Sudan University of Science and Technology

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Performance Analysis of VoIP over Mobile WiMAX Networks

تحليل الأداء لتقنية نقل الصوت عبر بروتوكول الانترنت في شبكات واي ماكس المتنقلة

By:

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بسم الله الرحمن الرحيم

كلية الدراسات العليا

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الآية

بسم الله الرحمن الرحيم

(وَقُلِ اعْمَلُوا فَسَيَرَى اللَّهُ عَمَلَكُمْ وَرَسُولُهُ وَالْمُؤْمِنُونَ ۖ

وَسَتُرَدُّونَ إِلَىٰ عَالِمِ الْغَيْبِ وَالشَّهَادَةِ فَيُنَبِّئُكُمْ بِمَا كُنْتُمْ

تَعْمَلُونَ)

صدق الله العظيم

سورة التوبة الآية

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DEDICATION

Every challenging work needs self efforts as well as guidance of elders especially those who were very close to our heart.

My humble effort I dedicate to my sweet and loving

Mother,

Whose affection, love, encouragement and prays of day and night make me able to achieve such success and honor…

(Saeed Abdulmonem)

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All praise and thanks to Allah, Lord of the Worlds, and may the peace and blessings be on the most noble of Prophets and Messengers, our Prophet Muhammad, and on his family and all of his Companions. I seek refuge in Allah from the evils of our souls and the wickedness of our deeds. Whomsoever Allah guides, none can misguide, and whomsoever Allah misguides, none can guide. I thank Allah, the Exalted, for the completion of this thesis. Alhamdulillah, Allah gave me enough strength and patience to tackle every problem with calm and ease.

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ABSTRACT

Due to its large coverage area, low cost of deployment and high speed data rates, the Worldwide Interoperability for Microwave Access or WiMAX is a promising technology for providing wireless last-mile connectivity. Voice over Internet Protocol (VoIP) is one of the rapidly growing technologies and is expected to replace the conventional circuit switched voice services. VoIP is also considered one of the killer applications for WiMAX which requires careful design of QoS configurations.

The main aim of this research is to study and analyze the performance of VoIP on mobile WiMAX networks. This has been done through the study and analysis of several aspects related to the WiMAX network and VoIP configurations such as: WiMAX service classes, mobility, number of nodes and VoIP Codecs. The investigated parameters include: throughput, mean opinion score (MOS), jitter and delay. The results were obtained through the design and simulation of different WiMAX network scenarios using OPNET Modeler. Results showed that UGS service class has the best performance parameters serving VoIP. It is also observed that the G.723.1 codec has lower delay and higher MOS meanwhile sustaining minimal bandwidth consumption.

المستخلص

نظرا لسعة تغطيتها الواسعة، وانخفاض تكلفة تركيبها ونشرها ومعدلات نقل البيانات عالية السرعة التي توفرها، تعتبر تقنية واي ماكس تكنولوجيا واعدة لتوفير الاتصال اللاسلكي. تعتبر تقنية نقل الصوت عبر بروتوكول الإنترنت واحدة من التقنيات التي تنمو بسرعة، ويتوقع أن تحل محل التقنيات التقليدية لتقديم الخدمات الصوتية. كما تعتبر تقنية نقل الصوت عبر بروتوكول الإنترنت أيضا واحدة من التطبيقات القاتلة لشبكة واي ماكس الأمر الذي يتطلب تصميم دقيق لتكوينات جودة الخدمة.

الهدف الرئيسي من هذا البحث هو دراسة وتحليل أداء تقنية نقل الصوت عبر بروتوكول الإنترنت على شبكات واي ماكس المتنقلة. وقد تم ذلك من خلال دراسة وتحليل العديد من الجوانب المتصلة بشبكة واي ماكس وتقنية نقل الصوت عبر بروتوكول الإنترنت مثل: طبقات الخدمة، والتنقل، وعدد العقد وتقنيات ترميز وفك ترميز الصوت. تشمل عوامل الأداء التي تمت دراستها: معدل معالجة البيانات، ومعدل رضا الزبائن عن الخدمة، وزمن وصول البيانات و الاختلاف في زمن وصول البيانات. وقد تم الحصول على النتائج من خلال تصميم ومحاكاة سيناريوهات مختلفة لشبكة واي ماكس باستخدام برنامج النمذجة والمحاكاة (OPNET Modeler). وأظهرت النتائج أن طبقة الخدمة (UGS) لديها أفضل معايير الأداء لخدمة نقل الصوت عبر بروتوكول الإنترنت. كما تلاحظ أيضا أن الترميز (G.723.1) يمتاز بأدنى تأخير وأعلى قيمة لمعدل رضا الزبائن عن الخدمة وفي نفس الوقت يحافظ على الحد الأدنى من استهلاك السعات.

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List of Abbreviations

3GPP	3G Partnership Project
3GPP2	3G Partnership Project 2
AAA	Authentication, Authorization & Accounting
ACELP	Algebraic-Code Excited Linear Prediction
ACU	Admission Control Unit
ARQ	Automatic Repeat Request
ASN	Access Service Network
ASN-GW	Access Service Network Gateway
ASP	Application Service Provider
ATM	Asynchronous Transfer Mode
BE	Best Effort
BPSK	Binary Phase Shift Keying
BS	Base Station
CID	Connection Identifier
CPS	Common Part Sub-layer
CQI	Channel Quality Indicator
cRTP	Compressed Real-Time Protocol
CS	Convergence Sub-la
CS-ACELP	Conjugate-Structure ACELP
CSN	Connectivity Service Network
DE2E	End-to-End Delay
DHCP	Dynamic Host Control Protocol
DL	Downlink
ertPS	Enhanced Real-time Polling Service
ETSI	European Telecommunications Standards Institute
FBSS	Fast Base Station Switching
FDD	Frequency Division Duplex

FTP	File Transfer Protocol
HARQ	Hybrid Automatic Repeat Request
HHO	Hard Handover
HIPERMAN	High Performance Metropolitan Area Network
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IP	Internet Protocol
ITU	International Telecommunication Union
ITU-T	ITU - Technology Standardization Sector
LDPC	Low-Density Parity-Check
LOS	Line-of-Sight
MAC	Media Access Control
MAN	Metropolitan Area Network
MBS	Multicast Broadcast Service
MDHO	Macro Diversity Handover
MGCP	Media Gateway Control Protocol
MOS	Mean Opinion Score
MPDU	MAC Protocol Data Unit
MPEG	Moving Pictures Expert Group
MS	Mobile Station
MSDU	MAC Service Data Unit
NAP	Network Access Provider
NLOS	Non Line-of-Sight
nrtPS	Non-Real-time Polling Service
NSP	Network Service Provider
OFDM	Orthogonal Frequency Division Multiple
OSI	Open System Interconnection
PCM	Pulse Code Modulation
PDU	Protocol Data Unit
PHY	Physical Layer

PoA	Point of Attachment
PSTN	Public Switched Telephone Network
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
rtPS	Real-time Polling Service
SCCP	Signaling Connection Control Part
SFID	Service Flow Identifier
SIGTRAN	Signaling Transport
SIP	Session Initiation Protocol
SOFDMA	Scalable Orthogonal Frequency Division Multiple
SS	Subscriber Station
TCP	Transmission control protocol
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
ToS	Type of Service
UDP	User Datagram Protocol
UGS	Unsolicited Grant Service
UL	Uplink
UMTS	Universal Mobile Telephone System
VoD	Video on Demand
VoIP	Voice over Internet Protocol
WiMAX	Worldwide interoperability for Microwave Access
WLAN	Wireless Local Area Network
WMAN	Wireless Metropolitan Area Network

CHAPTER ONE

INTRODUCTION

CHAPTER 1

INTRODUCTION

1.1. Background

The excessive demand for providing mobile users with broadband wireless access has attracted tremendous investment from the telecommunications industry in the development and deployment of mobile WiMAX networks. Voice over Internet Protocol (VoIP) will be one of the killer applications for the rapid deployment of mobile WiMAX networks. The legal desire for bundling voice and data will increase the portion of voice traffic in these networks. Therefore, VoIP, as the current technology for making voice calls through packet switch networks, will be a key application in mobile WiMAX networks [6]. The increase of VoIP applications such as Skype, Google Talk, WhatsApp and so many others along with emerging deployment of mobile WiMAX networks is making VoIP over WiMAX an attractive market and a driving force for both carriers and equipment suppliers in capturing and spurring the next wave of telecommunications innovation, though challenges remain. However, the enhancement on the hardware and application sides only seems inadequate. The selection of proper network environment is also crucial in order to deliver the voice communication and multimedia session over the Internet. Optimization of the VoIP call capacity over WiMAX networks is one such crucial challenge and remains an open research issue [6]. Hence, in this thesis, we present a performance analysis of VoIP over mobile WiMAX networks.

1.2. Previous Studies

A rapid growth has been noticed in various wireless technologies in recent years. This has resulted in an increase in demand for wireless data services and multimedia application such as VoIP, streaming audio and video [13]. In order to provide good service and to meet the user demands, research has been in progress in both wireless technologies and VoIP systems. VoIP is becoming more and more popular especially after the deployment of mobile WiMAX network in many countries [14]. Different aspects of VoIP over WiMAX have been addressed by researchers.

The authors in [15] have investigated the data and voice support in the WiMAX network. The aim of their work was to examine the QoS deployment over WiMAX network and compare the performance obtained using two different WiMAX services classes i.e. UGS and ertPS. In [16], the authors have considered a fixed WiMAX network in order to evaluate the performance of VoIP. They have measured the performance of different transmission schemes in term of cumulative good-put, packet rate, sample loss rate and Mean Opinion Score (MOS). In [17], the authors have proposed a traffic- aware scheduling algorithm for VoIP applications in WiMAX networks. They have studied the performance of their proposed method and compared it with that of some conventional methods. They have discussed the trade-off between delay and bandwidth efficiency and it is shown that using their scheduling methods enhances the efficiency of VoIP over WiMAX.

The authors in [18] have discussed different issues related to VoIP and voice quality measurement models. They have outlined a new methodology for developing models for non-intrusive prediction of voice quality. The researchers in [19] have presented a voice quality measurement tool based on the ITU-T E-model [10]. They have tested the tool in some calls generated between two endpoints located at different Brazilian cities. In [20], the authors have conducted extensive simulation study to evaluate the performance of WiMAX and UMTS for supporting VoIP traffic. Their results show that WiMAX outscores the UMTS with a sufficient margin, and is the better technology to support VoIP applications, compared with UMTS.

1.3. Problem Statement

The availability of wireless Internet is rising with the continued deployment of WiMAX networks which features long range coverage. This widespread availability means there is potential for portable devices running VoIP that operate on WiMAX to become a popular alternative to cellular phones [6]. However, physical phenomena of the wireless transmission medium can have effect on how applications running over WiMAX networks perform. Network issues such as packet loss and delay affect the quality of the real-time applications, especially VoIP, and these issues are more prevalent in wireless networks. Ultimately, this discussion leads to the importance of studying the mobile WiMAX standard to analyze the performance of VoIP applications.

1.4. Objectives

The aim of this thesis is to study and analyze the performance of VoIP over mobile WiMAX networks. The main objectives of this research are:

- To study the mobile WiMAX (IEEE802.16e) standard with concentration on the MAC layer and QoS features.
- To study the mechanisms of QoS in WiMAX.
- To perform a brief review of VoIP technology and its main features.
- To identify the best methodology in order to design a model for the mobile WiMAX network in order to study QoS metrics and analyze VoIP performance.
- To simulate the WiMAX network model, using different scenarios addressing QoS service classes, mobility, number of users and voice codecs in order to investigate and analyze VoIP performance.

1.5. Methodology

For the thesis work both qualitative and quantitative approaches are used. First a detailed review of literature is done from studying the case studies and ethnographies and then we focus on the state of problem and how it can be solved. Computer simulation is most widely used as a research methodology for analyzing and studying the performance of wired and wireless networks. In our thesis work we had taken OPNET Modeler [12]. Using OPNET, several WiMAX network scenarios have been deployed to study and analyze VoIP performance. The different scenarios are aimed and designed to address several aspects of the WiMAX network that affect VoIP traffic, such as: available bandwidth, network capacity & load, QoS classes, data traffic and mobility. Several simulation runs have been performed for each scenario, with consideration to the following issues: length of runs, model "settle-down time", initial variables values and random numbers generation. Simulation outputs have been collected and statistically analyzed. After that, results were studied, interpreted and graphically presented. According to the obtained results recommendations were provided to address the problem statement of the thesis.

1.6. Thesis Layout

The thesis consists of five chapters, in addition to the list of references. Chapter Two provides a theoretical overview of mobile WiMAX networks and VoIP technology.

As for Chapter Three, it’s the core of the thesis and it describes the methodology, modeling, simulation parameters, implementation and results.

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

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

CHAPTER TWO

THEORETICAL OVERVIEW

CHAPTER 2

THEORETICAL OVERVIEW

2.1. WiMAX Overview

The Wireless Interoperability for Microwave Access or WiMAX is a promising alternative to 3G and wireless LAN for providing last-mile connectivity by radio links due to its large coverage area, low cost of deployment and high data rates. WiMAX is based on wireless metropolitan area networking (WMAN) standards developed by the IEEE 802.16 group and adopted by both IEEE and ETSI HIPERMAN group. A broad industry consortium, the WiMAX Forum [3], is responsible for certifying broadband wireless products for interoperability and compliance with the WiMAX standard. Since the thesis is considering the IEEE 802.16e (Mobile WiMAX) standard, this section presents a brief technical overview of the Mobile WiMAX standard.

2.1.1. Historical Background

In 1998, the IEEE formed a group to work on the new IEEE 802.16 standard. The group's initial focus was the development of an air-interface standard for LOS-based point-to-multipoint wireless broadband system for operation in the 10-66 GHz band. The standard was approved in December 2001 and was based on single-carrier modulation techniques with burst time division multiplexing (TDM) and supported both FDD and TDD [2]. To include NLOS applications and lower the cost of terminal devices, the group subsequently amended the standard to IEEE 802.16a using OFDM in the 2-11 GHz band. After further revisions, a new standard was produced in 2004, called IEEE 802.16d-2004, which replaced all preceding versions and formed the basis for the first WiMAX solution [2]. In December 2005, the group completed and approved the IEEE 802.16e-2005 standard, an amendment to the 2004 standard that added mobility support. The 2005 standard forms the basis for nomadic and mobile applications and is often referred to as ***Mobile WiMAX*** [1]. Figure 2.1 illustrates the evolution of WiMAX standards.

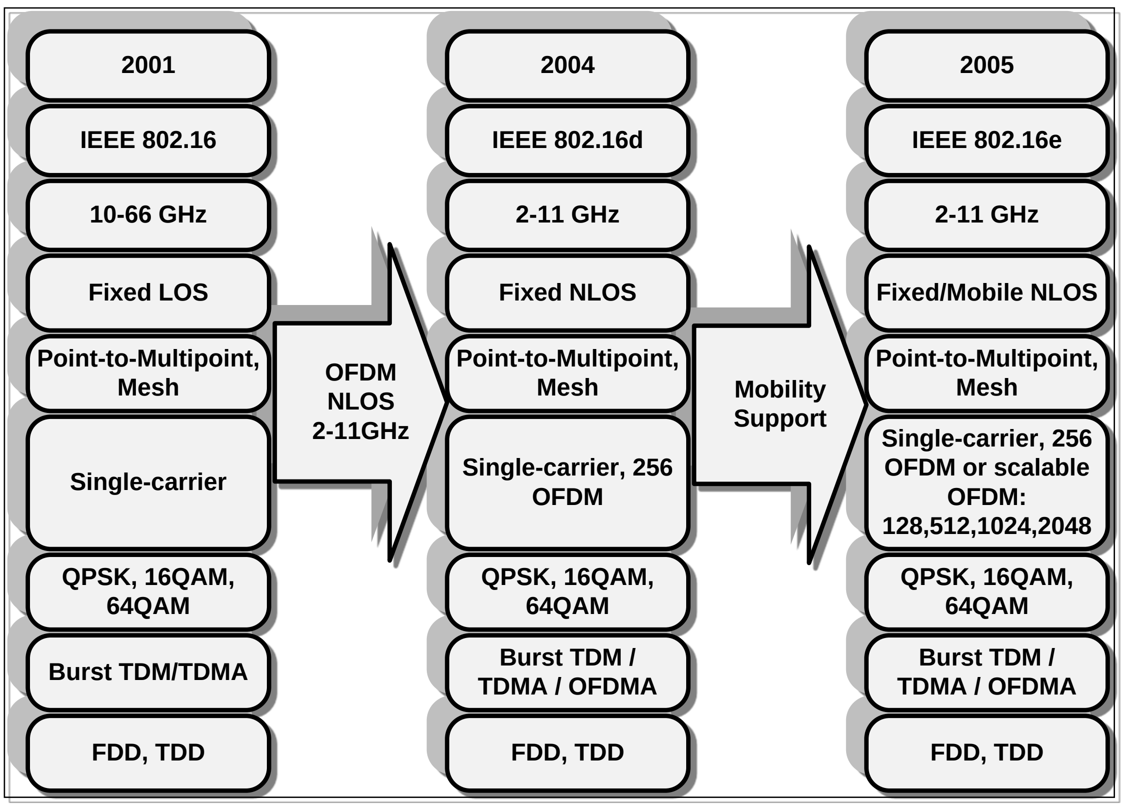

Figure 2.1: WiMAX Evolution

With the completion of the IEEE 802.16e-2005 standard, interest within WiMAX group and WiMAX Forum has shifted toward deploying and certifying mobile WiMAX system profiles based on the newer standard. It should be noted here that the 2004 and 2005 standards specifications are limited to the control and data plane concepts of the air-interface. The WiMAX Forum Network Working Group (NWG) [4] is developing a complete end-to-end network architecture to address service and network management aspects and fill in some of the missing pieces. In the next sections, concepts and entities related to the main aim of the thesis in the standard amendment IEEE 802.16e-2005 will be discussed in details.

2.1.2. WiMAX Network Architecture

The IEEE 802.16e standard provides the air-interface for mobile WiMAX, but does not define the end-to-end network. The WiMAX Forum's Network Working Group (NWG), is responsible for developing the end-to-end network requirements, architecture and protocols for mobile WiMAX using IEEE 802.16e as the air-interface. Furthermore, there are other aspects completely defined in the mobile WiMAX standard, yet there is no implementation of them in the WiMAX Forum consideration [4]. This is important to be taken into consideration, since at the present time only WiMAX Forum tested and tagged devices are being used by WiMAX operators around the world as those devices are approved to be interoperable.

The WiMAX NWG has developed a network reference model to serve as an architecture framework for WiMAX deployments and to ensure interoperability among various WiMAX equipments and operators. The network reference model envisions a unified network architecture for supporting fixed, nomadic, and mobile deployments and is based on an IP service model [4]. The network reference model developed by the WiMAX Forum NWG defines a number of functional entities and interfaces between those entities (The interfaces are referred to as reference points).

- ***Mobile Station (MS):*** The MS is used by the end user to access the network.
- ***Access Service Network (ASN):*** The ASN comprises one or more BSs and one or more ASN gateways that form the radio access network.
- ***Base Station (BS):*** The BS is responsible for providing the air interface to the MS. Additional functions that may be part of the BS are micro-mobility management functions, such as handoff triggering and tunnel establishment, radio resource management, QoS policy enforcement, traffic classification, DHCP (Dynamic Host Control Protocol) proxy, key management, session management, and multicast group management.
- ***Access service network gateway (ASN-GW):*** The ASN gateway typically acts as a layer 2 traffic aggregation point within an ASN. Additional functions that may be part of the ASN gateway include intra-ASN location management and paging, radio resource management and admission control, caching of subscriber profiles and encryption keys, AAA client functionality, establishment and management of mobility tunnel with base stations, QoS and policy enforcement, foreign agent functionality for mobile IP, and routing to the selected CSN.
- ***Connectivity service network (CSN):*** The CSN provides connectivity to the Internet, ASP, other public networks, and corporate networks. The CSN includes AAA servers that support authentication for the devices, users, and specific services. The CSN also provides per user policy

management of QoS and security. The CSN is also responsible for IP address management, support for roaming between different NSPs, location management between ASNs, and mobility and roaming between ASNs. Further, CSN can also provide gateways and inter-working with other networks, such as PSTN (public switched telephone network), 3GPP, and 3GPP2.

In addition to functional entities, the reference architecture defines interfaces, called *reference points*, between functional entities [4]. The interfaces carry control and management protocols— mostly IETF-developed network and transport-layer protocols—in support of several functions, such as mobility, security, and QoS, in addition to bearer data. Figure 2.2 illustrates reference points between functional entities (R1, R2, R3, etc…). Figure 2.2 shows some of the more important functional entities.

Furthermore, the architecture framework is designed such that the multi-players can be part of the WiMAX service value chain to enable richer ecosystem for WiMAX service business, leading to more competition and hence better services [4]. The architecture allows for three separate business entities:

- ***Network Access Provider (NAP):*** Owns and operates the ASN.
- ***Network Service Provider (NSP):*** Provides IP connectivity and WiMAX services to subscribers using the ASN infrastructure provided by one or more NAPs.
- ***Application Service Provider (ASP):*** Provides value-added services, such as multimedia applications and corporate VPN that run on top of IP.

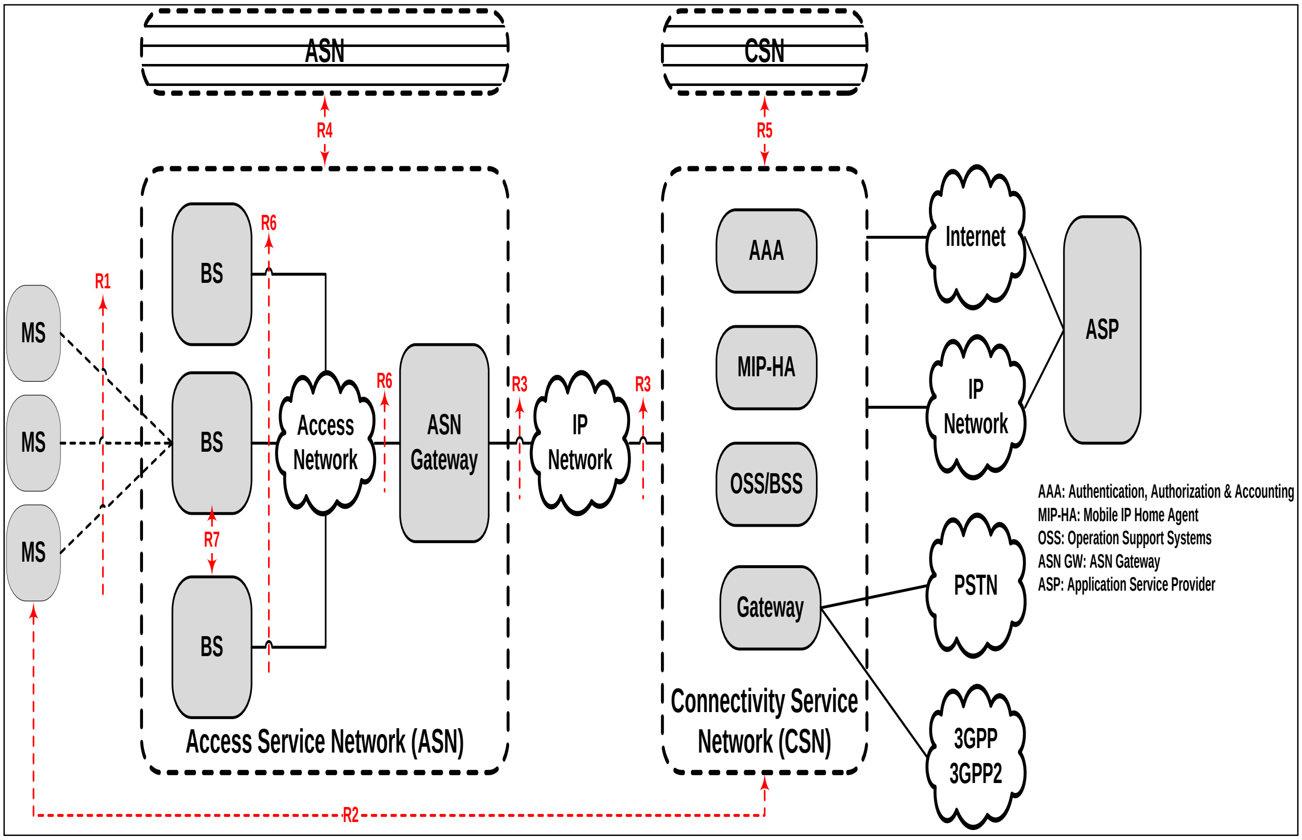

Figure 2.2: WiMAX Network Architecture

2.1.3. WiMAX PHY-Layer

The WiMAX physical layer is based on orthogonal frequency division multiplexing (OFDM). OFDM is the transmission scheme of choice to enable high-speed data, video, and multimedia communications and is used by a variety of commercial broadband systems [2]. OFDMA provides improved multi-path performance and operation in non-line-of-sight environments. Scalable OFDMA (SOFDMA) is introduced in the IEEE 802.16e amendment to support scalable channel bandwidths. Through the following parts of this section a brief overview of WiMAX physical layer is provided.

2.1.3.1. OFDM Basics

OFDM belongs to a family of transmission schemes called multicarrier modulation, which is based on the idea of dividing a given high-bit-rate data stream into several parallel lower bit-rate streams and modulating each stream on separate carriers, often called subcarriers or tones [1]. Multicarrier modulation schemes eliminate or minimize inter-symbol interference (ISI) by making the symbol time large enough so that the channel-induced delays are an insignificant (typically, < 10 percent) fraction of the symbol duration. Therefore, in high-data-rate systems in which the symbol duration is small, being inversely proportional to the data rate splitting the data stream into many parallel streams increases the symbol duration of each stream such that the delay spread is only a small fraction of the symbol duration.

OFDM is a spectrally efficient version of multicarrier modulation, where the subcarriers are selected such that they are all orthogonal to one another over the symbol duration, thereby avoiding the need to have non-overlapping subcarrier channels to eliminate inter-carrier interference [1]. In order to completely eliminate ISI, guard intervals are used between OFDM symbols. By making the guard interval larger than the expected multipath delay spread, ISI can be completely eliminated. Adding a guard interval, however, implies power wastage and a decrease in bandwidth efficiency [2].

2.1.3.1. Adaptive Modulation & Coding

WiMAX supports a variety of adaptive modulation and coding (AMC) schemes and allows for the scheme to change on a burst-by-burst basis per link, depending on channel conditions [1]. Using the channel quality feedback indicator, the mobile station can provide the base station with feedback on the downlink channel quality. For the uplink, the base station can estimate the channel quality, based on the received signal quality. Table 2.1 provides a list of the various modulation and coding schemes supported by WiMAX.

Table 2.1: WiMAX AMC Schemes

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

2.1.3.1. WiMAX Data Rates

Because the physical layer of WiMAX is quite flexible, data rate performance varies based on the operating parameters. Parameters that have a significant impact on the physical-layer data rate are channel bandwidth and the modulation and coding scheme used. Other parameters, such as number of sub-channels, OFDM guard time, and oversampling rate, also have an impact [1]. Following is the PHY-layer data rate at various channel bandwidths, as well as modulation and coding schemes shown in table 2.2.

Table 2.2: WiMAX Data Rates

Channel Bandwidth	3.5MHz		1.25MHz		5MHz		10MHz		8.75MHz	
PHY Mode	256 OFDM		128 OFDMA		512 OFDMA		1,024 OFDMA		1,024 OFDMA	
Oversampling	8/7		28/25		28/25		28/25		28/25	
Modulation and Code Rate	**PHY-Layer Data Rate (kbps)**									
	DL	UL	DL	UL	DL	UL	DL	UL	DL	UL
BPSK, 1/2	946	326	**Not Applicable**							
QPSK, 1/2	1,882	653	504	154	2,520	653	5,040	1,344	4,464	1,120
QPSK, 3/4	2,822	979	756	230	3,780	979	7,560	2,016	6,696	1,680
16 QAM, 1/2	3,763	1,306	1,008	307	5,040	1,306	10,080	2,688	8,928	2,240
16 QAM, 3/4	5,645	1,958	1,512	461	7,560	1,958	15,120	4,032	13,392	3,360
64 QAM, 1/2	5,645	1,958	1,512	461	7,560	1,958	15,120	4,032	13,392	3,360
64 QAM, 2/3	7,526	2,611	2,016	614	10,080	2,611	20,160	5,376	17,856	4,480
64 QAM, 3/4	8,467	2,938	2,268	691	11,340	2,938	22,680	6,048	20,088	5,040
64 QAM, 5/6	9,408	3,264	2,520	768	12,600	3,264	25,200	6,720	22,320	5,600

2.1.4. WiMAX MAC Layer

WiMAX MAC protocol is a centralized MAC protocol which uses hybrid access methods that combines the top properties of random access and guaranteed access protocols [2]. It uses demand assignment to facilitate and support efficient QoS mechanisms. Fundamentally the WiMAX MAC layer, just like any other MAC layer is situated between the Network Layer and Physical Layer of the OSI stack. However, the WiMAX MAC is subdivided into three sub-layers with different functionalities. Figure 2.3 is a basic illustration of the tasks and services that the MAC sub-layers are responsible for.

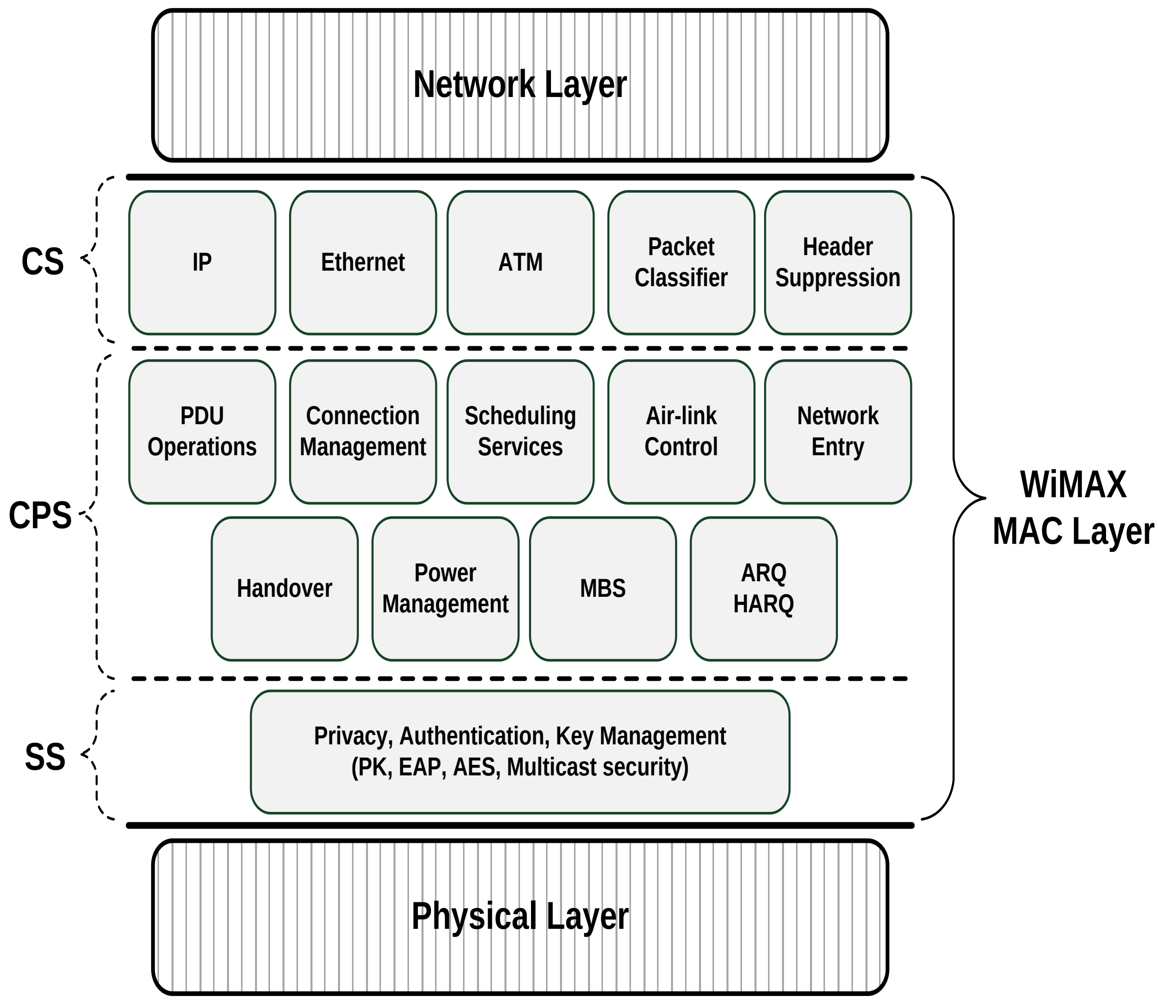

Figure 2.3: WiMAX MAC Layer Functionalities

WiMAX MAC sub-layers functionalities are briefly discussed below:

- ***The Convergence Sub-layer (CS):*** The main function of the CS is the classification of incoming higher-layer packets which is done through Service Flow Identifiers (SFIDs) as will be explained later [2]. Packets received from the higher-layer, MAC service data units (MSDUs), are organized into MAC protocol data units (MPDUs) for transmission over the air. The reverse is done for packets received from the lower-layer. The CS also supports MSDU header suppression to reduce higher-layer overheads on each packet. The CS provides interface with a variety of higher-layer protocols such as ATM, IP, Ethernet and any unknown future protocols.

- ***The Common Part Sub-layer (CPS):*** The CPS is considered the main sub-layer of the MAC layer, since most of the fundamental functionalities of the MAC are handled in this sub-layer [2]. Some of the functions are: PDU operations, connection management, scheduling, air-link control, network entry, handover, power management, multicast broadcast service (MBS), automatic repeat request (ARQ) and hybrid ARQ.
- ***The Security Sub-layer (SS):*** The SS is responsible for functions like support for privacy, user/device authentication and key management.

The WiMAX MAC layer at the BS is fully responsible for allocating bandwidth to all MSs, in both directions downlink and uplink [5]. For the downlink, the MS is not involved and BS allocates bandwidth to each MS based on the needs of the incoming traffic. For the uplink, bandwidth allocation is based on requests from the MS (demand assignment). The IEEE 802.16e standard support several mechanisms by which a MS can demand bandwidth [2]. This process, which is called *Polling*, can be done either individually (unicast) or in groups (multicast). In multicast polling, the allocated slot for making bandwidth requests is a shared slot. The standard defines a contention mechanism with back-off algorithm to control the access to the shared slot. The bandwidth allocation procedure can be summarized as follows:

- The MS gets a bandwidth request opportunity (unicast or multicast).
- The BS replies to the bandwidth request with a bandwidth request response.

- The MS transmits its bandwidth request message (containing the bandwidth request size).
- On approval, the BS allocates the required radio recourses to the MS and sends a bandwidth grant message.

2.1.5. QoS in WiMAX

The support for QoS is a fundamental part of the IEEE 802.16e MAC layer design [5]. The standard enforces a connection-oriented MAC to achieve advanced QoS control. A connection identifier (CID) is used to uniquely address data transmission over a particular link. Moreover, service flow identifier (SFID) is used to distinguish the unidirectional flow of packets with particular QoS parameters. The BS is responsible for issuing SFIDs and mapping them to unique CID. These QoS parameters could include the following:

- Traffic priority.
- Maximum reserved rate.
- Maximum sustained rate.
- Maximum burst rate.
- Minimum reserved rate.
- Maximum latency.
- Tolerated jitter.
- Scheduling type.
- Request/transmission policy (PDU type and size, bandwidth request mechanism).
- ARQ type.

The IEEE 802.16e standard includes the QoS mechanism in the MAC layer architecture. Among other things, the MAC layer is responsible for scheduling of bandwidth for different users. The MAC layer performs bandwidth allocation based on user requirements as well as their QoS profiles. The standard is designed to support a wide range of applications. These applications may require different levels of QoS. To accommodate these applications, the 802.16e standard has defined five service flow classes [2]. Table 2.3 summarizes the service classes supported in WiMAX along with their associated QoS parameters and application examples.

The Unsolicited Grant Service (UGS) is designed to support real-time applications (with strict delay requirements) that generate fixed-size data packets at periodic intervals, such as T1/E1 and VoIP without silence suppression. The guaranteed service is defined so as to closely follow the packet arrival pattern. Uplink grants are granted by the BS regardless of the current estimation of backlog; hence, UGS connections use the unsolicited granting bandwidth-request mechanism. Thus UGS connections never request bandwidth. It is given periodic bandwidth without any polling or contention. The grant size is computed by the BS based on the minimum reserved traffic rate, which is defined as the minimum amount of data transported on the connection when averaged over time. If additional bandwidth is required, the MS may request the BS to poll it to allocated bandwidth.

Table 2.3: WiMAX QoS Service Classes

Service Class	Description	QoS Parameters	Application Examples
UGS Unsolicited Grant Service	Support fixed-sized packets at constant bit-rate	- Max sustained rate - Max latency - Jitter Tolerance - Req/Tx policy	VoIP without silence suppression
rtPS Real-Time Polling Service	Support service flows the generate variable-size packets periodically	- Max sustained rate - Min reserved rate - Max latency - Req/Tx policy	Audio and video streaming, MPEG
nrtPS Non-Real-Time Polling Service	Support delay-tolerant data streams	- Traffic priority - Max sustained rate - Min reserved rate - Req/Tx policy	FTP
ertPS Extended Real-Time Polling Service	Support real-time applications with variable data rates but require guaranteed data rate and delay.	- Traffic priority - Max sustained rate - Min reserved rate - Max latency - Jitter Tolerance - Req/Tx policy	VoIP with silence suppression
BE Best Effort	Support data streams that do not require minimum service level guarantee	- Traffic priority - Max sustained rate - Req/Tx policy	Data transfer, Web browsing, etc.

The Real-Time Polling Service (rtPS), is designed to support real-time applications (with less stringent delay requirements) that generate variable-size data packets at periodic intervals, such as Moving Pictures Expert Group (MPEG) video and video conference [2]. The key QoS parameters for rtPS connections are the minimum reserved traffic rate, which has the same meaning as with UGS, and the maximum latency, which upper bounds the waiting time of a packet at the MAC layer. Since the size of arriving packets with rtPS is not fixed, as it is with UGS-tailored applications, rtPS connections are required to notify the BS of their current bandwidth requirements. The BS periodically grants unicast polls to rtPS connections. The polling period may be explicitly specified as an optional QoS parameter, namely, the unsolicited polling interval.

The ertPS (Extended Real-Time Polling Service) class was added by the 802.16e amendment. The standard [2] indicates that ertPS is a scheduling mechanism that builds on the efficiency of both UGS and rtPS. The BS provides unicast grants in an unsolicited manner like in UGS, thus saving the latency of a bandwidth request. However, whereas UGS allocations are fixed in size, ertPS allocations are dynamic. The ertPS is suitable for variable rate real-time applications that have data rate and delay requirements. An example is voice with activity detection.

Unlike UGS and rtPS scheduling services, the Non-Real-Time Polling Service (nrtPS) and Best Effort service (BE) are designed for applications that do not have any specific delay requirement [2]. The main difference between the two is that nrtPS connections are reserved a minimum amount of bandwidth (by means of the minimum reserved traffic rate parameter), which can boost performance of bandwidth-intensive applications, such as

File Transfer Protocol (FTP). Both nrtPS and BE uplink connections request bandwidth by either responding to broadcast polls from the BS or piggybacking a bandwidth request on an outgoing PDU. These requests are contention based.

The below procedure summarizes the QoS mechanism in WiMAX, which is also illustrated in Figure 2.4:

1. Each MS trying to communicate to the BS will be allocated a basic duplex communication channel with a specific Connection Identifier (CID) number.
2. Packets possessing the same CID number but different QoS parameters are classified into Service Flows (SF). Since there may be more than one SF then each SF is assigned a Service flow Identifier (SFID). The classification of packets into different service flows is done on the basis of Type of Service (ToS) region of each particular IP packet. This classification is done in the MAC CS sub-layer.
3. The required bandwidth, or the bandwidth request message, is then sent to the BS to allocate uplink resources for the sending MS. The Admission Control Unit (ACU) handles the bandwidth granting task. This is done by simply reducing the total available system bandwidth by the current bandwidth request size. If the result of the operation is greater than or equal to zero, then bandwidth is granted. Otherwise the MS is rejected.
4. Then the request is put into 5 different classes of queues to be scheduled. Each queue representing a scheduling service type. In other words, bandwidth request messages are classified according to their scheduling service types, and hence the associated QoS parameters.

5. The scheduling procedure is determined by the embedded scheduling algorithm.
6. Afterwards, the scheduled packets are used to construct the UL-MAP message.
7. An UL-MAP message is broadcasted in the beginning of each OFDMA frame back to the MSs.
8. All MSs listen to the broadcasted MAP message. If a MS found its CID addressed in the UL-MAP then this means that MS can now send its information in the slot mentioned in the UL-MAP.

2.1.6. Mobility Support in WiMAX

Mobility support is the major development presented by the IEEE 802.16e which amended the preceding standard, IEEE 802.16d, basically to address mobility requirements in WiMAX networks. The standard envisions four mobility scenarios [5]:

- ***Nomadic:*** Fixed SS that can connect from different points of attachment (PoA).
- ***Portable:*** Portable SS (like PCs) with best effort handover.
- ***Simple Mobility:*** MS (speed up to 60 Km/h) with less than 1 second interruption during handover.
- ***Full Mobility:*** MS (speed up to 120 Km/h) with seamless handover (<50ms latency & <1% packet loss).

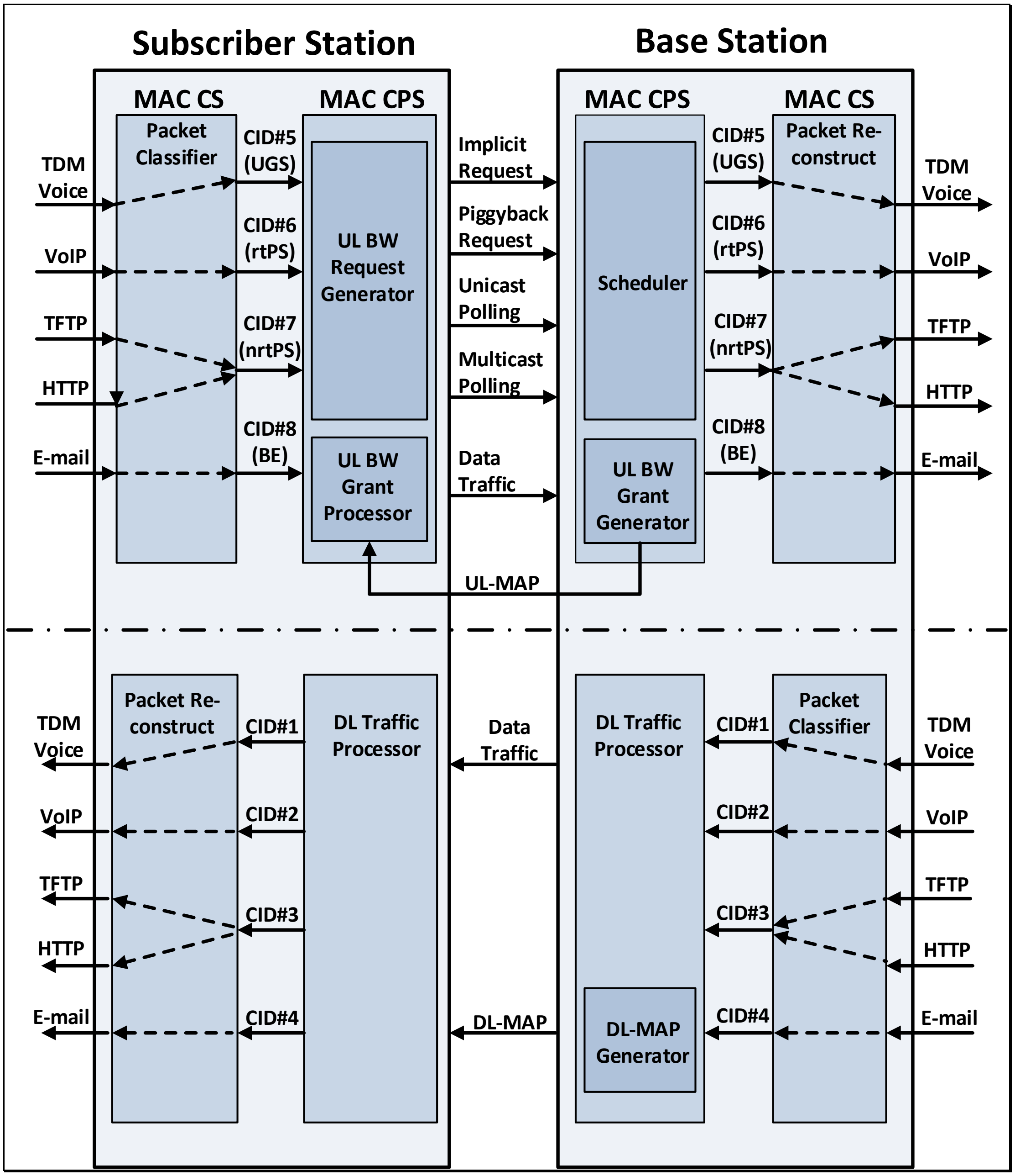

Figure 2.4: WiMAX QoS Mechanism

The mobile WiMAX standard defines signaling mechanisms for tracking the MSs as they move from the coverage area of one BS to another when active, or as they move from one paging group to another when idle. The standard also defines protocols to enable seamless handover of ongoing connections from one BS to another [1] [2]. The WiMAX Forum has used the mobile WiMAX framework to develop mobility management within an end-to-end network architecture framework that also supports IP-layer mobility using mobile IP [4].

In general, there are three methods of handover supported by mobile WiMAX; the first is mandatory and the other two are optional, as explained below:

- ***Hard Handover (HHO):*** Is the mandatory method required by mobile WiMAX. In HHO handover decisions are made based on results reported by the MS. The MS periodically performs a RF scan and measures the signal quality of neighboring BSs. Once the handover decision is made, the MS begins synchronizing with the downlink transmission of the selected BS, perform ranging if it was not done while scanning, and then terminates the connection with the previous BS.
- ***Fast Base Station Switching (FBSS):*** Here the MS maintains a valid connection simultaneously with a set of BSs, called the *active set*. The MS continuously monitors the active set, performs ranging, and maintain a valid CID with each BS in the active set. However, the MS communicates with only one BS, called the *anchor BS*. The handover decision is made as follows: the MS simply reports the selected anchor BS on the CQICH, and then the connection is switched from one BS to another without having to explicitly perform handover signaling.
- ***Macro Diversity Handover (MDHO):*** Here the mechanism is similar to FBSS except that the MS communicates with all BSs in the set, which is called the *diversity set*. In the downlink, multiple copies received at the MS are combined using diversity-combining techniques. In the uplink,

the MS sends data to multiple BS and diversity selection is performed to pick the best uplink.

FBSS and MDHO have superior performance compared to HHO, but require the involved BSs to be synchronized, use the same carrier frequency and share network entry related information.

2.2. VoIP Overview

Voice over Internet Protocol (VoIP) is a methodology and group of technologies for the delivery of voice communications and multimedia sessions over IP networks, such as the Internet [7]. It is undoubtedly a powerful and innovative communication tool, compared with the public switched telephone network (PSTN), VoIP offers the benefit of reducing communication and infrastructure cost. This makes it possible for everyone to easily keep in touch with family, friends, and clients around the world. Furthermore, VoIP has the potential to create new and attractive communication tools by integrating with various applications, such as Web systems, presentation software, and photo viewers. VoIP is available on many Smartphone's, personal computers, and on Internet access devices. Calls and SMS text messages may be sent over 3G or Wi-Fi [7].

VoIP uses a combination of protocols for delivering phone data over networks. Various signaling protocols are used; SIP and H.323 can be regarded as the enabler protocols for voice over IP (VoIP) services [8]. 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 putted in data packets using the RTP protocol. The RTP packets, enclosed inside the UDP packets, are then transferred to the receiver.

2.2.1. VoIP Protocol Stack

The voice over IP (VoIP) protocol stack is generically broken into two categories, control plane protocols and data plane protocols [7]. The control plane portion of the VoIP protocol is the traffic required to connect and maintain the actual user traffic (signaling). It is also responsible for maintaining overall network operation (router to router communications). The data plane, voice portion of the VoIP protocol, is the actual traffic that needs to get from one end to another. Within the VoIP suite of protocols, voice packets are commonly referred to as the data plane. Likewise, signaling packets are commonly referred to as the control plane [8].

- ***Data Plane Protocols:*** Both Real-Time Protocol (RTP) and Compressed Real-Time Protocol (cRTP) are currently available using any of the control plane protocols [9]. RTP is the protocol that supports user voice. Each RTP packet contains small sample of the voice conversation. The size of the packet and the size of the voice sample inside the packet will depend on the codec used. RTP information is encapsulated in a UDP packet. If an RTP packet is lost or dropped by the network, it will not be retransmitted (as is standard for UDP).This is because a user would not want a long pause or delay in the conversation due to the network or the phones requesting lost packets. The network should be designed, so that few packets are lost in transmission. A variant of RTP is compressed RTP (cRTP), which eliminates much of the overall packet header. By eliminating this overhead, a more efficient packet is placed onto the network. With a system running cRTP, a user can place approximately twice as many calls as compared to a system running standard RTP.

- ***Control Plane Protocols:*** The control plane is used for the various signaling protocols, allowing users of VoIP to connect their phone calls. There are several different types of VoIP signaling available today, including H.323, SIP, SCCP, MGCP, MEGACO, and SIGTRAN [9]. The most prevalent types of signaling protocols today, H.323 and SIP. H.323 was the first widely adopted and deployed VoIP protocol suite. The H.323 standard was developed by the International Telecommunications Union-Technology Standardization Sector (ITU-T) for transmitting audio and video over the Internet. Session Initiation Protocol (SIP) is designed to manage and establish multimedia sessions, such as video conferencing, voice calls, and data sharing. SIP is still in its early stages of deployment and is a growing and evolving protocol standard. This is the standard that many element manufacturers are using to develop products [7].

2.2.2. VoIP Codecs

RTP and UDP protocols are the logical choice to carry voice when TCP protocol favors 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 accomplished by one of various audio codecs, each of which vary in the sound quality, the bandwidth required, the computational requirements, encoding algorithm and coding delay [9]. Table 2.4 shows some features of the most common codecs: G.711, G.723.1 and G.729.

- **G.711** is the default standard for all vendors, very low processor requirements. This standard digitizes voice into 64 Kbps and does not compress the voice; it performs best in local networks where we have lots of available bandwidth.

- **G.729A** is supported by many vendors for compressed voice operating at 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 6.3 and 5.3 Kbps. High compression with high quality audio. Although this standard reduces bandwidth consumption, voice is much poorer than with G.729 and is not very popular for VoIP.

Table 2.4: VoIP codecs Characteristics

Codec	Algorithm	Delay (ms)	Bit Rate (kbps)	Packets Per Second	Packet Size (bytes)
G.711	PCM	0.375	64	100	120
G .729A	ACELP	35	8	100	50
G.723.1	CS-ACELP	97.5	5.3	33	60

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CHAPTER THREE

SYSTEM MODEL
&
SIMULATION

CHAPTER 3
SYSTEM MODEL & SIMULATION

3.1. Methodology Description

The analysis of VoIP performance in a network can be performed through various methods. However, simulation will be used because it has several advantages:

- Normal analytical techniques make use of extensive mathematical models which require assumptions and restrictions to be placed on the model. This can result in an avoidable inaccuracy in the output data. Simulations avoid placing restrictions on the system and also take random processes into account; in fact in some cases simulation is the only practical modeling technique applicable.
- Analysts can study the relationships between components in detail and can simulate the projected consequences of multiple design options before having to implement the outcome in the real-world.
- It is possible to easily compare alternative designs so as to select the optimal system.
- The actual process of developing the simulation can itself provide valuable insights into the inner workings of the network which can in turn be used at a later stage.

On the other hand, the disadvantages of simulation have also been considered and were the main drive for the selection of a reliable simulation tool:

- Accurate simulation model development requires extensive resources.

- The simulation results are only as good as the model and as such are still only estimates / projected outcomes.
- Optimization can only be performed involving a few alternatives as the model is usually developed using a limited number of variables.
- Simulations cost a lot of money to build and are very expensive to make.

Equipped with its high fidelity modelling, incomparable model libraries, flexible and scalable wireless models, powerful simulation capabilities and sophisticated analysis; OPNET Modeller 14.5A has been chosen as a simulation tool. In addition OPNET has the advantage of providing a specialized WiMAX model suite that includes a discrete event simulation model designed to analyze network performance in wireless metropolitan area networks. The WiMAX model suite includes the features of the IEEE 802.16e standard.

Using OPNET, several WiMAX network scenarios have been developed to study and investigate VoIP performance. The different scenarios are aimed and designed to address several aspects of the WiMAX network, such as: available bandwidth, network capacity & load, QoS classes, data traffic and mobility. In general each scenario consists of the following models:

- ***The network model,*** defines the network architecture: number of cells (BS's), number of nodes (MS's), transmission power, backbone links, mobility, application servers, etc…
- ***The service model,*** defines the services available on the network (data, voice, video, etc…) and the services available for each node.
- ***The QoS configuration model,*** defines WiMAX QoS configurations for each application and node.

Several simulation runs have been performed for each scenario. Here great consideration has been taken to the following: length of runs, model "settle-down time", initial variables values and random numbers generation.

3.2. OPNET Simulation

OPNET Modeler 14.5A is used to create a system model that is used as a simulation test bed to evaluate the performance of VoIP over mobile WiMAX network. The system model consists of a network model incorporated with different scenarios to investigate the effects of mobility, WiMAX QoS service classes, number of users and voice codecs on VoIP performance. General aspects of the simulation process are described below:

- ***Network Services:*** The simulation assumed one type of services only, which is VoIP.
- ***Network Load:*** Simulation scenarios were designed by varying the network size, i.e. number of mobile stations, to generate different traffic load volumes.
- ***WiMAX QoS:*** In our simulation, we tested WiMAX UGS, rtPs and BE service classes to evaluate the best class to carry VoIP.
- ***VoIP Codec:*** The simulation considered three codecs: G.711, G.723.1 and G.729A to investigate VoIP performance.
- ***Mobility model:*** In our simulation, we choose the Random Way Point model [12] to evaluate mobility effects on quality of voice for users under different speeds. the implementation and configuration of this mobility model is as follows: as the simulation starts, each MS selects

destination randomly and moves toward with constant speed distributing over (0, MaxSpeed) where Max speed is the maximum allowable speed. The speed and direction of mobile node is chosen randomly and indecently of other mobile stations in a system. Once the mobile station reaches its selected destination the mobile station paused for duration of some time and then again chooses another random destination and starts moving towards it at the speed of (0, MaxSpeed). The whole process is repeated again and again until the simulation ends.

3.2.1. WiMAX Network Model

The WiMAX network model consists of one cell and an IP backbone. The cell radius is set to 5 Kilometers. The cell consists of one Base Station (BS) and a number of Mobile Stations (MS) depending on the simulation scenario. There is a server backbone containing only one Voice Server. Figure 3.1 illustrates the WiMAX network model considered in the simulations. In addition, Table 3.1 presents the general parameters of the WiMAX network model. The parameters of Base Station (BS) and Mobile Station (MS) can be seen in the Figure 3.2 and Figure 3.3. The parameters of the WiMAX Configuration Node are as shown in Figure 3.4. Figure 3.5 shows the mobility model configurations.

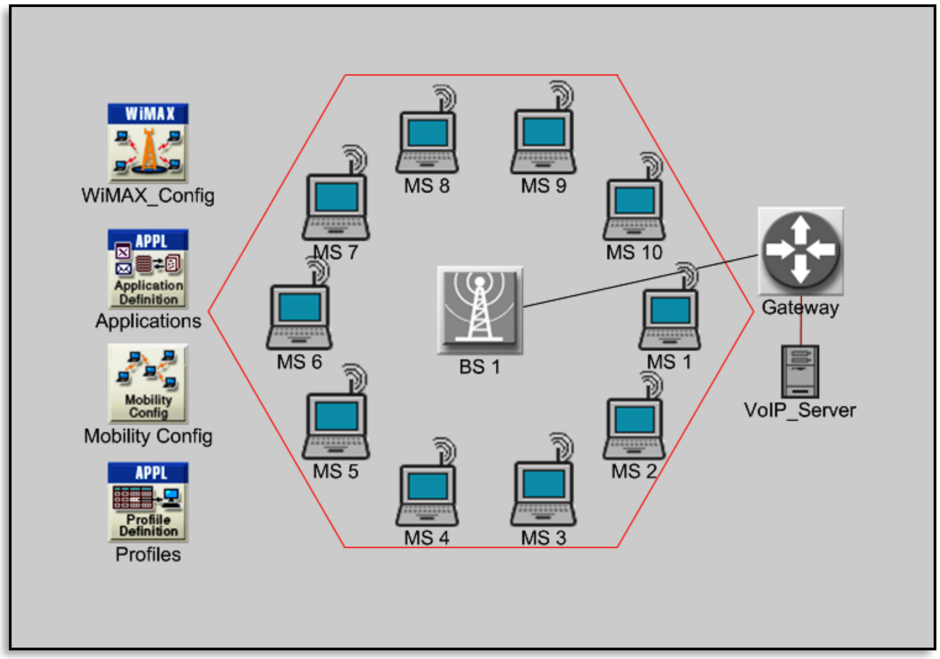

Figure 3.1: WiMAX Network Model

Table 3.1: General Simulation Parameters

Parameter	**Value**
Terrain Size	10 x 10 km
Number of cells	1
Cell Radius	5 km
Number of BSs	1
Number of MSs	2, 4, 6, 8 & 10
Operating Frequency	5.8 GHz
System Bandwidth	20 MHz
PHY Scheme	OFDMA
Duplex scheme	TDD
Modulation Technique	Adaptive
Mobility Model	Random Way Point (RWP)
Fading Model	Rayleigh Fading Model
Multipath Channel Model	ITU Vehicular A
BS transmit power	1 W
MS transmit power	0.5 W
QoS Service Class	UGS, rtPS & BE
Application	VoIP
Voice Codec	G.711, G.723.1 & G.729A
Simulation time	1000 Seconds

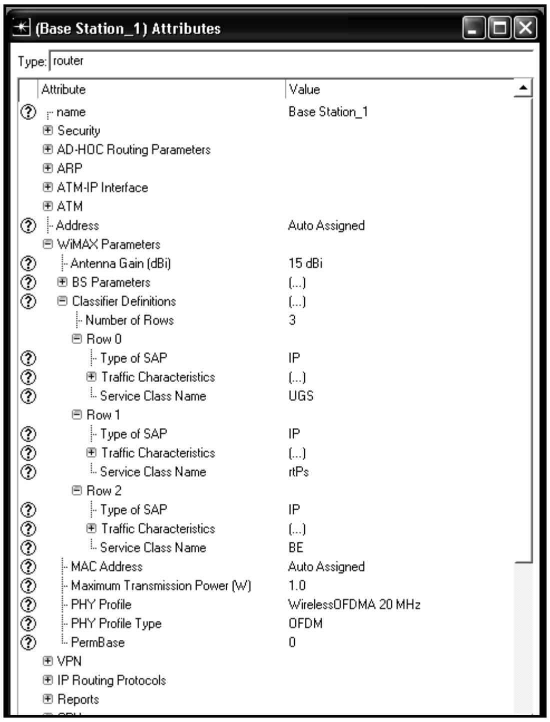

Figure 3.2: Base Station Configuration

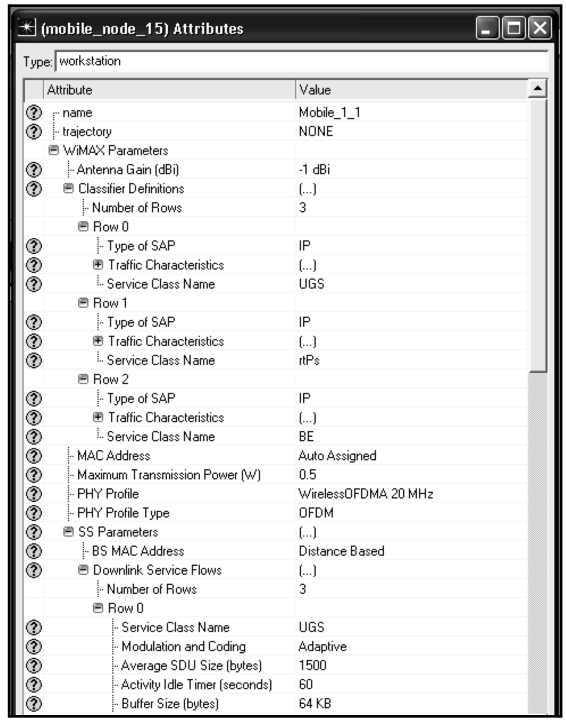

Figure 3.3: Mobile Station Configuration

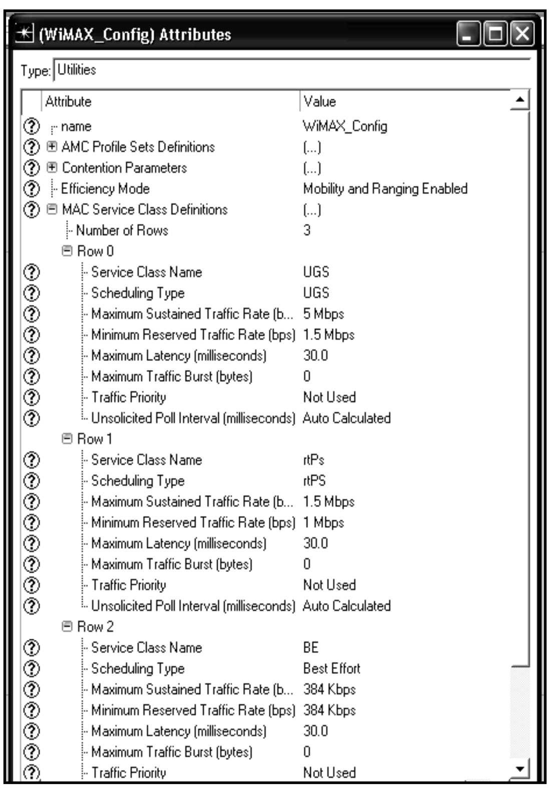

Figure 3.4: WiMAX Configuration

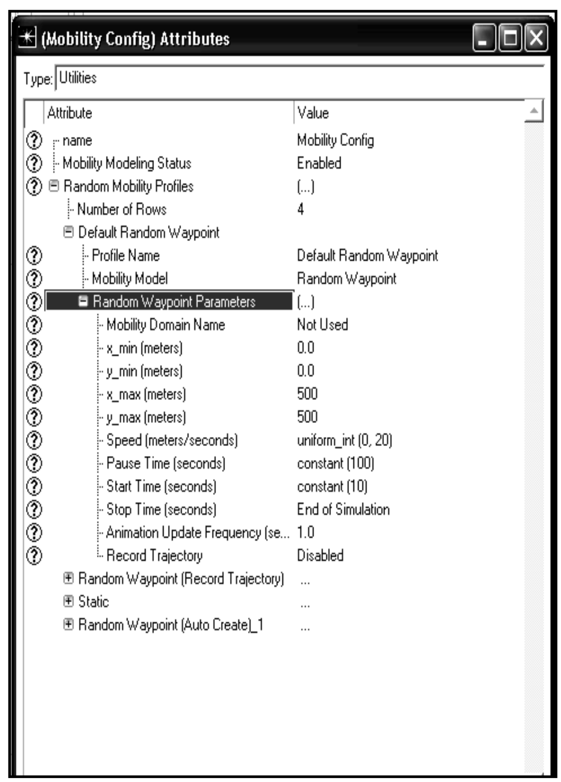

Figure 3.5: Mobility Configuration

3.2.2. Simulation Scenarios

The WiMAX network model described in the previous section has been used to create two simulation scenarios in order to analyze the performance of VoIP over mobile WiMAX.

- ***Scenario (1):*** in this scenario VoIP performance has been investigated with regards to WiMAX QoS configurations. Different WiMAX service classes have been tested: BE, rtPS and UGS to evaluate the best class to carry VoIP. The number of mobile stations (MS) has been varied: 2, 4, 6, 8 and 10 Nodes. The scenario assumed only one codec: G.711.

- ***Scenario (2):*** in this scenario VoIP performance has been investigated with regards to voice codecs. Different voice codecs have been tested: G.711, G.723.1 and G.729 to evaluate VoIP performance. The number of mobile stations (MS) has been varied: 2, 4, 6, 8 and 10 Nodes. The scenario assumed only one WiMAX service class which has resulted from Scenario (1).

3.3. VoIP Performance Metrics

In VoIP, quality means the ability to talk and listen clearly without any unwanted noise. The major factors that affect the speech quality in VoIP are Throughput, Jitter, MOS and Delay or Latency [22].

3.3.1. Throughput

Throughput is the amount of number of packets effectively transferred in a network, in other words throughout is data transfer rate that are delivered to all terminals in a network. It is measured in terms of packets per second or per time slot [21]. Throughput can be calculated using Equation 3.1:

$$\boldsymbol{Throughput_{bps}} = \frac{\boldsymbol{Total\ Bytes\ Received * 8\ (bit)}}{\boldsymbol{End\ Time\ (s) - Start\ Time\ (s)}} \qquad \textbf{(3.1)}$$

3.3.2. Mean Opinion Score (MOS)

The most common measurement metric of voice quality is the MOS [14]. The relationship between audio performance characteristics and a quality score makes MOS a valuable standard for network assessments, benchmarking, tuning, and monitoring. The MOS value can be ranged from 1 bad to 5 excellent. MOS is calculated using a non-linear mapping from R-factor [10], as in Equation 3.2:

$$\boldsymbol{MOS = 1 + 0.35\ x\ R + 7x10^{-6}[R\ (R - 60)(100 - R)]} \qquad \textbf{(3.2)}$$

Where: $R = 100 - I_s - I_e - I_d + A$

$\boldsymbol{I_s}$: the effect of impairments that occur with the voice signal, $\boldsymbol{I_e}$: the impairments caused by different types of losses occurred due to codec's and network, and $\boldsymbol{I_d}$: represents the impairment caused by delay particularly mouth-to-ear delay.

3.3.3. Jitter

Jitter can be observed as the end-to-end delay variation between two consecutive packets. The value of jitter is calculated from the end to end delay. Jitter reveals the variations in latency in the network caused by congestion, route changes, queuing, etc. It determines the performance of network and indicates how much consistence and stable the network is [22]. Jitter is defined as the signed maximum difference in one-way delay of the packets over a particular time interval. It can be calculated using Equation 4.3:

$$\boldsymbol{Jitter = Max_{1\le i\le n}}\{[\boldsymbol{t'(n) - t'(n-1)}] - [\boldsymbol{t(n) - t(n-1)}]\} \qquad \textbf{(4.3)}$$

Where: $t(i)$ and $t'(i)$ are the time transmitted at the transmitter and the time received at the receiver, respectively.

3.3.4. Delay

Delay or latency represents the time taken by a packet to reach from source to destination across the network. The main sources of delay can be categorized into: propagation delay, source processing delay, Queuing delay, transmission delay and destination processing delay. Here we have calculated packet end to end delay (De2e) which 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 [22]. De2e is calculated using Equation 3.4:

$$\boldsymbol{D_{e2e} = D_n + D_e + D_d + D_c + D_{de}} \qquad \textbf{(3.4)}$$

Where D_n, D_e, D_d, D_c and D_{de} represent the network, encoding, decoding, compression and decompression delay, respectively.

3.4. Results

Simulation outputs from several runs have been collected and statistically analysed. The simulation results are: throughput, delay, jitter and mean opinion score (MOS). The first three results are used to identify the impact of the mobile WiMAX network configuration on VoIP performance. The last result, MOS, is used to measure VoIP call quality. Results are studied, graphically presented and discussed in the next chapter.

CHAPTER FOUR

RESULTS & DISCUSSION

CHAPTER 4

RESULTS & DISCUSSION

As explained in the previous chapter, two scenarios have been deployed to study and analyze the performance of VoIP over mobile WiMAX networks. Several simulation runs have been performed for each scenario. Simulation outputs from several runs have been collected and statistically analysed. The simulation results are: throughput, delay, jitter and mean opinion score (MOS). Results are studied, graphically presented and discussed in the following sections of this chapter.

4.1. Scenario (1) Results

Scenario (1) has been designed to investigate the best WiMAX service class that enables the highest VoIP performance. To make the comparison between the different service classes, the data gathered from all three service classes is presented on the same chart. This makes it easier to see the behavior of different QoS parameters for the same traffic over different type of service classes. Figure 4.1 through Figure 4.4 show the comparative plots.

Figure 4.1 shows the throughput for all three service classes on a single chart. The throughput for UGS flow is the highest among the three. The reason for this is that UGS service class is designed with constant bit rate traffic in mind. Since a periodic bandwidth is allocated by the BS to the MS, the MS is able to send more data. The throughput for rtPS traffic is better than BE. It is also noticed that once the number of nodes exceeds 8, throughput starts decreasing.

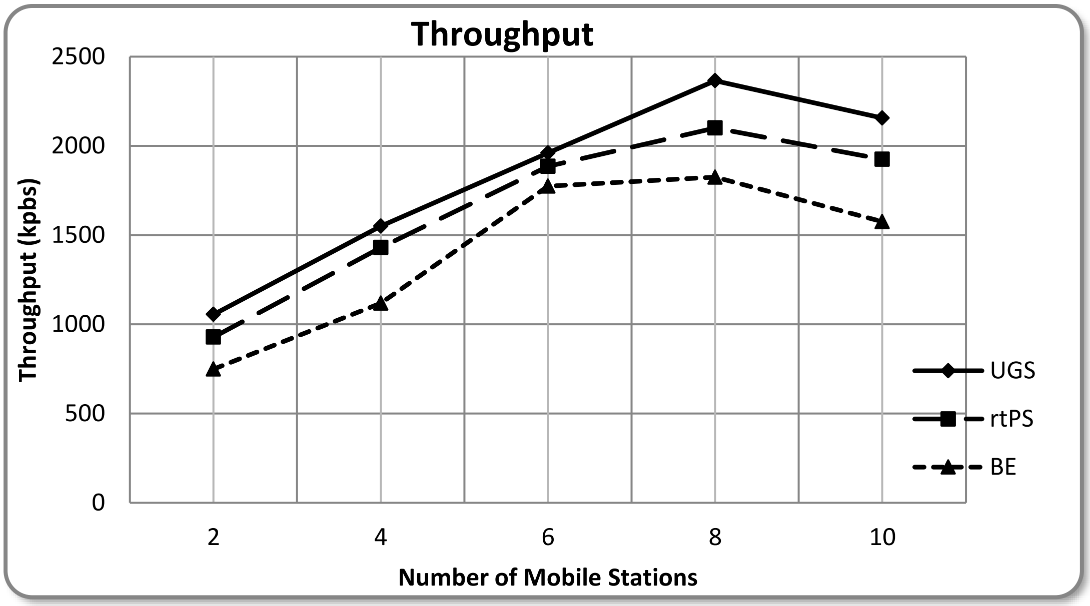

Figure 4.1: Throughput for different service classes

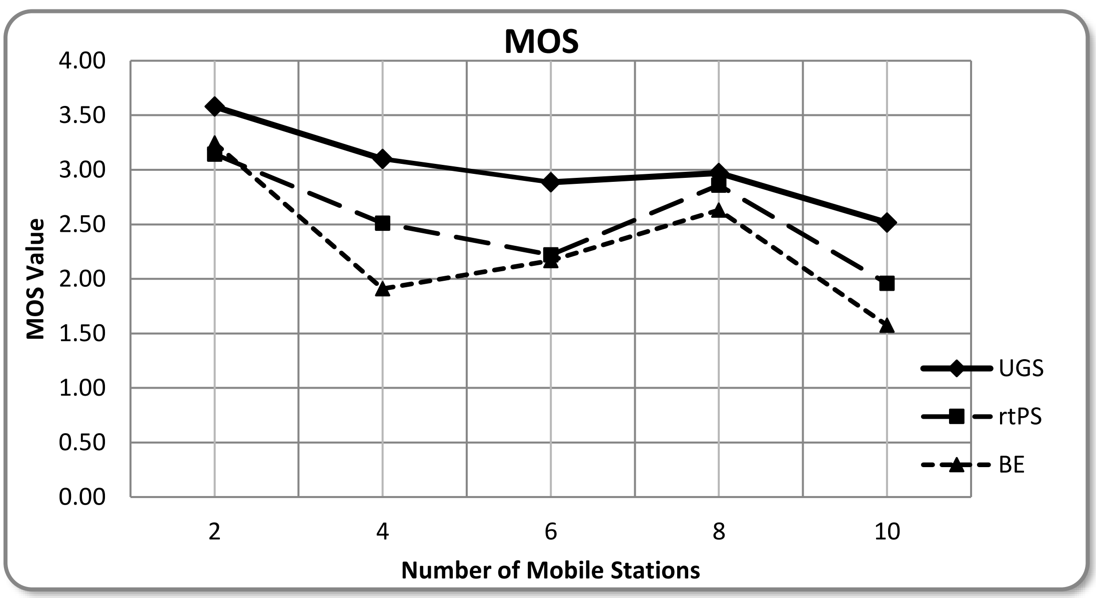

Figure 4.2: MOS for different service classes

Figure 4.3 shows the Mean Opinion Score (MOS) value for all the three service classes. The UGS service class scored the highest MOS value among the three. rtPS has slightly better MOS value than BE. It is also observed that MOS value drops as the number of nodes increases.

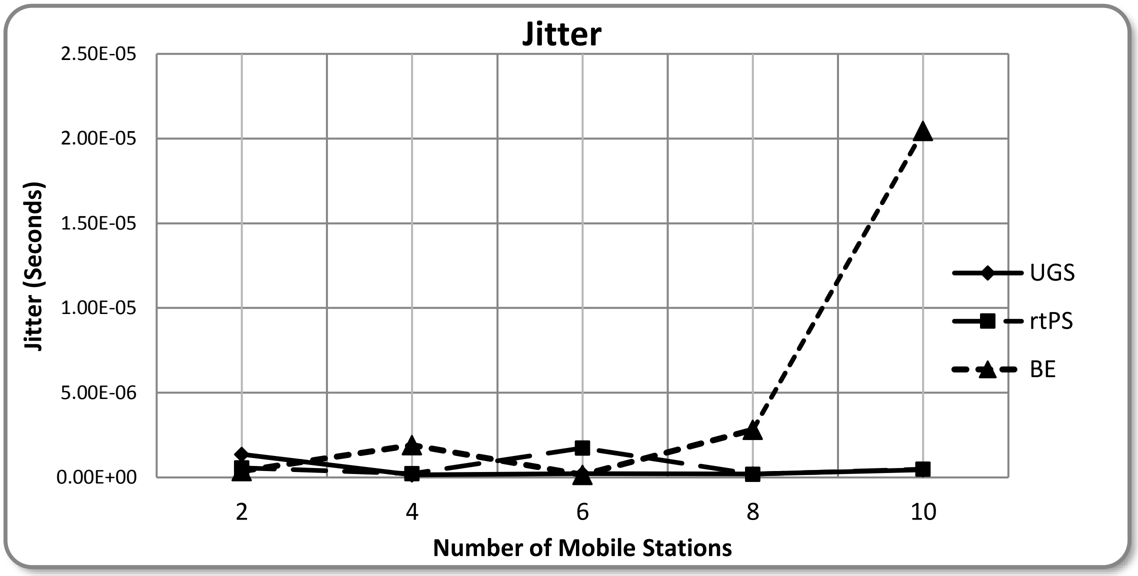

Figure 4.3: Jitter for different service classes

Figure 4.3 shows the average jitter for all the three service classes. The graph is drawn on logarithmic scale to make the comparison easy. BE service class has the highest jitter. The average jitter for UGS service class does not vary as much as the number of nodes increases. Secondly the value is very small. It is also observed that rtPS has low jitter values and falls very close to UGS. Jitter is one of the main parameters for indicating the perceived quality of voice as it reaches the destination node.

Figure 4.4 shows the end-to-end delay for the three service classes. The values for UGS and rtPS service classes fall very close to each other and didn’t exceed 0.09s regardless of the number of nodes. On the other hand, the BE service class has the highest delay and it increases as the number of nodes increases.

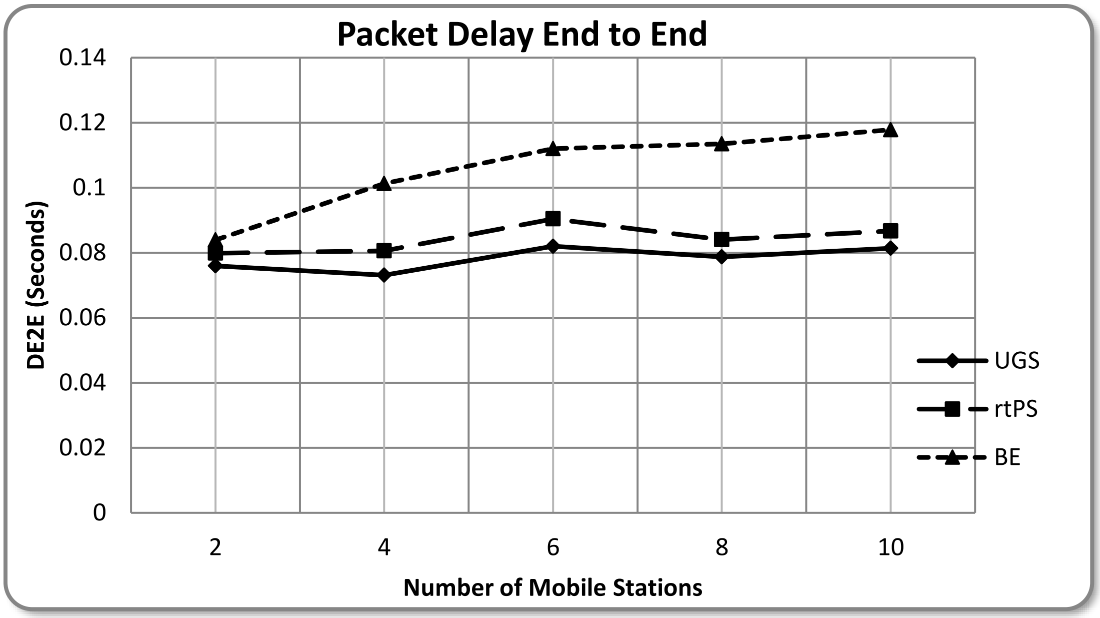

Figure 4.4: DE2E for different service classes

In conclusion, it is observed that UGS service class has the highest throughput, best MOS value and lowest jitter and delay. This makes it best suited for VOIP traffic. Indeed the UGS service class is designed to handle fixed sized packets generated a regular interval and the simulation results validate that.

4.2. Scenario (2) Results

As the results of Scenario (1) revealed, UGS is the best suited WiMAX service class for VoIP traffic. Scenario (2) has been designed to run under the UGS service class while using different voice codecs (G.711, G.723.1 and G.729A) in order to analyze VoIP performance. The data gathered from all codecs is presented on the same chart. This makes it easier to see the performance of VoIP traffic over different type of codecs. Figure 4.5 through Figure 4.8 show the comparative plots.

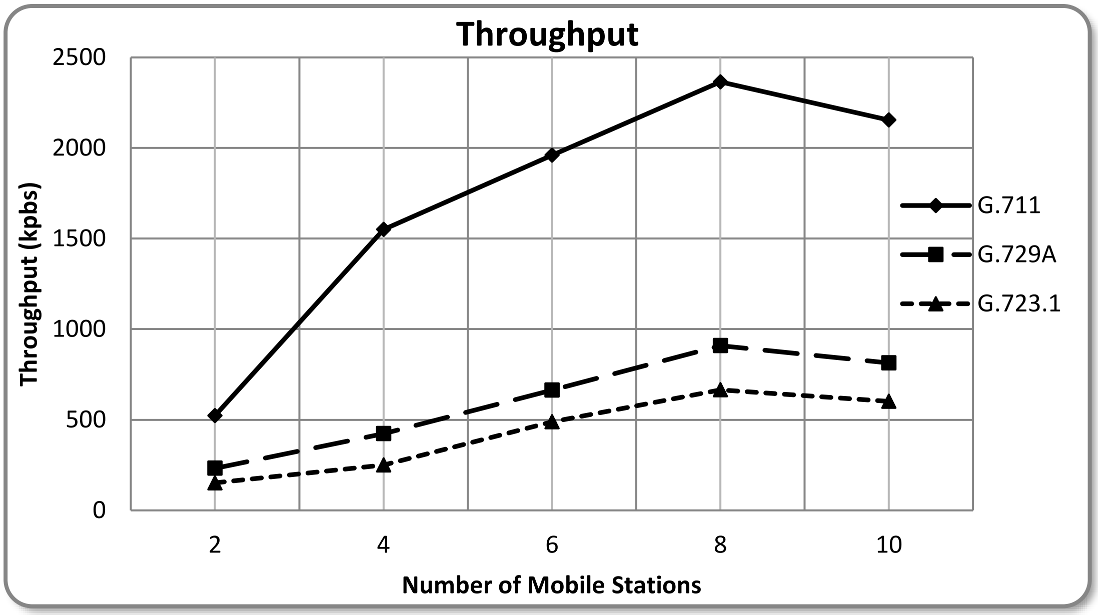

Figure 4.5: Throughput for different VoIP codecs

Figure 4.5 shows the throughput for the different codes. As expected, G.711 codec has reached the highest throughput and this is due to its high bandwidth consumption. The next higher throughput is for G.729A. The G732.1 codec has the lowest throughput.

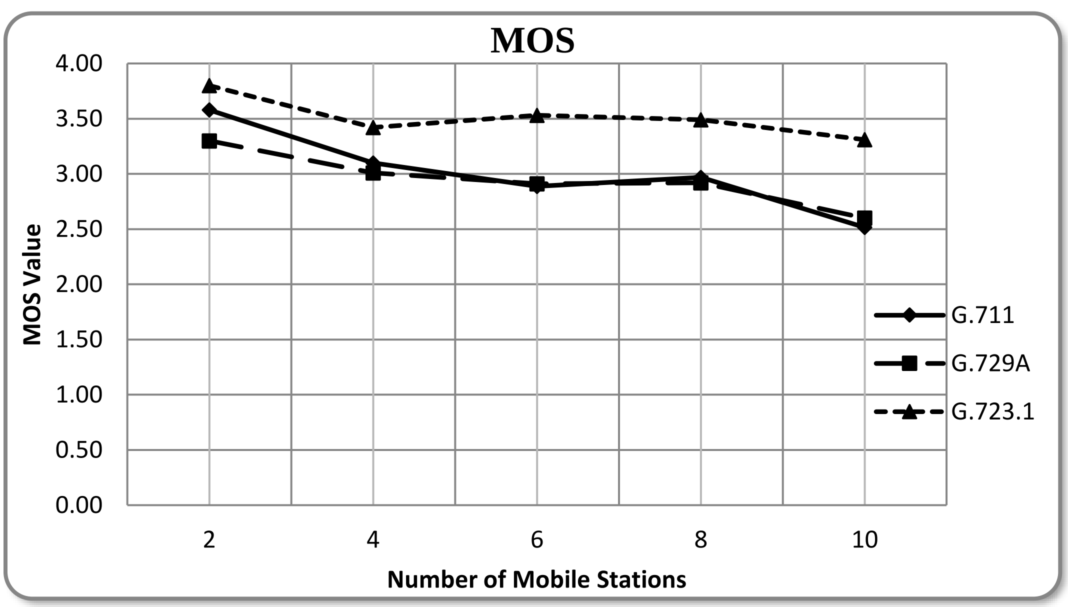

Figure 4.6: MOS for different VoIP codecs

Figure 4.3 shows the Mean Opinion Score (MOS) value for all the three codecs. The G.723.1 codec has the best MOS value among the three. G.711 and G.729A codecs have roughly scored similar MOS values.

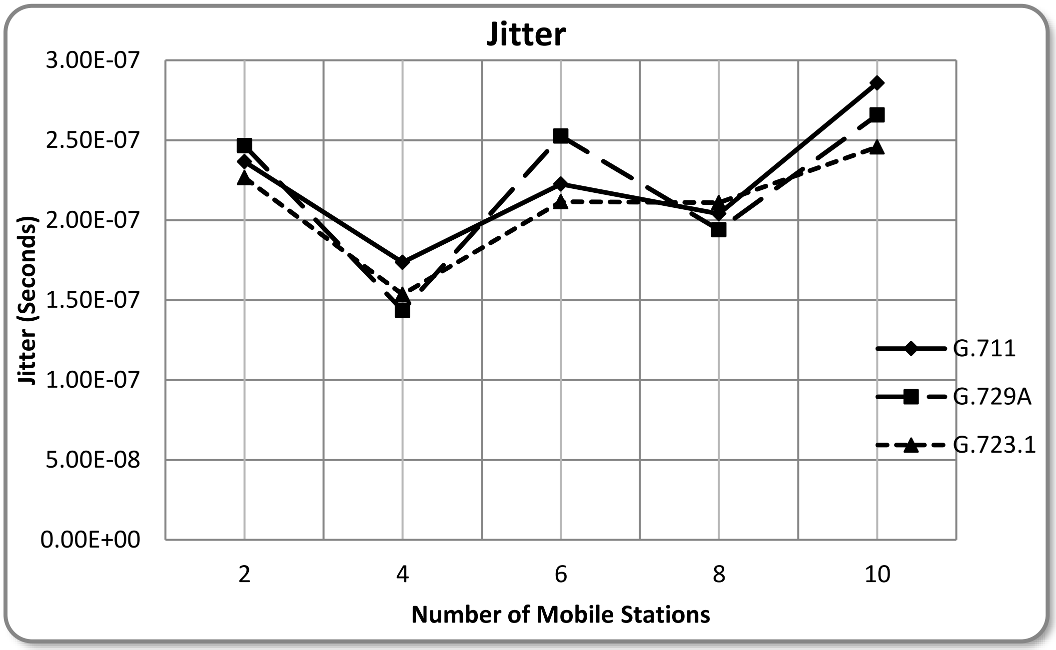

Figure 4.7: Jitter for different VoIP codecs

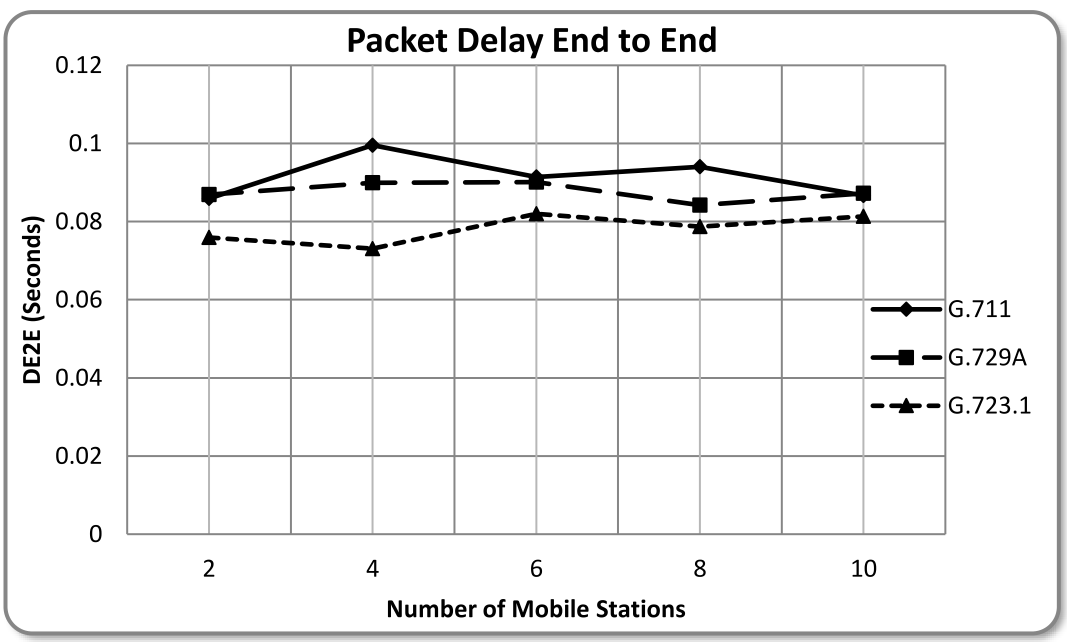

Figure 4.8: DE2E for different VoIP codecs

Figure 4.7 shows the jitter across different VOIP codecs. The values are very small and the pattern indicated is not significant. For all practical purposes, the average jitter is negligible for all the codecs.

Figure 4.8 shows the end-to-end delay across different VOIP codecs. It can be seen that G711 has the maximum and G723.1 has the minimum delay. The values vary in a very small range and there is no significant pattern in the variation as the number of nodes increases.

In conclusion, it is observed that G.723.1 codec has better performance characteristics than the other two codecs. G.723.1 codec scored the best MOS value and lowest delay combined with minimal bandwidth consumption.

CHAPTER FIVE

CONCLUSION
&
RECOMMENDATIONS

CHAPTER 5

CONCLUSION & RECOMMENDATIONS

5.1. Conclusion

In this thesis; we have conducted extensive simulation study to analyze the performance of VoIP over mobile WiMAX networks. We have achieved this through the investigation of WiMAX QoS service classes (UGS, rtPS & BE) and VoIP codecs (G.711, G.729 & G.723.1). We have analyzed several important critical parameters such as throughput, MOS, jitter and end-to-end delay. The results were obtained through computer simulation using OPNET Modeler 14.5A. Simulation results show that UGS service class has the best performance parameters serving VoIP. Indeed, UGS service class is designed to handle real-time service flows that generate fixed size packets at regular interval, which is the case for VoIP. It is also observed that the G.723.1 is better than codecs G.711 and G.729A because it has lower delay and higher MOS meanwhile sustaining minimal bandwidth consumption.

The biggest challenge with the research was the lacking of detailed WiMAX model documentation. This made it difficult to initially prepare the WiMAX scenarios, which required many trials to determine all of the required settings for a successful simulation. In addition, there was a steep learning curve to reach the point of reasonable competency with OPNET. Overall, the project gave a good insight into the technical details of WiMAX while learning the intricacies of the OPNET simulator.

5.2. Recommendations & Future Work

In this thesis we focused on VoIP performance analysis based on the investigation of WiMAX service classes and VoIP codecs. This performance study could be enhanced by the following suggestions:

- Consideration for different traffic types (VoD, FTP, HTTP, etc…).
- Study of ertPS for VoIP with silence suppression.
- Consideration for other mobile WiMAX aspects (mobility patterns/speeds, handoff, large number of mobile stations, transmission power, cell radius, etc…).
- Study of MAC layer scheduling algorithms.
- Study of advanced antenna techniques and MIMO solutions.

Nowadays, WiMAX has taken another step forward and currently a newer standard is under development. The standard is IEEE 802.16m and aims for 1Gbps for nomadic and 100 Mbps for Mobile terminals. This is indeed a revolution in the field of mobile communications and thus further research on this standard would be very fruitful.

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