing sequence numbers of mobile system generations. What is important is the actual system capabilities and how they have evolved[6].

2.1.1.1 First Generation

The first generation cellular networks were invented in the 1980s. The thought 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 subscribers to be supported in a given area. All systemsfor first generation were analog systems popularly known as early cellular phone technology working in the frequency band of 150 MHz[6].The Technology used in the 1st Generation is Advanced Mobile Phone System (AMPS) cellular technology which uses discrete frequencies to be held. There is a need for proper band width in this technique for a many number of users. The basic disadvantage of 1G is the quality of voice[7].

2.1.1.2 Second Generation

The Second Generation (2G) cellular telecom networks were commercially launched on the Global System for Mobile (GSM) standard in Finland by *Radiolinja* in 1990. 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 such as Voice Mail Service (VMS) was also and value added service also there was a new feature called Short Message service (SMS) in 2G that use Band width range of 30-200KHZ[6][7].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 appropriate for web browsing, multimedia and real time applications[8].Some key advantages 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. Farther more, 2G was enhanced to 2.5G. This is a technology which was introduced in 1990's. It uses a technology 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 2.75G[7].

2.1.1.3 Third Generation

International Mobile Telecommunications-2000 (IMT-- 2000), better 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 human can contact with other human located in any part of the world and can even send messages too[6][9].3G technology 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[10].

2.1.1.4 Fourth Generation

Third generation (3G) mobile networks faces a new competitor; so called 4G. An amazing new network may be even more profitable. The goal of 4G is to replace the current increasing of core cellular networks, with a single worldwide cellular core network based on standard IP for control and media[10].In March 2008, the International Telecommunications Union-Radio communications sector (ITU-R) determine a set of requirements for 4G standards, called the International Mobile Telecommunications Advanced (IMT-Advanced) specification, setting peak speed requirements for 4G service at 100 megabits per second (Mbit/s) for high mobility communication (such as from trains and cars) and 1 gigabit per second (Gbit/s) for low mobility communication (such as pedestrians and stationary users). Extra to voice and other 3G services 4G system provides ultra-broadband network access to mobile devices. There are many applications such as IP telephony, HD Mobile Television, video conferencing, online gaming 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[6].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 data networks must be available to the subscriber 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 just for humans as have been the rule in previous systems, but also to any that needs to communicate and response[8][11].

- **A multi-service platform** is an important 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 lose its weight in the overall user bill with the raise of more and more data services[8][11].
- **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[8][11].

To accomplish the proposed goals, a very flexible network that aggregates different radio access technologies, must be created. This network must provide high bandwidth, from 50-100 Mbps for high mobility users, to 1Gbps for low mobility users, technologies that permit fast handoffs, an efficient delivery system over the different wireless technologies available, and a method of choosing the wireless access from the available ones. Very important to implement 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[12].

2.1.2 WiMAX Overview

The WiMAX is an evolving IEEE standard and 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 the solution of the last-mile 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 many aspects compatibility, technicality, regulatory, and marketing aspects of the WiMAX[13].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)[14][15].There are 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 allows 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 provide data rate of up to 75 Mbps[14][15].

2.1.3 Systems 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 grantee 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[13].

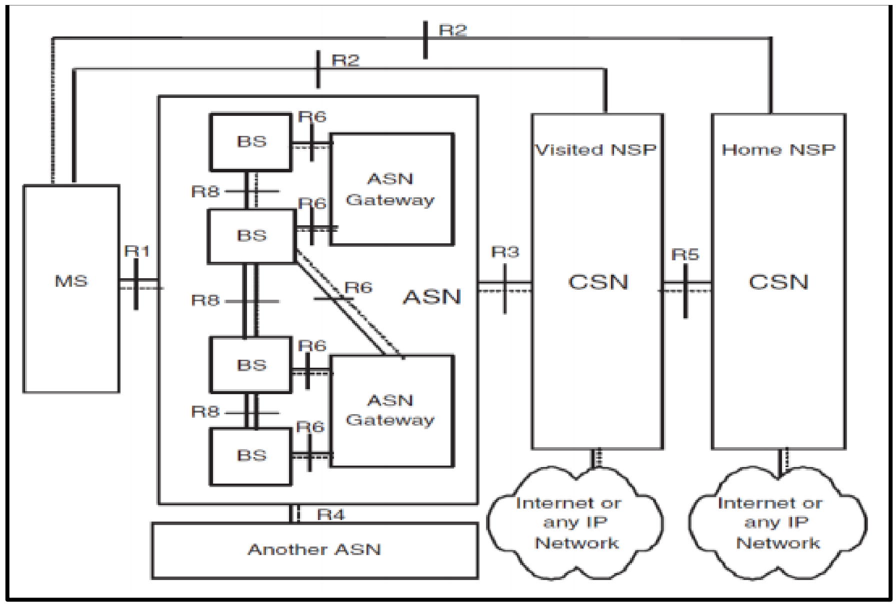

Figure 2. 1:Network Reference Model for WiMAX [13]

- Mobile Station (MS): Used to access the network.

- Access Service Network (ASN): Comprised of ASN Gateways (GWs) and base stations(BSs) to form Radio Access Network (RAN) at the edge[13].

- 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[13].

- Access Service Network Gateway: Acts as layer 2 traffic aggregation point within an ASN and it 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)[13].

- 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[13][16].

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)

2.1.3.2 Air Interface

WiMAX network supportboth ofFrequency Division Duplex (FDD) and Time Division Duplex(TDD) mode in radio access modes, although the previous version of WiMAX just supports TDD mode. Besides, cellular operators prefer to choose the FDD mode since most of current cellular systems depend 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 and the Multi-Input Multi-Output (MIMO) technology is used. WiMAX network defines 2x2 MIMO. Six aspects of air interface standard in table 2-2[17].

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: 2X2 MIMO UL: 2X2 MIMO

2.1.4 Technology of Multiple Accesses

Downlink and uplink transmission in 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[18].

2.1.4.1 OFDMA for WiMAX Uplink/Downlink

OFDMA is define as 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[18].

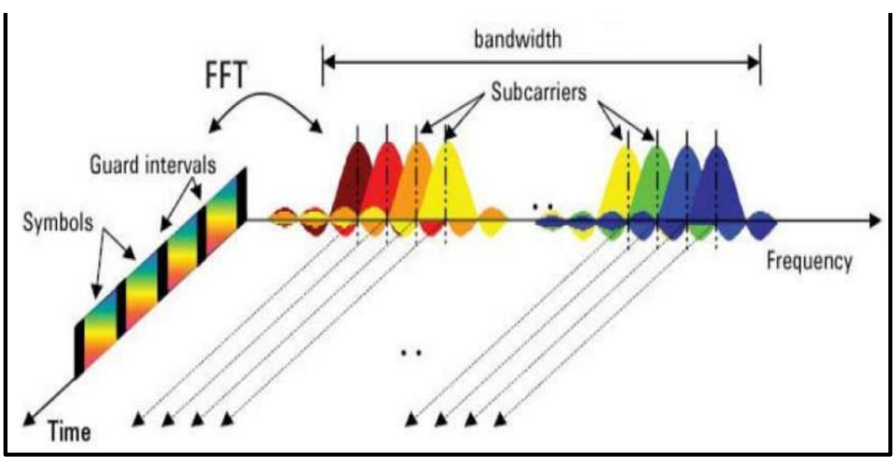

Figure 2. 2: OFDM Signal Representation in Frequency and Time Domain [18]

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 users among different users on the channel. OFDMA provides a robust system with increased capacity and resistance to multipath fading[18].

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[18].

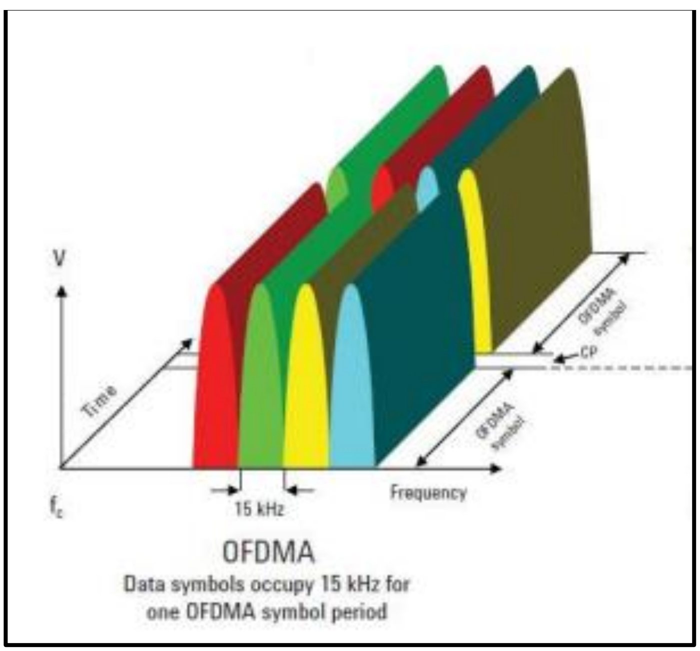

Figure 2. 3: OFDMA Transmitting a Series of Data Symbols [18]

2.1.4.2 Multiple Input Multiple Output (MIMO)

MIMO technology refers to a system having minimum two antennas at the base station as well as at the mobile station. MIMO systems enhance the performance of WiMAX including spatial multiplexing and reduce each of diversity and interference. 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[18].

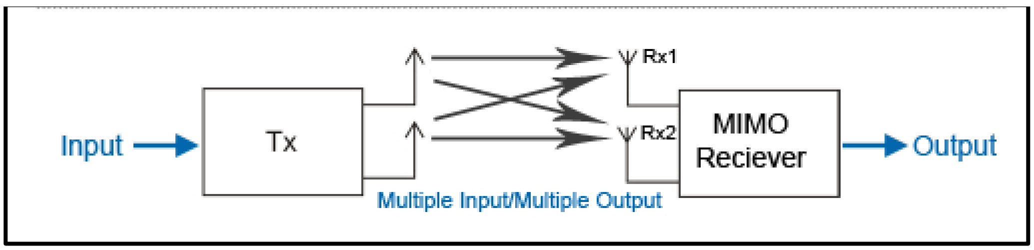

Figure 2. 4: General MIMO System [18]

a. Open loop MIMO System

Open loop techniques increase the range and capacity of WiMAX. these 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[18].

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 Maximum Ratio Transmission (MRT).[16], The Multiple antenna organization chart for WiMAX is shown in Figure 2-5[18].

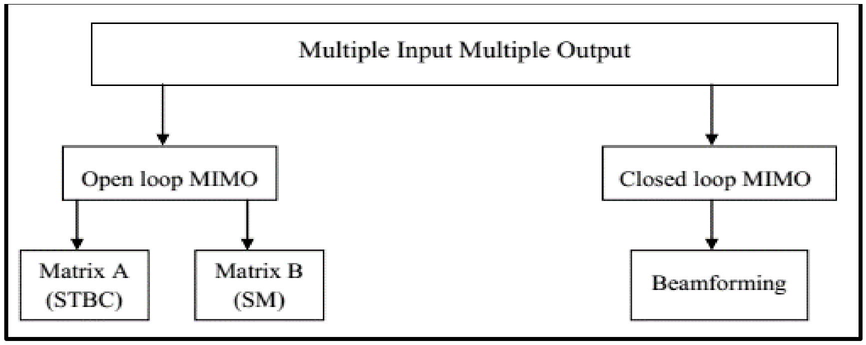

Figure 2. 5: WiMAX Multiple Antenna Implementation Organization Chart [18]

2.1.5 QoS in WiMAX

As the huge rapid growing of real time applications over Internet, it is required to utilize the QoS, which is ensuring the guaranteed service through Internet. , real time applications such as Voice and Video services are bandwidth extensive, which requires less delay to maintain the QoS. It is not always possible to maintain the quality of all requirements. Quality of service is the guarantee of the service-level performance for a data stream from a source to destination [19][20].

2.1.5.1 QoS Service Classes in WiMAX

QOS 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 decrease ng order. Within all these classes of services resources are allocated to manage and satisfy the QoS of higher priority services[21][22].

- **Unsolicited Grant Service (UGS)**

Support 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[13].

- **Real-time Polling Service (rtPS)**

Support 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 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[13].

- **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[13].

- **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[13].

- **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[13].

IEEE 802.16 has five QoS classes Table 2-3 broadly classifies various service classes defined in WiMAX and its applications [21][22].

Table 2. 3: QoS Classes in WiMAX

Service classes	Description	Applications
Unsolicited Grant service (UGS)	For constant Bit rate and delay dependent applications	VOIP and Video conference
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

In this research, the performance analysis for QoS of real time applications video and voip over WiMAX network using OPNET modeler 14.5 is carried out with respect to different scenarios . To better analyze the performance service classes UGS,ertps and rtPS are used .QoS requirements become very important in WiMAX technology to guarantee their performance these parameters are define as below:

- **Delay**: Delay is the time of generation of packet by source to destination reception. So it is the time taken by packet to go across the network. The time can be expressed in seconds. All the delay in the network are called packet end to end delay.

- **Load**: This represents the total load submitted to WiMAXlayers by all higher layers in all WiMAX nodes of thenetwork. It can be represented in bits/sec and packets/sec.
- **Throughput**: Presents the amount of successful datatransferred from one location to another over a specific period of time.
- **Packet End-to-End delay**: Packet end to end delay is also known as time taken for a packet to be transferred from source to destination. End-to-end delay less 150msis considered acceptable. It is the most important parameter in video conferencing because it affects the QoS and degrades the system performance
- **Packet Delay Variation**: In packet delay variation, variation in packet is seen which are transmitted, but not received by receiver in intended time. It can degrade the performance of system. Low packet delay variation is important[23].

2.1.6 Video Conferencing

A video conference is a live, visual connection between two or more people residing in separate locations for the purpose of communication. At its simplest, video conferencing provides transmission of static images and text between two locations. At its most sophisticated, it provides transmission of full-motion video images and high-quality audio between multiple locations. Video conference has stringent requirements on a network with an acceptable end-to-end delay of around 150 ms (including time for encoding and decoding, transmission). The requirements for real-time communications require strict minimum bandwidth guarantees. As a result, UDP is used instead of TCP[24].

2.1.6.1 The System Components of Video Conferencing

The core of a video conferencing system consists of elements that enable the capture and transfer of video images and audio sounds. These elements are:

- Video input : 2 or more video cameras or web cams; possibly digital projectors / whiteboards.
- Audio input : microphones either centrally located or on individuals.
- Videooutput : monitor, computer screen, television and/or projector.
- Audio output : professional speakers, headphones or laptop computer speakers.
- Codec : hardware or software-based coder-decoder technology that compresses analog video and audio data into digital packets and decompresses the data on the receiving end.
- Echo cancellation software : diminishes audio delays to enable real-time conversation.
- Network for data transfer :today most video conferencing is transmitted over a high-speed broadband Internet connection, using similar technology as VoIP (Voice over Internet Protocol)[24].

Table 2-4 define the constraint QOS parameters of video conference application [24].

Table 2. 4:QOS metrics for video conference application

Traffic Class	Technology Attributes	QoS Metrics						
		Timeliness			Preciseness			Accuracy
		Response time Expected by Users	Delay (ms)	Jitter (ms)	Data Rate (bps)	Required Bandwidth (bps)	Loss Rate	Error Rate
Video onferencing	Real Time and Symmetric	Lip-synch: <100 ms	<150	<400			<0.01%	< 0.01%
Coding Standard								
H.320					64–1920K	80K–2M		
H.323					64X K	80X K		
H.324					<64K	<80K		
Network Capacity								
Link: Refer to Appendix I					Router: Refer to Appendix II			

2.1.6.2 The Data Compression and Transfer

The camera and microphone capture analog video and audio signals from a video conference. These data are a continuous wave of amplitudes and frequencies representing sounds, color shades, depth and brightness. Enormous bandwidth would be required to transmit this data without compression, so codecs (hardware/software technology) compress and decompress the data into digital packets, once digitally compressed; the video and audio data can be transmitted over a digital network. In most cases, a broadband Internet connection is the preferred network. Data is sent to the other participant’s video conferencing system and then decompressed and translated back into analog video images and audio sounds[24].

2.1.7 Introduction to VoIP

Ever since its advent VoIP has opened new doors for telephony bringing forward immense possibilities. The basic reason for the popularity of VoIP is the cost which is very low as compared to the conventional telephony services. The concept of transmission of voice over data stream makes it possible to have VoIP transmitted and received using anything that uses IP - laptops, PC's, WiFi enabled handsets etc[25].

2.1.7.1 VoIP Processing

VoIP uses Internet Protocol for transmission of voice as packets over IP networks. The process involves digitization of voice, the isolation of unwanted noise signals and then the compression of the voice signal using compression algorithms/codecs.After the compression the voice is packetized to send over an IP network, each packet needs a destination address and sequence number and data for error checking. The signaling protocols are added at this stage to achieve these requirements along with the other call management requirements. When a voice packet arrives at the destination, the sequence number enables the packets to be place in order and then the decompression algorithms are applied to recover the data from the packets. Here the synchronization and delay management needs to be taken care of to make sure that there is proper spacing. Jitter buffer is used to store the packets arriving out of order through different routes, to wait for the packets arriving late[25].

2.1.7.2 Signaling Protocols

The signaling protocols H.323, SIP are used to setup the route for the transmission over the IP network, the Gateway protocols like the Media Gateway Control Protocol are used to establish control and status in the media and signaling gateways. Routing (UDP, TCP) and transport protocols (RTP) are used once the route is established for the transport of the data stream as shown in Figure2-6[25].

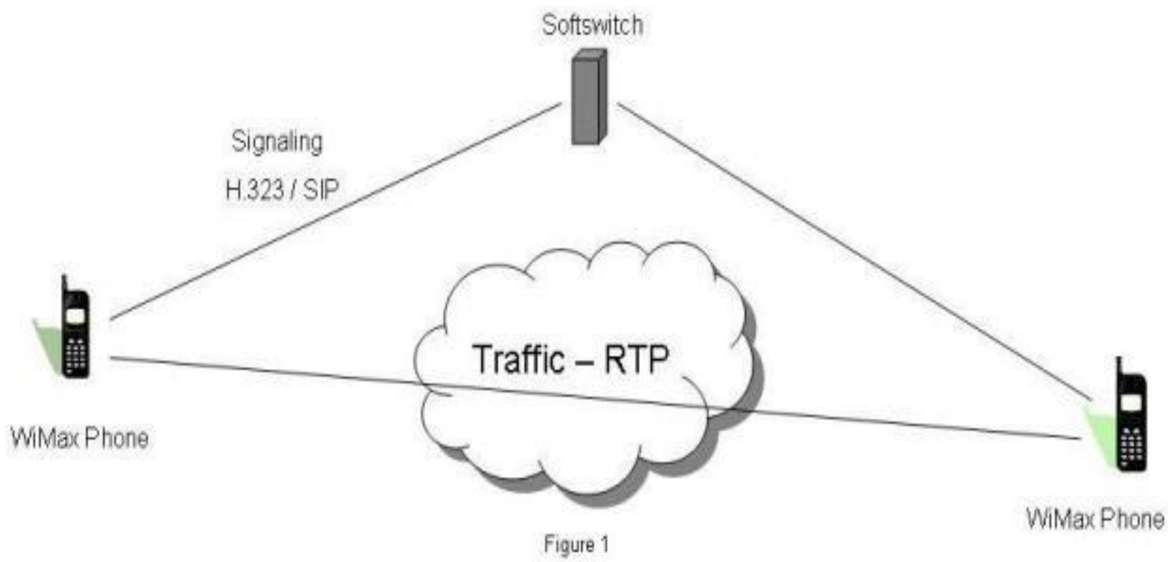

Figure 2. 6: Establishing route for transport data stream [25]

H.323 is the ITU-T standard for packet based multimedia communication, though originally developed for multimedia conferencing over LAN's it was later modified for VoIP as well. With versions coming out in 1996 and 1998, the standard has faced stiff competition from the other protocol SIP which was specifically designed for VoIP, but is more used because of its wide existence in the already installed networks. 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[25].

1-H.255.0 - Registration, Admission, Status, Call Signaling, Control

2-H.245 - Terminal Capability Exchange, Media Description, Control of Logical Channel[25].

Also H.323 offers specifications for call control, channel setup, codecs for the transmission of Real time video and voice over the networks where the QoS and guaranteed services are not available. For the transport RTP is used for real time audio and video streaming[25].

SIP is the IETF standard for VoIP signaling. 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, also the PSTN phone numbers are also compatible in a SIP network ensuring interfacing with PSTN systems. SIP also provides a mobility function to the users. SIP 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), Session Announcement Protocol (SAP). A lot of other protocols are also needed when it comes to the transport and signaling with the PSTN networks - RSVP, LDAP, RADIUS. SIP works on a client server architecture, where the clients are referred to as User Agents (UA). UA interact with the server mainly through a PC with a telephony agent or IP phone[25].Table 2-5 summarize and clarify QoS parameter constrains for VoIP application [26].

Table 2. 5: Respective Quality of Service (QoS) Requirements for VOIP application

.

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	Total delay for datatransfer to	Milliseconds	≤140 ms

Delay	occur.		
Jitter	Variation in packetarrival.	Milliseconds	≤0.5 ms

2.2 Related Works

The authors of [26] proposed that a framework for delivering real-time P2P streaming service over a WiMAX network. A detailed architecture of the ASN-GW whose operation is based on the combination of engineering and analytical techniques was illustrated. The work demonstrated the effectiveness of using probabilistic solution in dimensioning the resource requirement of a network.

The authors of [27], present the key issues involved in the multicasting of video stream over wimax is carry out. The mathematical solution is analyzed for selecting the optimal sub streams of scalable video streams under bandwidth constraints to maximize the quality for mobile receivers.

The authors of [28],proposed that The results obtained from simulation have shown that Weighted Fair Queue scheduler present efficient mechanism for service differentiation and hence presents better quality of service to the WiMAX than Priority Queue scheduler. While End-to-end delay of packets using WFQ scheduler is lower than the PQ scheduler. In case of various applications like voice, ftp and video conferencing the average traffic received using WFQ scheduler is more than by using PQ scheduler.

The authors of [29] , suggested that the general concepts of Quality of service (QoS) in wireless networks were studied. The IEEE 802.16/WiMAX network architecture was presented and the MAC layer advantages that enable end-to-end QoS mechanism in the network. VOIP traffic and video streaming traffic was analyzed using a simulation based on network simulator, ns-2. The effect of various service flows on QoS parameters like throughput, packet loss, average jitter and average delay was studied. In general, it was observed that the UGS service flow has the least overhead in terms of bandwidth request while rtps service flow has the highest overhead. It can be concluded from the results that the VOIP traffic can be best served with UGS service flow. The UGS service flow is indeed designed for constant bit rate traffic. Periodic bandwidth is already allocated by the BS to the MS.VOIP traffic generates fixed sized packets at a fixed interval. Thus the UGS service flow handles the traffic generated by VOIP calls in the most optimum way .ThertPS service flow is designed for applications such as streaming audio and streaming video. During the analysis of the video traffic, as the number of nodes increases, rtPS service flow comes out to be better than BE service flow for average jitter. UGS still has lower packet loss. However, UGS service flow does not utilize the network resources effectively when the traffic is not constant bit rate traffic. Streaming video traffic is variable bit rate traffic. The bandwidth can be periodically requested in the rtPS service flow instead of fixed bandwidth already being allocated, which may or may not get used.

The authors of [30], presented contributions is a decoding energy consumption model and its verification for the x264 coder. he formulated the problem of selecting the mode for each frame ,finding the coding bit rates for intra-coded and inter-coded frames, and burst scheduling in WiMAX wireless networks.

The authors of [31], created and evaluated an effective driving video VANET simulation platform with which other researchers can develop and test various VANET applications. The platform was built using the OPNET simulation tools. They have built a simple but practical traffic controller for driving video traffic over RTP applications, and provided a complete tier of communication layers for proper performance analysis.

The authors of [32], studied a framework for disseminating scalable video streams over mobile WiMAX networks. their problem statement is how to select the optimal sub streams of scalable video streams under bandwidth constraints ,they mathematically analyzed the problem .the solution of this problem is important because it enables the network operator to transmit higher quality video or more number of video streams at the same capacity.The authors of [33], suggested that the increasing of BS buffer size will reduced the packet loss and Results are understated since the model used worst case BE schedule.

The authors of [34], focused on this paper to presented the throughput characteristics of the AW and HWW Video surveillance systems. they had also established the true analytical theoretical maximum throughput of each CPE including the derivation of the CPE throughput equations. Results define the HWW outperforms the AW video surveillance system in terms of throughput by a factor of 1.75. It has also been shown that throughput increases with increase in number of nodes.

The authors of [35]focused into the internet is challenging task due to the stringent QoS required by video transmission over wireless networks and also affected by many channel impairments. By using a fast mode decision algorithm for H.264 intra prediction and an adaptive transmission control methods of video can gain good QoS and achieves 30% to 60% computation reduction on aspects of video coding, so that the stability and good qualities of video transmission can be ensured. Based on the above investigation the work can be continued in the following areas which includes efficient video coding, reliable wireless transmission, QoS, transmission rate, energy efficiency of handheld devices to improve the overall wireless video transmission system.

Researchers in [36]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.).

Chapter Three

Simulation Design

In this project, OPNET Modeler 14.5 is used to simulate real time applications. The topology has contained sex scenarios of network and a certain combination of users that are allocated a QoS and a selected application e.g.,voice or video conferencing, UGS offers higher performance than ertPS and rtps.

as categorized in the following we used OPNET14.5 modelers in order to evaluate the performance of QOS classes overWiMAX network. These sets are to study the average throughput, average delay and average packet delay variation of three

3.1 Simulation and scenario steps

The configuration of the WiMAX scenarios arecreated, there were sex scenarios are investigated, three refer to voip evaluation and three refer to to video evaluation as define below:

1. First scenario is the WiMAX simulated for video conference application using UGS service flow.
2. Second scenario is simulated using ertps service flow for the video application.
3. Third scenario is simulated using rtps service flow for the video application.
4. The results achieved on previous scenarios are discussed and compared between the three scenarios in terms of QoS parameters that shared video applications.
5. Fourth scenario is the WiMAX simulated for voip application using UGS service flow.
6. Fifth scenario is simulated using ertps service flow for voip application.
7. Sixth scenario is simulated using rtps service flow for voip application.
8. The results achieved on these scenarios are discussed and compared between the three scenarios in terms of QoS parameters that shared voip applications.

3.2 Description of the OPNET simulator

OPNET (Optimized Network Engineering Tool) is a network simulator that simulate the behavior of real networks by provides virtual network communication environment.

3.2.1 OPNET modeler Features

- It is excellent and helpful for analysis and evaluation the performance for different technologies. Also for new network models and architectures Pre-defined network models
- .design exists for user education and development purpose.

3.3 Network Topology

In this research,six wireless scenarios for analyzing QoS of videoconferencing and VOIP have been considered named corresponding to the QOS classes that mean different network situations at the same topology which uses fixed base stations configuration as show below in figure 3-1. Inthe simulation in order to have fairer comparison, we have to used each of UGS, ertps and rtPS service classes for video conferencing and VOIP in WiMAX. The BSs' connected to the IPcloud via ppp_adv.Table 3-1 illustrates the parameters that are used for WiMAX model.

Table 3. 1: Simulation Parameters

No. of Subscriber Stations (SS)	5 per cell
No. of Base Stations(BS)	Four
No. of Servers	One
Base Station Voltage	0.5 Watt
Mobility of Nodes	Mobile
Simulation Time	30 minutes

In this simulation setup, Video service traffic and VOIP are performed as the following:

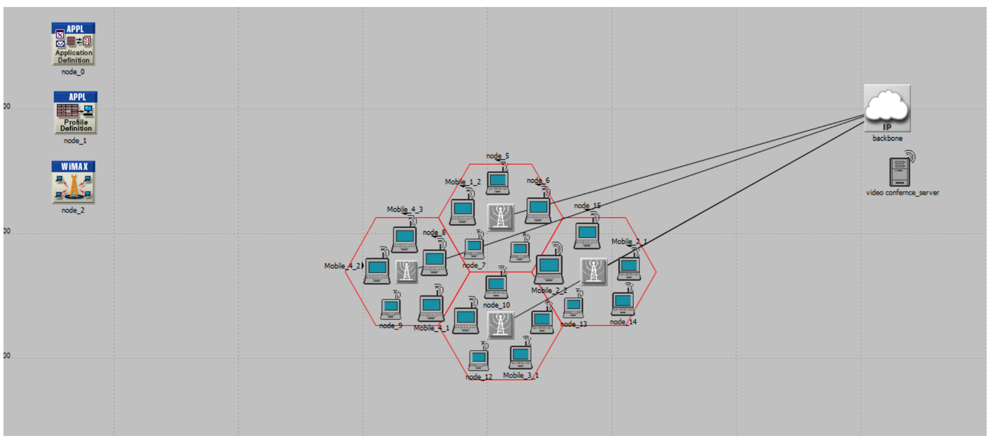

figure 3. 1: WiMAX Network Topology

3.4 Traffics Model

For all scenarios, each of Video traffic and voiptraffic models had to be created with background traffic which is FTP traffic, the Application, and Profile had to be configured.

3.4.1 Application configurations

It specifies standard and custom applications, used in simulation including traffic and QOS parameters. This standard application can be light or heavy. Selected application in this research are namedVideo, Voice andftp, they created in the Application definition node which consist of predefined applications was modified to support the traffics. The settings are shown in figure 3-2.

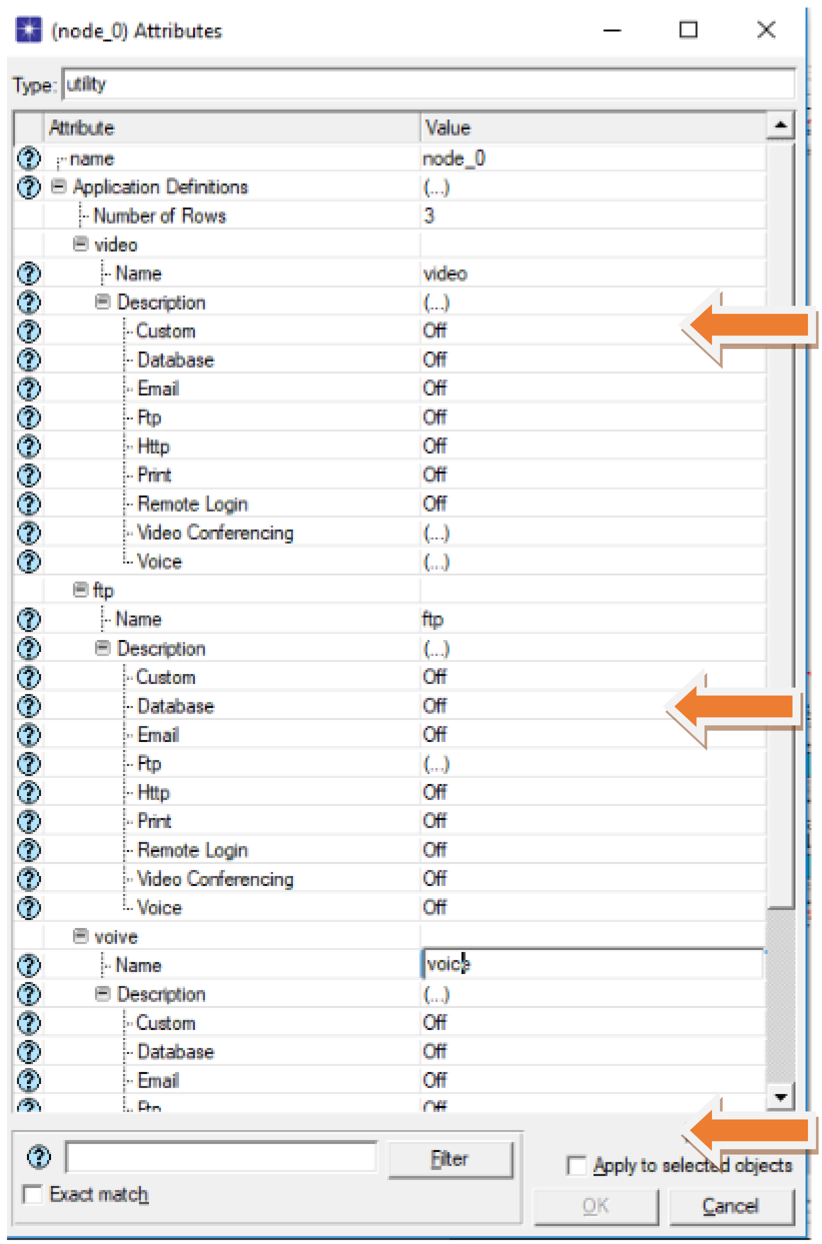

figure 3. 2: Application Attributes

3.4.2 Profile Configurations

Profiles called video_pro,voip_pro and ftp_pro are created in the Profile definition node. These profiles are used by the user nodes in order to generate traffic.

The profile was defined to use the Video conference call and voice application they were set to begin at the start of the simulation and continue until the simulation is completed other parameters were sitting as OPNET default values. Figure3-3 shows the profile configuration menu.

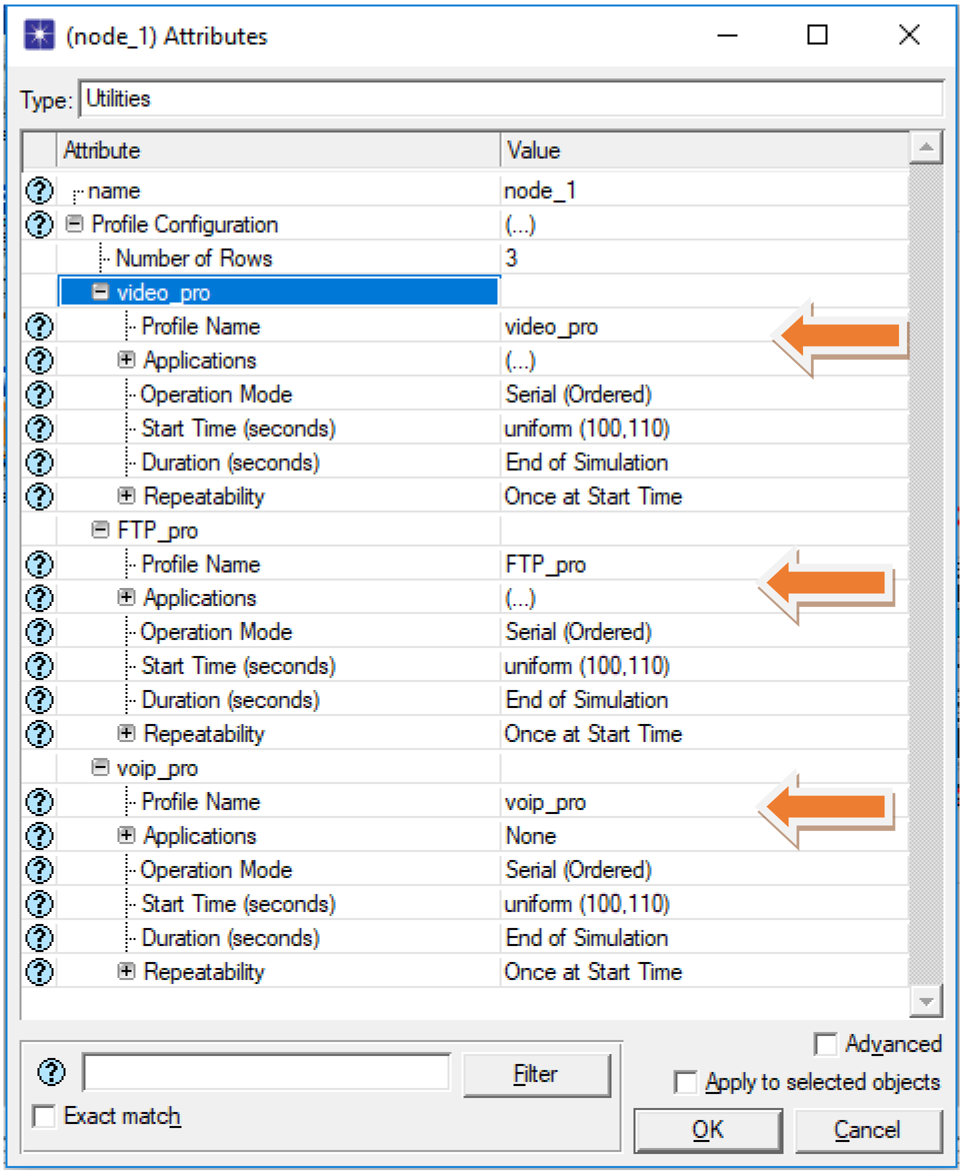

figure 3. 3: Profile Attributes

3.4.3WiMAX_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 and others parameters were sitting as default values. The configuration can be seen in figure 3-4 .

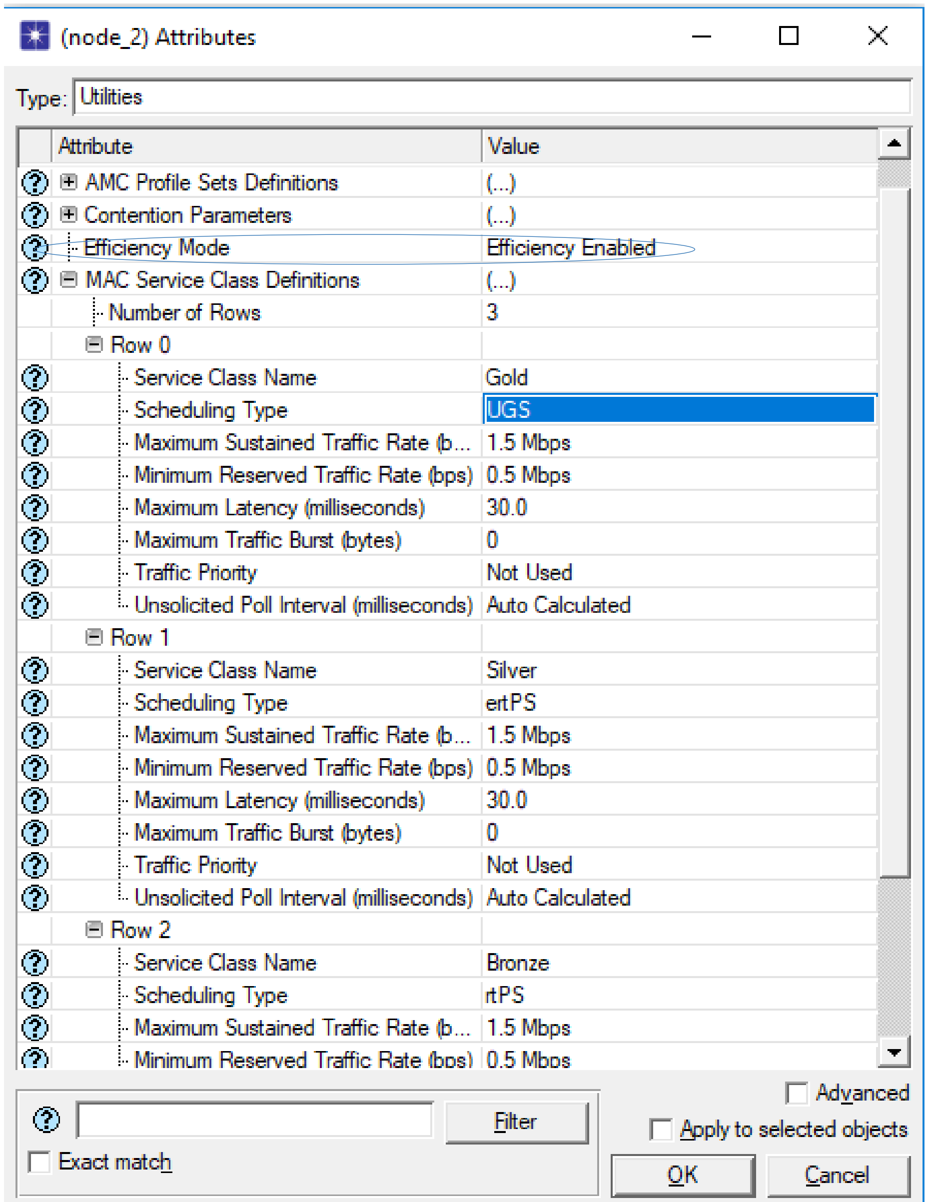

figure 3. 4: WiMAX Configuration:UGS,ertps and rtPS Scheduling Type

3.4.4 WIMAX Nodes Configurations

The topology is conducted six sets of simulations scenarios ,The aim of each one of these sets is to study the average throughput, average delay and average jitter of each UGS,ertps and rtPS service classes to both VOIP and Video system over WiMAX traffics. Configurations of base stations, and subscriber station(SS) are needed .

3.4.4.1 WiMAX Base Station configuration

The configuration of Base Station (BS) different in each group of scenarios (VOIP scenarios and video scenarios) as define below There are typical values that are commonly used in real life situations.

- **Base station (video conferencing_ scenarios)**

The Qos classes that implement at this scenario 20% of user upload and download video traffic and ftp traffic as background ,but others users sent only ftp trrafic, Service Class Names for Video are set to gold which is refer to UGS , silver which is refer to ertps and finally bronze which is refer to rtps with match value which is interactive multimedia as shown in figure 3-5.

- **Base station (VoIP_ scenarios)**

The Qos classes that implement at this scenario 60% of user sent and recives voip traffic and ftp traffic as background ,but others users sent only ftp trrafic,, Service Class Names for VOIP are set to gold which is refer to UGS , silver which is refer to ertps and finally bronze which is refer to rtps with match value which is interactive voice as shown in figure 3-6.

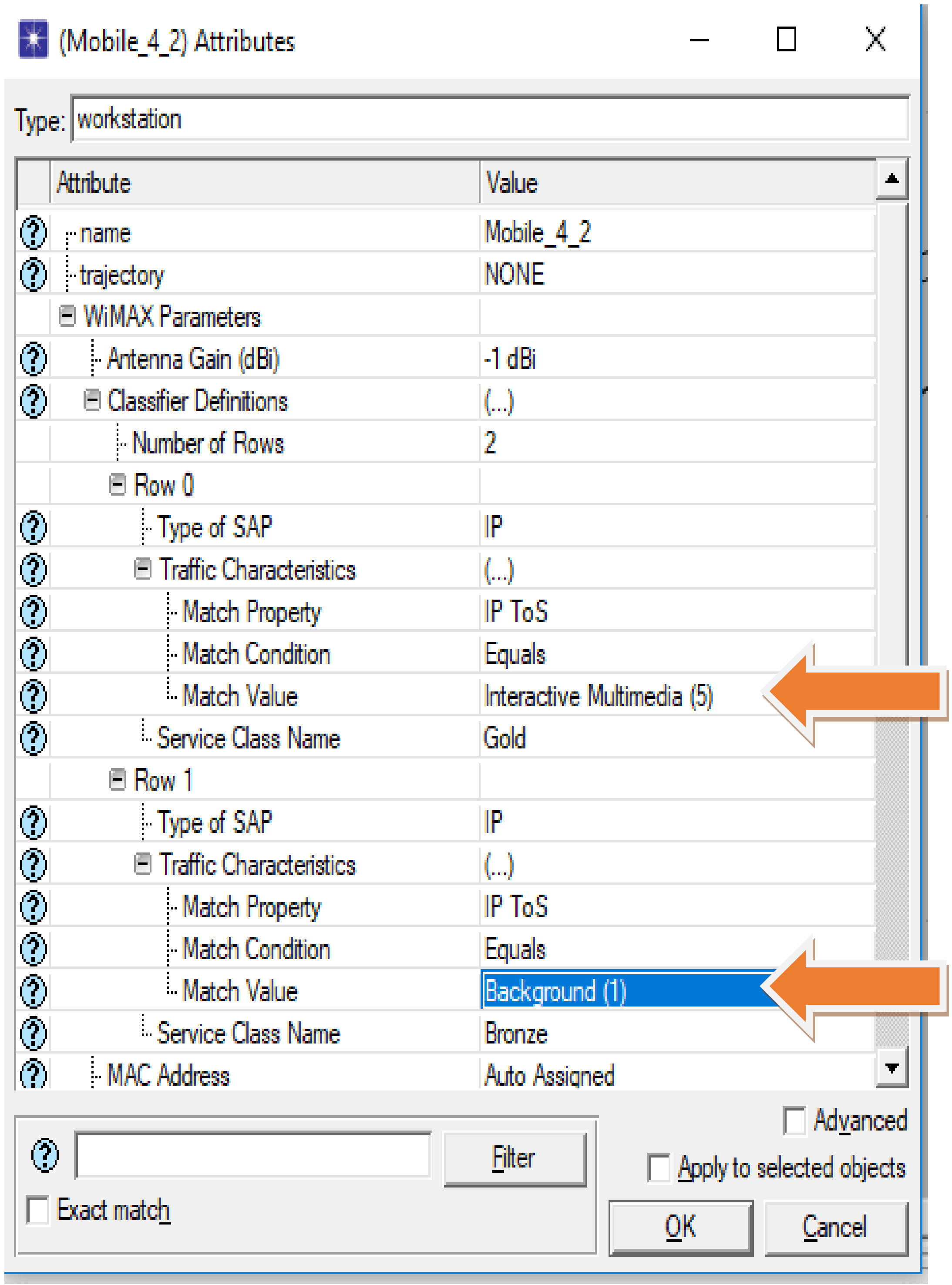

figure 3. 5: WiMAX base station Configuration(video conference)

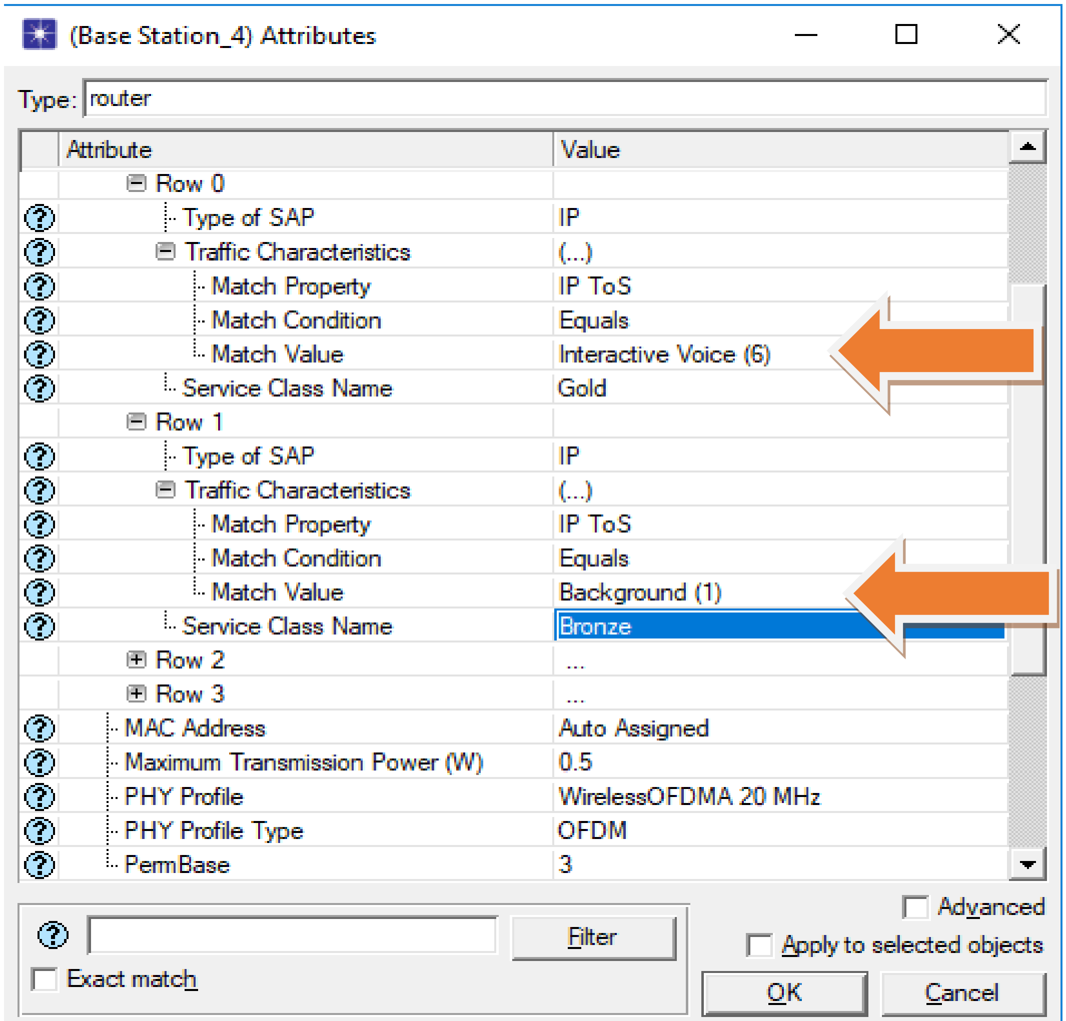

figure 3. 6: WiMAX base station Configuration(VOIP)

3.4.4.2 WIMAX Subscriber Station (SS)

- **Subscriber Station (video conferencing _ scenarios)**

WiMAX parameters, and applications. For SS parameters, 2 rows are added to represent main and background Traffic (i.e. video, FTP) as shown in figure 3-7 Interactive multimedia, Background respectively. Service Class Name for Video is set to Gold(gold refer to UGS) and so on.

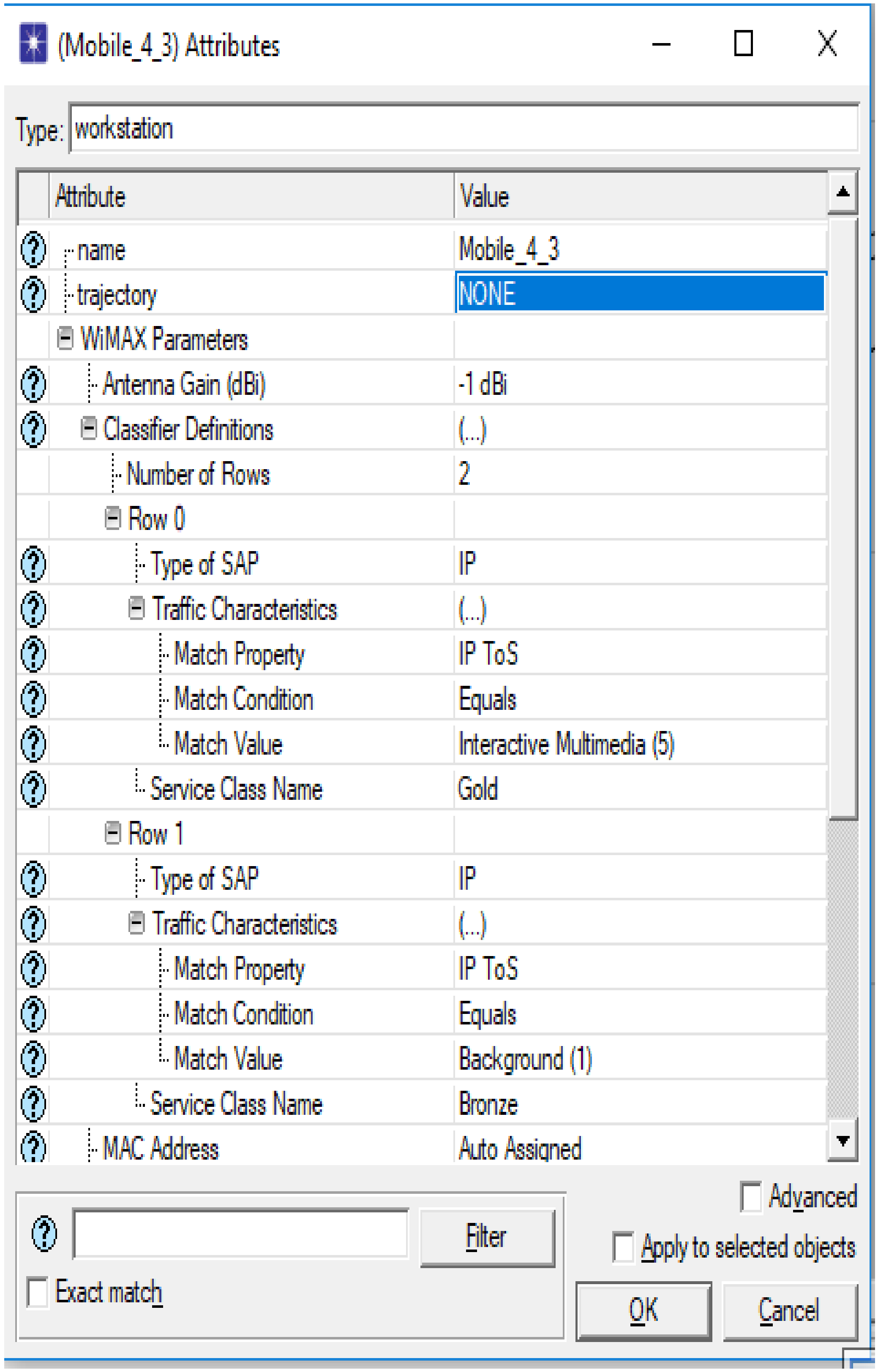

figure 3. 7: WiMAX Mobile Node: WiMAX Parameters Configuration

As show below figure 3-8define the uplink and down link flow that refer to request service class.

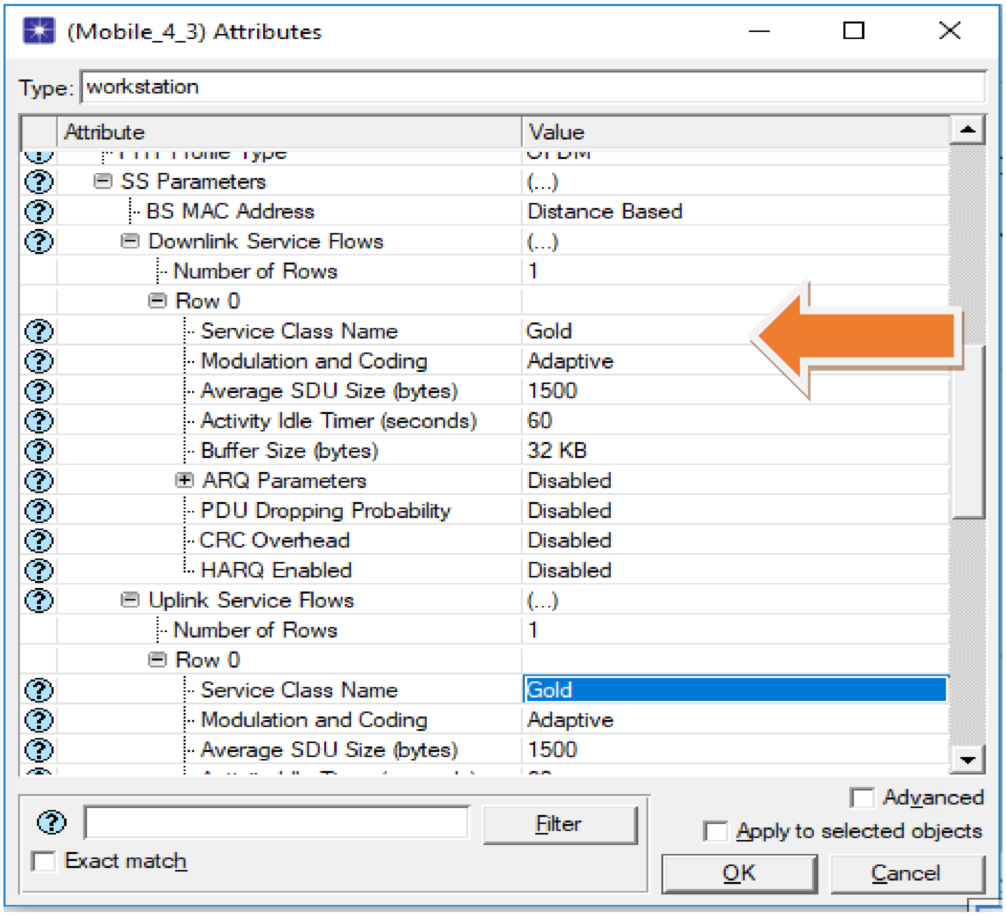

figure 3. 8: WiMAX Mobile Node: SS Parameters Configuration

the sitting of WiMAX Application Parameters, Video and ftp profiles are added to the supported profiles section also ftp application is added to supported service but as for the video service is supported by the video conference_server figures below 3-9 and 3-10.

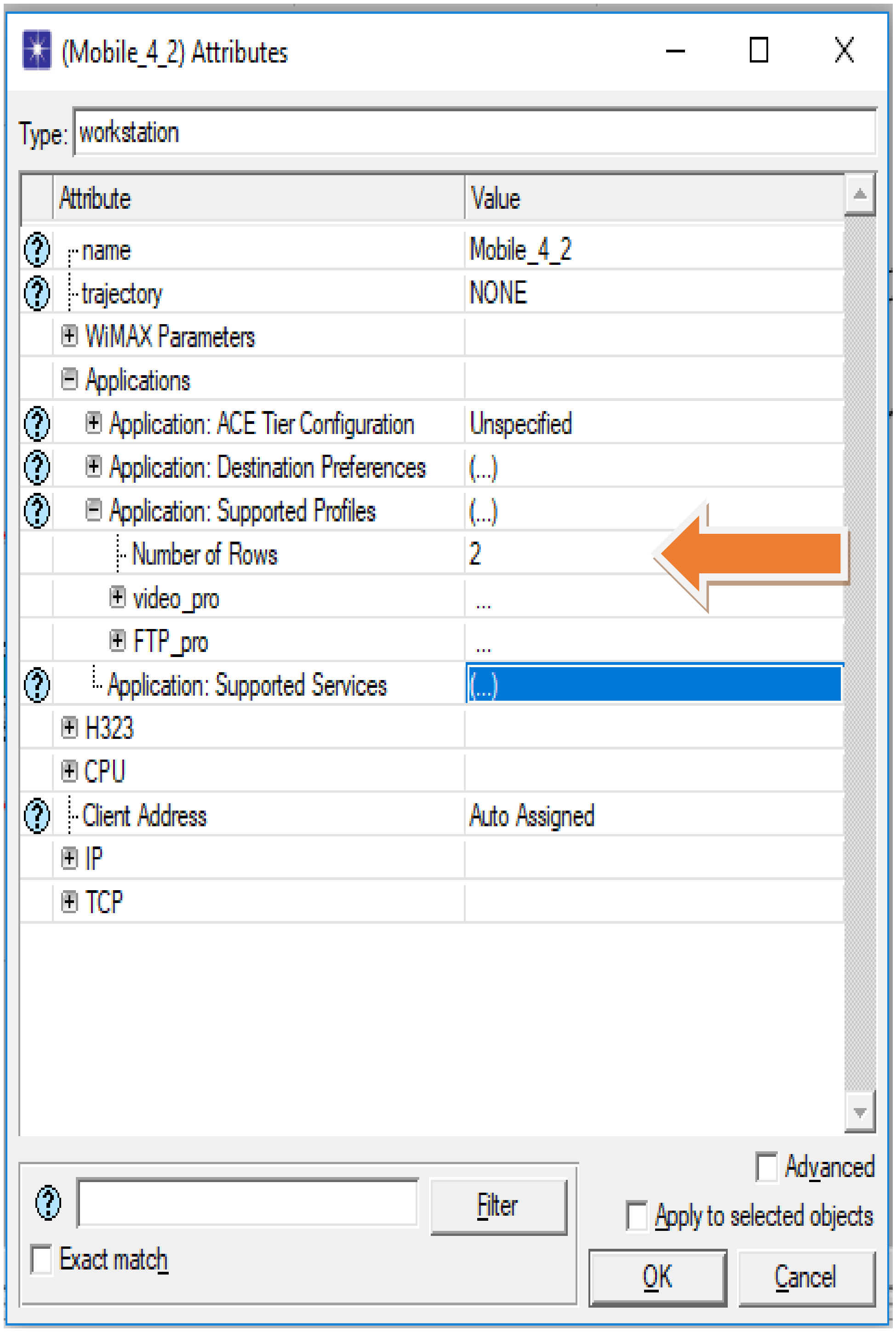

figure 3. 9: WiMAX Mobile Node: Application ParametersConfigurations

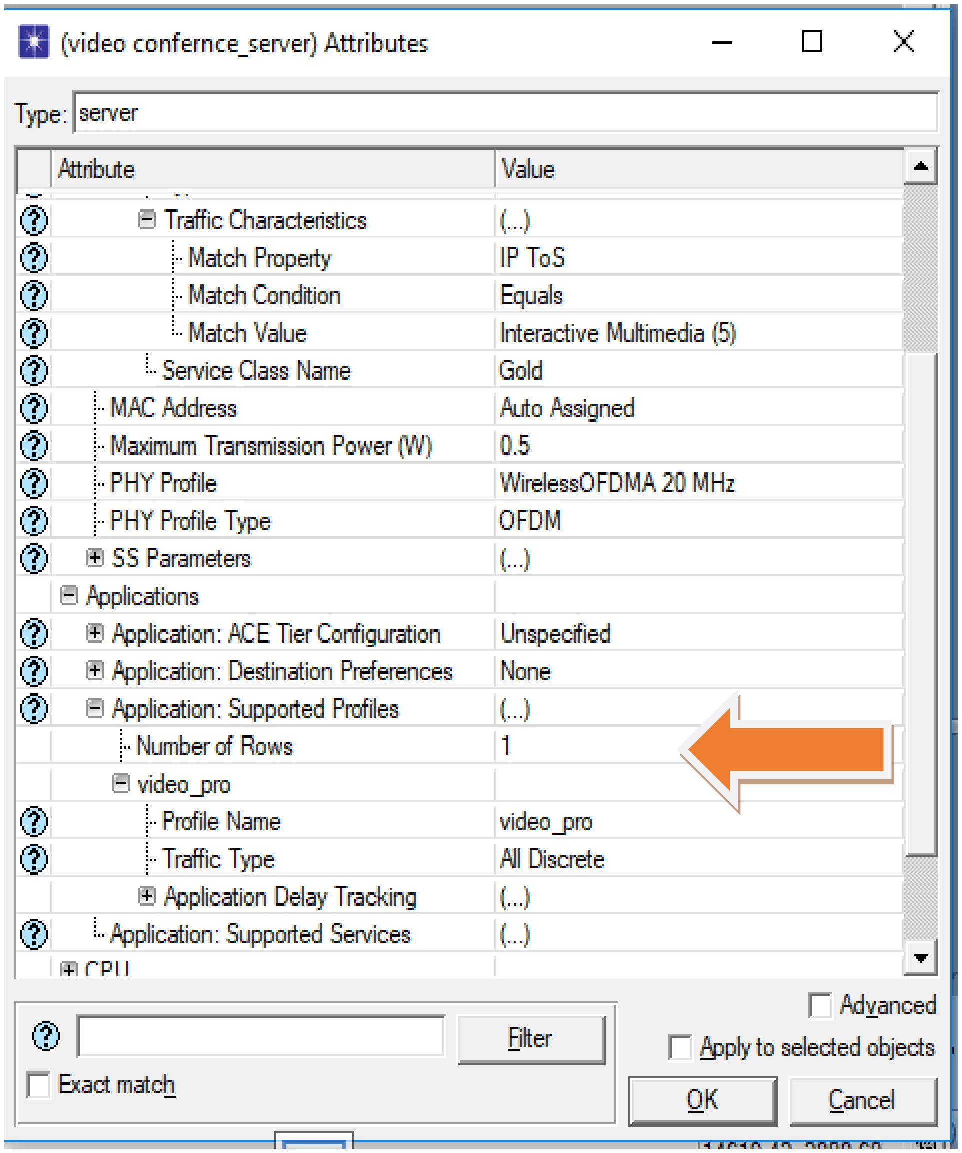

figure 3. 10: WiMAX server Node: Application ParametersConfigurations

- **Subscriber Station (VoIP_ scenarios)**

WiMAX parameters, and applications. For SS parameters, 2 rows are added to represent main and background Traffic (i.e. voice, FTP) as shown in figure 3-11 Interactive voice, Background respectively. Service Class Name for Voice is set to Gold (gold refer to UGS) and so on.

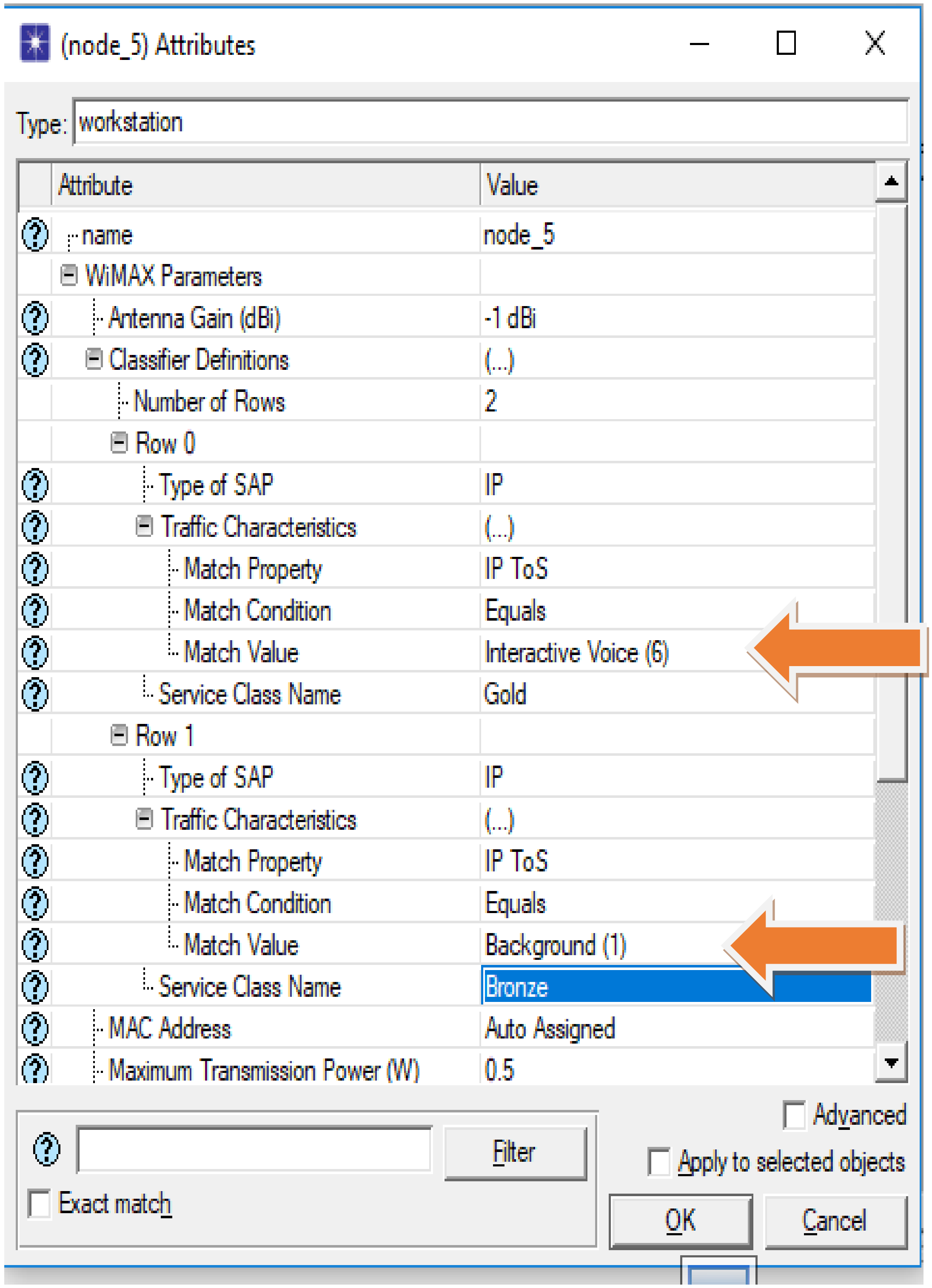

figure 3. 11: WiMAX Mobile Node: WiMAX Parameters Configuration

As show below figure 3-12 define the uplink and downlinkflow that refers to request service class.

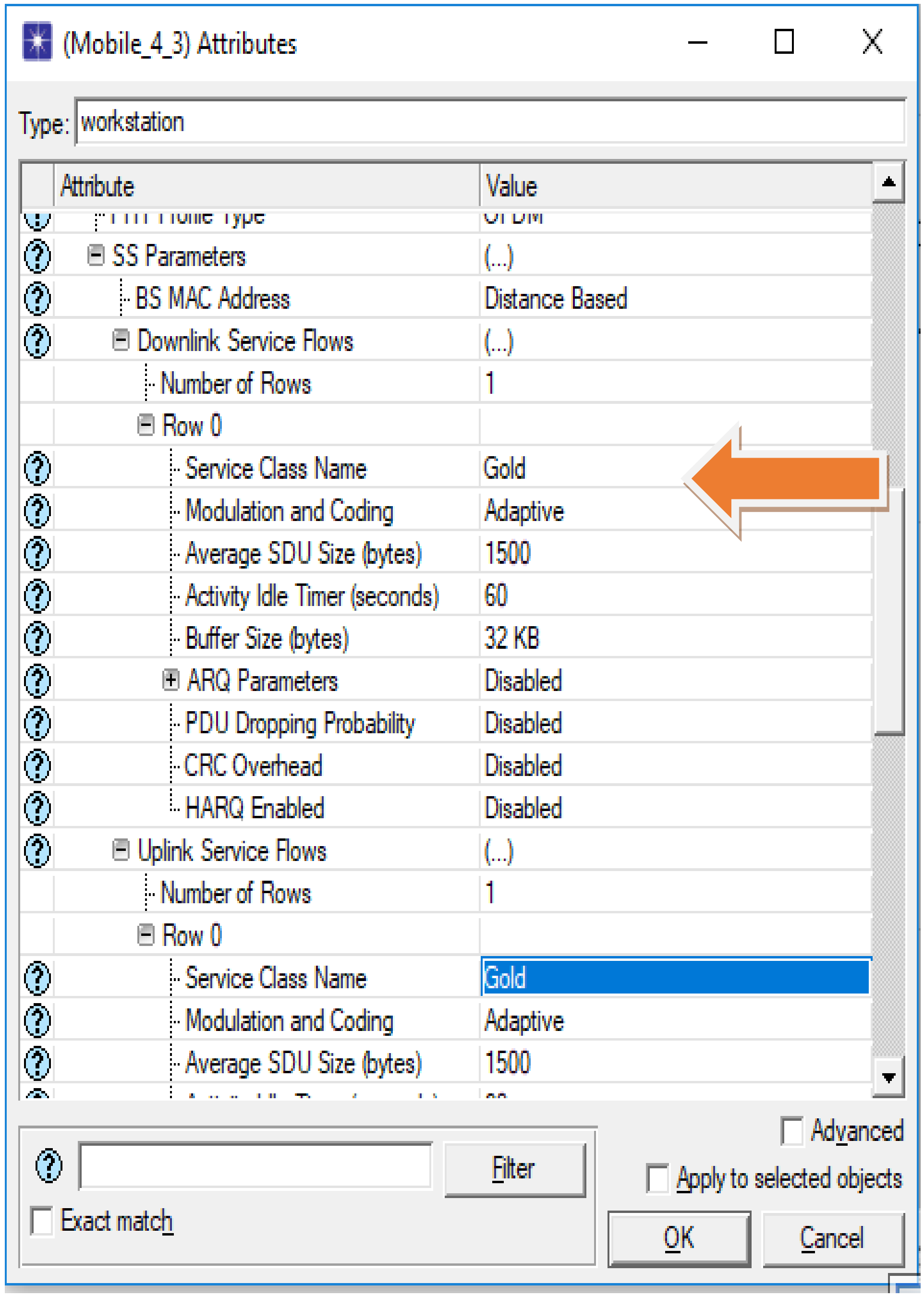

figure 3. 12: WiMAX Mobile Node: SS Parameters Configuration

the sitting of WiMAX Application Parameters, voip and ftp profiles are added to the supported profiles section and support all services as show figures below 3-13.

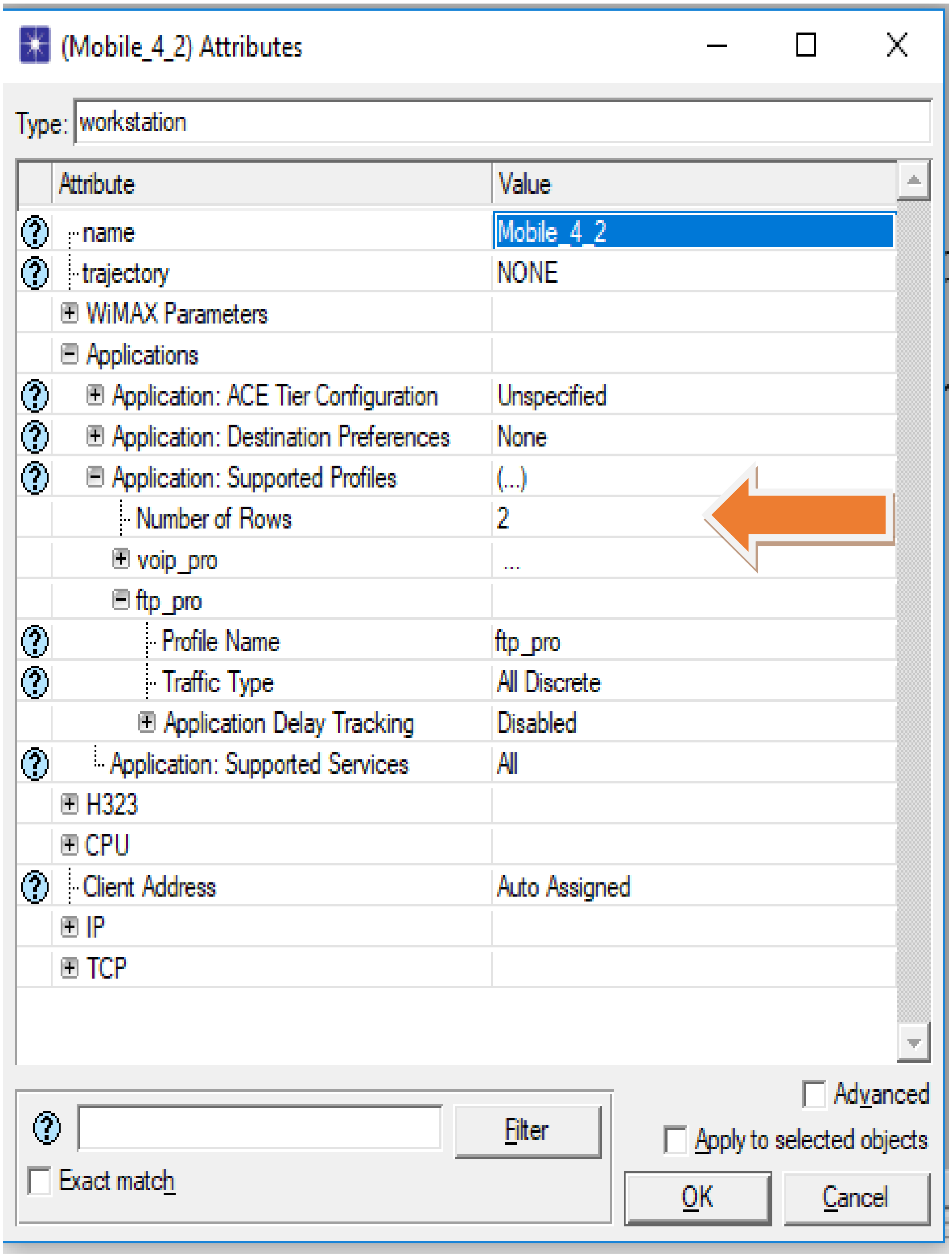

figure 3. 13: WiMAX Mobile Node: Application Parameters Configurations

- all unmarked parameters were sitting as default values. The duration of simulation was 30 minutes.

Chapter Four

Simulation Results and Analysis

The comparative analysis results of QoS classes in WiMAX networks are presented in this chapter. Six simulation scenarios were built. Each scenario used one type IEEE 802.16 QoSclasses (UGS,ertps and rtps), and initial QoSparamete. The following result compared all the aforementioned QOS classes to evaluate their performance, all scenarios were classified to two groups depend on the application that used as define below.

4.1 video conferencing Results

Three scenarios were simulated to study the effect of using QoS classes on video service over WiMAX networks. video results obtained from the simulation is shown below. As for the amount of ftp background traffic is shown in figure 4-1 and later Video results obtained from the simulation is shown below.

4.1.1 Received FTP as Background Traffic

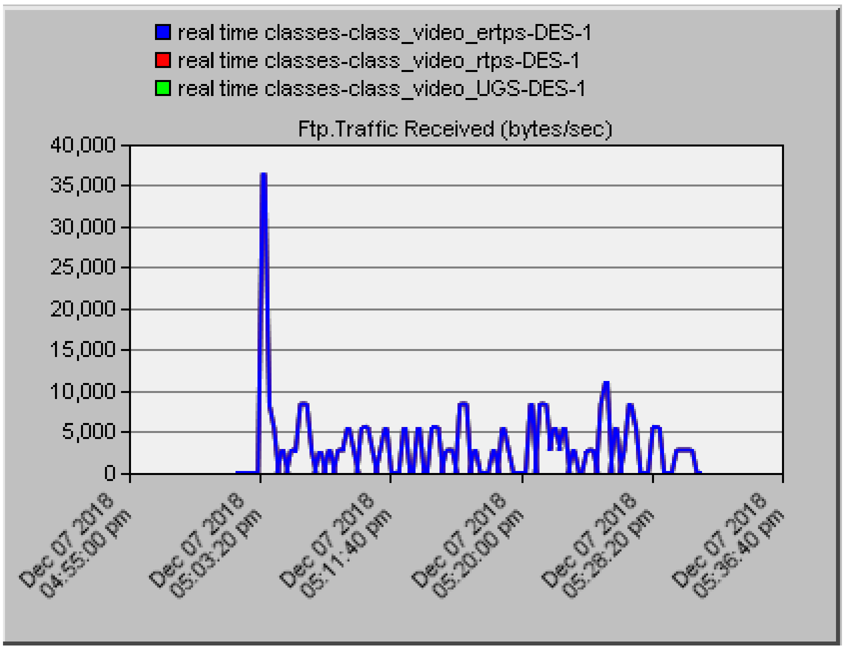

Figure 4. 1: FTP Background Traffic received (a)

4.1.2 Packet delay variation

Figure 4-2Compares video packet delay variation levels when UGS, ertps and rtPS are applied. While the number of nodes is 25, UGS class returned the lowest video packet delay variation of approximately**0.0019 second**but rtps provided the highest value of **0.022sec,** the packet delay variation decreased to **80%**when UGSused.

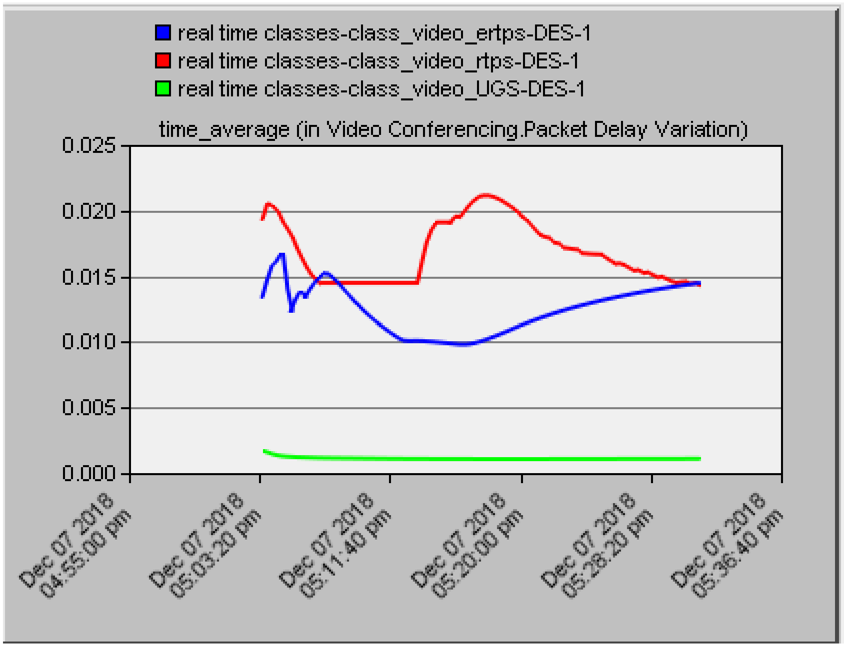

Figure 4. 2: Average Video Packet Delay Variation

4.1.3 End to End Delay

The Video traffic suffered packet end to end delay as clearly witnessed in Figure 4-3, rtps produced the higher packet end to end delay reach to **0.33** sec that exceeded acceptable value(<150ms) and UGS class yielding the lower value of **0.015** sec.the packet end to end delay decreased to **95.5%.**when usingUGS .

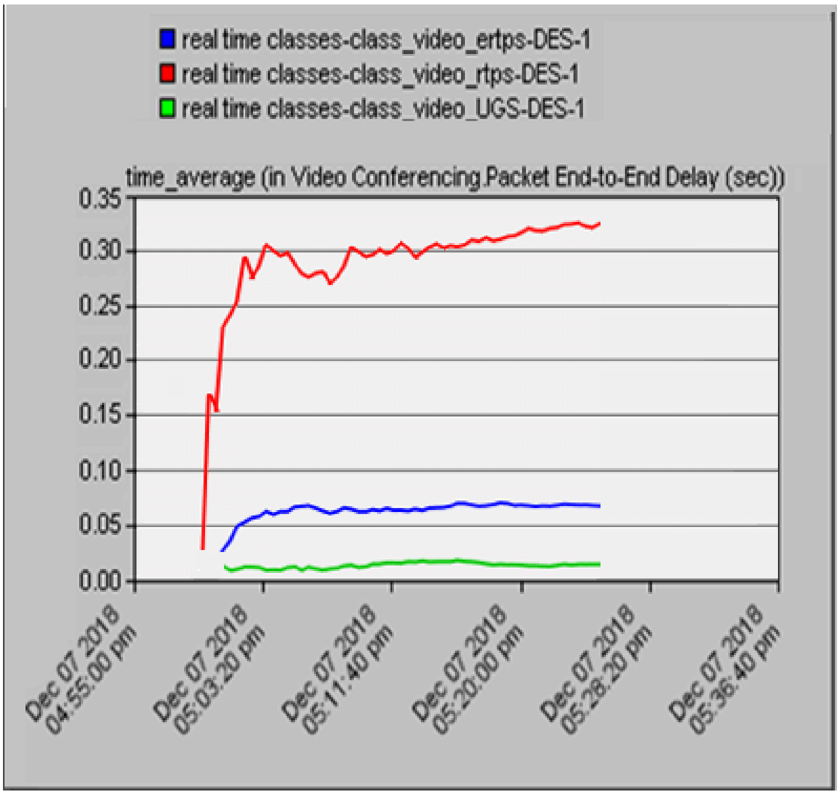

Figure 4. 3: Average Video End to End Delay

4.1.4 Throughput of WiMAX network

The overall throughput of UGS service flow was highest and rtps service flow was the lowest as shown in figure 4-4. For 25 nodes, the value of throughput for UGS service flow is **6999000 bits/sec** while **4000500bits/sec** for rtps class.the through put increased to **37.28%**when UGS used**.**

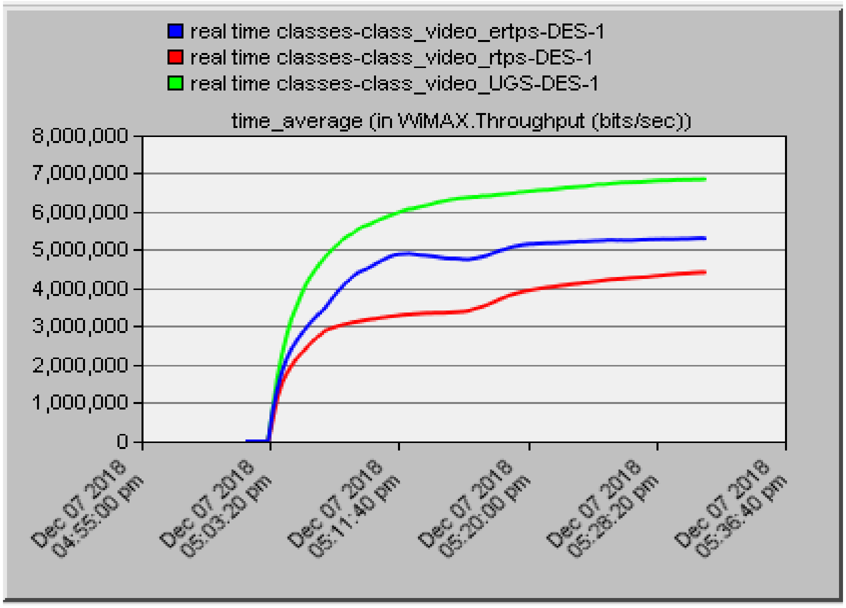

Figure 4. 4: Average Network Throughput

4.2 VOIP Results

VOIP scenario were simulated as previous results for video scenarios, they were simulated to study the effect of using QoS classes on VoIP service over WiMAX networks. As for the amount of ftp background traffic is shown in figure 4-5 and later VoIP results obtained from the simulation is shown below.

4.2.1 Received FTP as Background Traffic

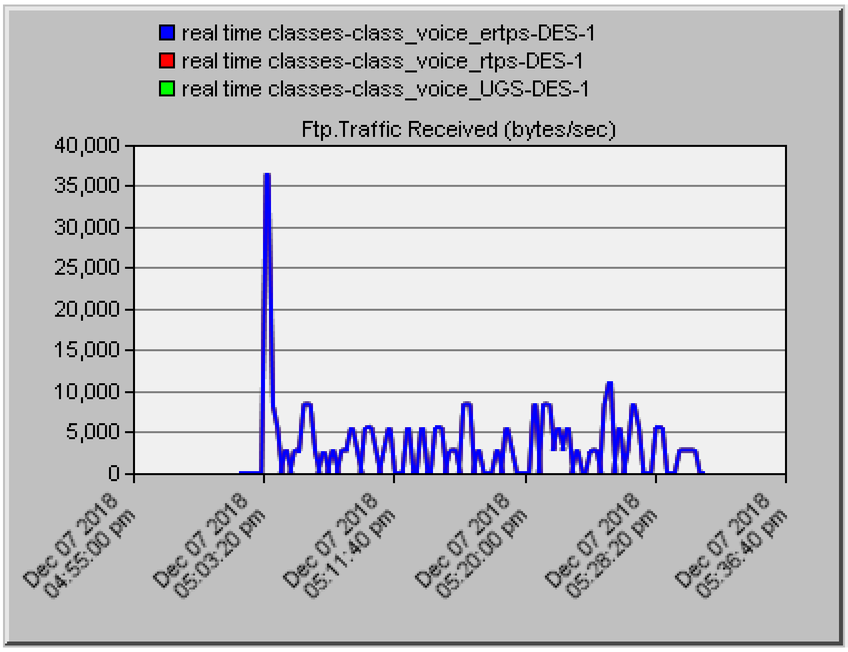

Figure 4. 5: FTP Background Traffic received (b)

4.2.2 Packet delay variation

Figure 4-6Compares voice packet delay variation levels when UGS, ertps and rtPS are applied. While the number of nodes is 25, UGS class returned the lowest voice packet delay variation of approximately**0.00001secand**rtps with the highest value of **0.00012 sec.**the packet delay variation decreased to **68.75%**when UGS used .

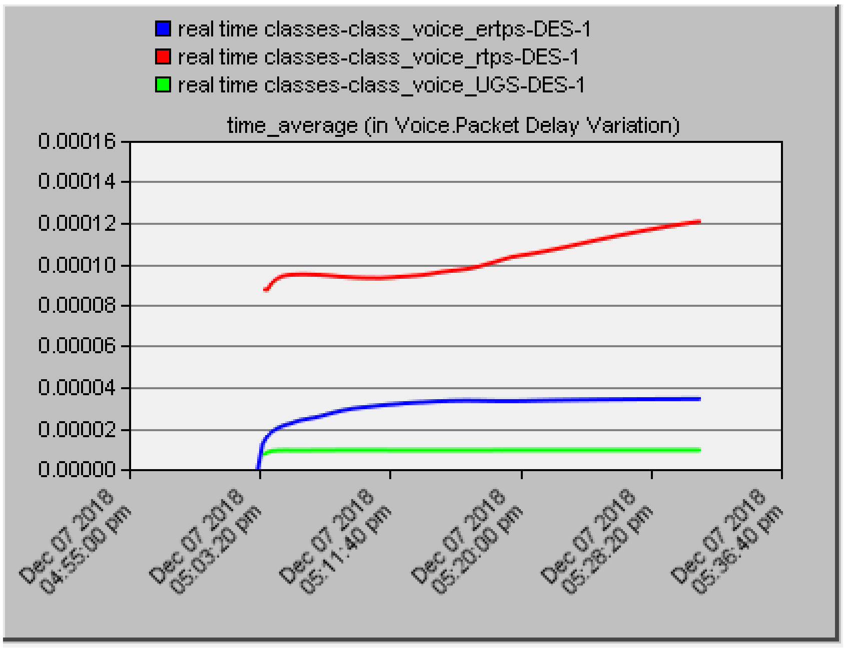

Figure 4. 6: Average Voice Packet Delay Variation

4.2.3 packet End to End Delay

The Voice traffic suffered packet end to end delay as clearly witnessed in Figure 4-7, rtps produced the higher packet end to end delay of **0.2**secand UGS class yielding the lower value of **0.07**sec.the packet end to end delay decreased to **52%**when UGS used .

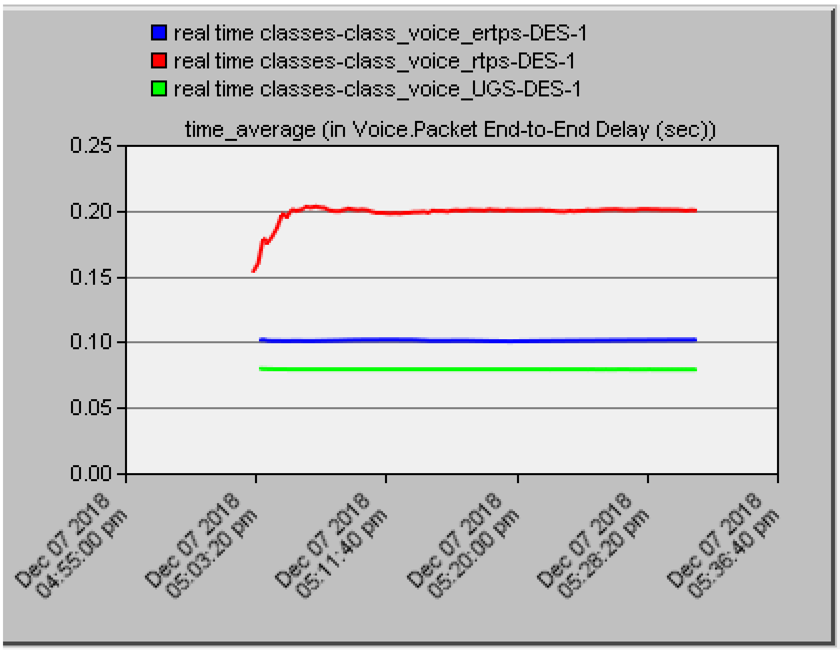

Figure 4. 7: Average Voice End to End Delay

4.2.4 Throughput of WiMAX network

The overall throughput of UGS service flow was highest and rtps service flow was the lowest as shown in figure 4-8. For 25 nodes, the value of throughput for UGS service flow is **1450120 bitss/sec** while **900500bits/sec**forrtps class.the Throughputincreased to **33%**when UGS used .

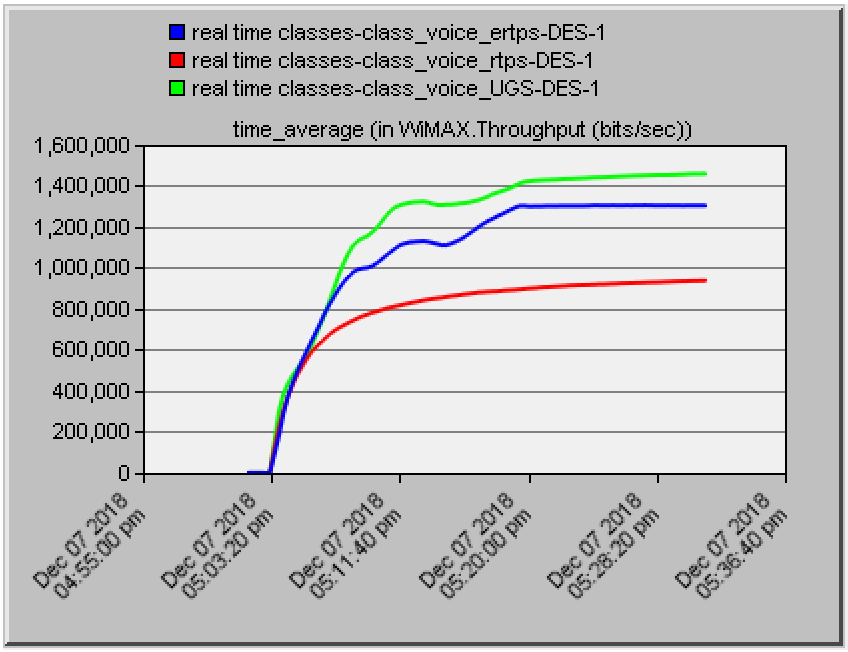

Figure 4. 8: Average Network Throughput

4.3 Summary of Results

For all above results justification is shared, the UGS has the best performance as shown below at figure 4-9, it has the best throughput, lowest average packet delay and lowest end to end delay due to it supports application with constant Bit Rate (CBR), It has given automatic grant by base station, unlike the rtps the delay is decreased due it supports the applications with Variable Bit Rate (VBR) It support variable grant sizes for data transport efficiency which made large overhead more than UGS. 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 and video. The bandwidth can be periodically requested in the ertPS and rtps service class instead of obtain it but default as UGW class.After applied all scenarios the results ofQoS parameters are in the acceptable range values refer to table 2-4 and table 2-5. The voice and video parameters values are within the range of acceptable packet delay variation threshold **(≤ 0.5 ms),** acceptable packet end to end delay threshold **(≤ 140 ms)**, and good throughput.

Table 4. 1: comparative result of performance of UGS class service

parameters	Improvement for Video results(%)	Improvement for VoIP results(%)
Packet delay variation	Decrease to 80%	Decrease to 68.75%
Packet end to end delay	Decrease to 66%	Decrease to 52%
throughput	Increase to 37.28%	Increase to 33%

Chapter Five

Conclusion and Recommendations

5.1 Conclusion

In the research, the general concepts of Quality of Service (QoS) in wireless networks were investigated through an extensive comparison of UGS,ertPS and rtps flows with the video conferencing and VoIP application. Simulation results of packet delay variation, packet end-to-end delay, and throughput refer to the UGS class service as the best QOS class. UGS flow shows minimum delay variation out of other service flow which is decreased to80% for video and 68.28% for VoIP unlike The value of rtps is the highest with compared to UGS service flow. UGS performs best with respect to average end-to-end delay performance parameter which decreased to 66% for video and 52% for VoIP. Also the throughput is the highest which is increased to37.28% for video and 33% for VoIP.

OPNET modeler 14.5 network simulation tools is used to investigate Several network performance parameters, they are used to determine the quality of VoIP and video calls that can be guaranteed with different QoS classes over WiMAX network. The purpose of this study to determine the best suitable class that will yield optimal quality for video and VoIP systems over WiMAX network.

5.2Recommendations

In this research the scheduling depends on the priority of QOS classes.In the future. It is good to study the performance of QOS over several scheduling algorithms in WiMAX like Round Robin(RR), Weighted Round Robin (WR), Weighted Fair Queuing (WFQ), Self-Clocked Fair (SCF) and Diff-serv Algorithm.

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