Dedication

This work is dedicated to my parents, wife, and my lovely son.

Acknowledgement

I wish to express my sincere gratitude to Dr. Sarah M. Eljack, my supervisor for her guidance, encouragement, suggestions and support during the progress of the research and realization of the research. This also extends to all the staff of School of Electronics Engineering and Sudan University of Science and Technology.

Special thanks also go to my parents and my wife for all their support and patience through all the time it took to complete this research.

List of Tables

List of figures

Abbreviations

ASN	Access Service Network

ASN-GW	Access service network gateway
ADPCM	adaptive differential PCM
ACF	Admission Confirm
ADSL	Asynchronous Digital Subscriber Line
AUC	Authentication Centre
AAA	Authentication, Authorization and Accounting
bwrequest	bandwidth request
BS	Base station
BSC	Base Station Controllers
BST	Base Station Transceivers
BE	Best Effort
CS	Circuit Switching
CDMA	code division multiple access
CELP	code-excited linear prediction
CODEC	Compression and decompression
CID	connection identifier
CSN	Connectivity Service Network
CBR	Constant Bit Rate
CN	Core Network
DCHs	dedicated channels
Diffserv	Differentiated Services
DSL	Digital Subscriber Line
DES	Discrete event simulation
EIR	Equipment Identity Register
FDD	Frequency Division Duplexing
GGSN	Gateway GPRS Support Node
GPRS	General Packet Radio Service
GHz	Giga Hertz
GSM	Global System for Mobile Communications
HLR	Home Location Register
IT	Information Technology
Intserv	Integrated Services
IETF	Internet Engineering Task Force
IP	Internet Protocol
IMS	IP Multimedia System
LPC	linear predictive coding
MPDUs	MAC packet data units

MSDUs	MAC service data units
MAC	Media Access Control
MOS	Mean Opinion Score
MAN	metropolitan access network
Mbps	Mega bit per second
MSC	Mobile services Switching Centre
MS	Mobile Stations
MPLS	Multi Protocol Label Switching
MP-MLQ	multipulse-multilevel quantization
NLOS	Non Line-of-sight
nrtPS	non-real-time polling service
OFDM	orthogonal frequency-division multiplexing
PS	Packet Switching
PC	Personal Computer
PHY	physical
PSTN	Public Switched Telephone Network
PCM	Pulse code modulation
QoS	Quality of Service
RNC	Radio Network Controller
ARQ	RAS Admission Request
RRQ	RAS registration request
rtPS	real-time polling service)
RTP	Real-time Traffic Protocol
RAS	Registration Admission Status
RFC	Registration Confirm message
RRJ	Registration Reject message
SAPs	service access points
SGSN	Serving GPRS Support Node
SDP	Session Description Protocol
SIP	Session Initiation Protocol
3GPP	Third Generation Project Partner
3G	Third Generations
TDD	Time Division Duplexing
TCP	Transmission control Protocol
USIM	UMTS subscriber identity module
UTRAN	UMTS Terrestrial Radio Access Network
UMTS	Universal Mobile Telecommunications System

UGS	Unsolicited Grant Service
UDP	User Datagram Protocol
UE	User Equipment
VLR	Visitor Location Register
VLR	Visitor location register
VoIP	Voice Over Internet Protocol
W-CDMA	wideband code division multiple access
CPE	Customer Premise Equipment
WiMAX	World Interoperability Microwave Access

Contents

Chapter One

Introduction

Voice over IP allows users to speak to each other using the Internet (or an intranet) as the transport network. It was invented for personal computer and workstations, (semi-)permanently connected to an internet, to enable people to communicate verbally and simultaneously allow them to use their data application.

Nowadays, telecommunication operators have seen the opportunities (and the threats) VoIP brings and they implemented the VoIP gateways in order to allow communication between a PC and any telephone connected to a PSTN. Some operators even offer a transcending to VoIP for normal voice calls at a reduced cost [1].

1.1. Motivation

The use of VoIP has some important advantages for the operators. Namely, the management of only one network for data and voice and the lower cost of IT equipment compared to telecommunication equipment. Using packet switched (PS) networks for the transportation of voice (real-time data) is also more efficient in terms of bandwidth usage than circuit switched (CS) networks.

Voice over IP (VoIP) was developed at some universities to diminish the cost of long distance calls, for the price of decreased speech quality. Initially, it was considered as a gadget, but nowadays most network operators are taking VoIP seriously and they are investigating and implementing solutions for integrating VoIP and their classical PSTN [2].

It's noticeable that the Internet has changed the way of thinking in telephony networks and will play an even more important role in the future. And it is clear that the internet technologies will also influence the architecture and implementation of future mobile networks [2].

As a real-time digital application, VoIP requires a transmission system that includes low delay, jitter and packet loss rates to ensure that the Quality of Service (QoS) is acceptable [1].

VoIP QoS mechanisms and application level controls have been developed to overcome some of the problems associated with best-effort IP networks and to maximize VoIP call quality including:

• Over-provisioning bandwidth to avoid congestion.

• Network and traffic monitoring to measure and monitor performance.

• Differentiated Services (Diffserv) [3], i.e. packet classification, marking and policing to give priority to certain classes of traffic;

• Integrated Services (Intserv) [4], i.e. the use of signaled QoS to reserve network resources across the network.

VoIP services in wireless networks, such as UMTS and WiMAX, are being progressively implemented. Wireless networks have their own characteristics

[7] which, together with a real-time applications transmission requirement, provide a challenge when a minimum QoS is required.

Considering the current large deployment of WiMAX and UMTS networks and the promising integration of the two networks sooner or later, it is necessary to study their QoS differences and possible ways to resolve the differences between their QoS models.

1.2. Problem statement

This research work includes analysis of selected QoS parameters for VoIP over UMTS and WiMAX networks. To facilitate the analysis a simulated UMTS and WiMAX networks have been created with OPNET simulation software. The next goal was to analyze the simulation results. The results for the UMTS and WiMAX simulations were analyzed and compared to each other.

The research outcome was to take VoIP as an application scenario to study the differences of QoS between UMTS and WiMAX, in order to investigate how well these two networks cope with real-time multimedia applications.

The research results are used to answer the following questions:

How does WiMAX network and UMTS network can provide a real-time application to its users?

How will the VoIP perform when used with WiMAX network?

How will the VoIP perform when used with UMTS network?

What is the best wireless technology to use with VoIP application?

1.3. Research Methodology

In this work the scientific methodology was used which is a combination of theoretical research and computer Simulation using OPNET modeler software.

The research was carried out in several steps:

- Configuring a UMTS network Simulation and implementing a VoIP call on the Simulation model.
- Configuring a WiMAX network Simulation and implementing a VoIP call on the Simulation model.
- Analyzing voice call quality over a UMTS network and a WiMAX network by comparing the simulations results for the both networks against each other and then investigating the results with previous works.
- Finally, a discussion and recommendations of possible future work.

1.4. Research Scope

In the UMTS network technology the scope for this research regarding the implementation of VoIP over a UMTS network, interconnected with the broader digital network, forming a hybrid network for real-time services. In this research, UMTS is considered as a VoIP implementation network for high density voice services.

Also WiMAX network technology is relatively new and the scope for study regarding the implementation of VoIP over a WiMAX network, interconnected with the broader digital network, forming a network for real-time services. In this research work, WiMAX is considered as a VoIP implementation network for high density voice services.

The scope of this research included:

- Background investigation of the current research about VoIP and VoIP QoS over wireless networks.
- Detailed research and investigation of UMTS networks and WiMAX networks.
- Implementation of simulation models using OPNET Modeler including:
    - UMTS network simulation.
    - WiMAX network simulation.
    - VoIP application and traffic configuration.
    - Background traffic configuration.
    - Simulation results collection.
    - Simulation results analysis.
- Comparison of results of the UMTS network simulation with the Wimax network simulation results.
- Results analyses.

1.5. Research Objective

The purpose of this research was to take the VoIP as an application scenario to study the differences of QoS between UMTS and WiMAX networks. The research purpose included the aim to identify a suitable wireless communication network for VOIP and real time applications. This study can be considered as a step towards exploring possible implementations of the next generation wireless networks.

1.6. Assumptions

The research presents an approach to investigate the QoS of VoIP services over a UMTS network and WiMAX network. The experimental data, used in the analysis, was collected from a simulation. A necessary number of assumptions were made to ensure reasonable results can be generated within the research time-frame. The major research assumptions made were:

- The network models and protocols provided and used within the Opnet Modeler simulation application were considered to be reasonable and suitable for the research. References to other research using Opnet Modeler and the network models have been provided.
- The different type of terminals used in a VoIP call will introduce different voice quality, as will the CODECs used to digitize and compress the audio, which reflects different E-Model parameter values. In this research study, it is assumed that typical computer base VoIP terminals are used.

1.7. Research limitations

In this research, there are some limitations which would impact the research results. Some were self-imposed limitations, while others are due to external factors. These limitations include:

- As Opnet Modeler is used to simulate the UMTS and the WiMAX networks the data collected and analyzed would be slightly different from results collected using a real system. Opnet Modeler is one of the leading simulation software applications available today and for this reason the limitations associated with its use are considered acceptable.

- The results have been taken from Opnet modeler where Opnet doing the mathematical and statistical calculation with his own and without the interference of the researcher. So the results have been obtained by Opnet and did not revised by the researcher because it is so difficult to do the calculation manually when the results have been collected by simulation software. However Opnet Modeler is one of the leading simulation software applications available today and for this reason the limitations associated with its use are considered acceptable.

1.8. Research Outline

The research commenced with a literature investigation of key technologies, standards and concepts. Chapter 2 included a background in a section that gives necessary background information about research area; especially details about the VoIP, UMTS and WiMAX technologies. And in Chapter 3 gives the answers for what must to do to achieve the research goals and why? Chapter 4 provided the detailed simulation work, implementing network simulation model in OPNET simulator, analysis and discussion of simulation results included in chapter 5 and the Conclusion of this work in chapter 6 providing what can be learned, did the goals have been met, what are the suggestions about the research area, and what is the untouched area in the research?.

Chapter Two

Literature Review

This Chapter provides a literature review in the areas of wireless networks and voice services and QoS. The key technologies, standards and concepts that form the basis for the research are reviewed and discussed.

This chapter begins with a description of UMTS wireless networks and discussion of the transmission of real-time connection oriented services over a UMTS network. A description of Wimax wireless networks and discussion of the transmission of real-time connection oriented services over a Wimax network is provided next. And then there is a description of VoIP technologies and concepts including VoIP over wireless networks such as UMTS and Wimax. Combining UMTS and Wimax networks and VoIP, the review moves on to considering QoS and how this is measured for wireless networks.

2.1. UMTS

Universal Mobile Telecommunications System (UMTS) is one of the third-generation (3G) cell phone technologies [11]. UMTS is broadly thought as the successor to Global System for Mobile Communications (GSM). It provides more capacity and bandwidth for voice and data services. Most UMTS networks use wideband code division multiple access (W-CDMA) as their underlying air interfaces, so UMTS is also referred as W-CDMA.

The UMTS provides support for both voice and data services [12]. The following data rates are targets for UMTS [13]:

• 144 kbps—Satellite and rural outdoor

• 384 kbps—Urban outdoor

• 2048 kbps—Indoor and low range outdoor

2.1.1 UMTS network architecture

A UMTS network consists of three domains: Core Network (CN), UMTS Terrestrial Radio Access Network (UTRAN) and User Equipment (UE). Figure 2-1 [13] illustrates the architecture of a UMTS network.

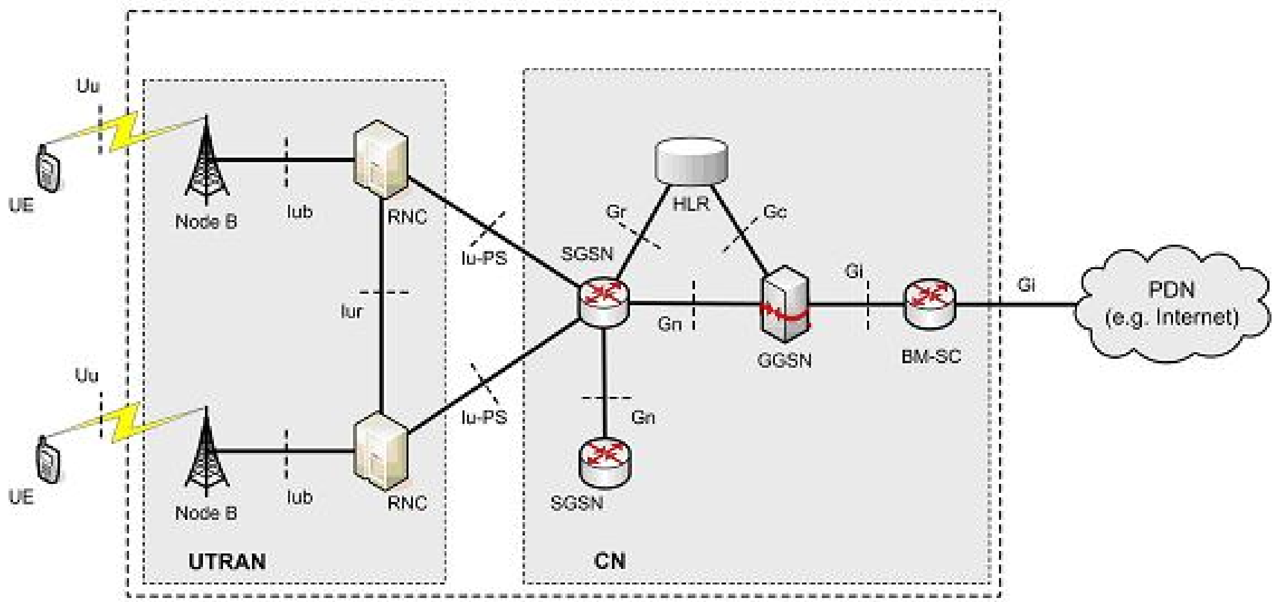


Figure 2-1: UMTS Architecture

2.1.1.1. Core network

The UMTS Core Network is based on GSM/GPRS network. The main functions of the core network are to transport, switch and route user traffic (both circuit switched and packed switched traffic).

The UMTS core network is contains circuit switched and packet switched elements. Circuit switched elements include: Mobile services Switching Centre (MSC), Visitor location register (VLR) and Gateway MSC. Packet switched elements include: Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN). Some network elements, like EIR, HLR, VLR and AUC are both circuit switched and packet switched elements [13].

2.1.1.2. UMTS Terrestrial Radio Access Network (UTRAN)

The UMTS Terrestrial Radio Access Network (UTRAM) includes two network elements: the Radio Network Controller (RNC) and Node B or the base station. The RNC connects to one or more Node B elements. Each Node B can provide service to multiple cells.

The functions of Node-B are [14]:

• Air interface Transmission / Reception.

• Modulation / Demodulation.

• CDMA Physical Channel coding.

• Micro Diversity.

• Error Handling.

• Closed loop power control.

The functions of RNC are [15]:

• Radio Resource Control.

• Admission Control.

• Channel Allocation.

• Power Control Settings.

• Handover Control.

• Macro Diversity.

• Ciphering.

• Segmentation / Reassembly.

• Broadcast Signaling.

• Open Loop Power Control.

2.1.1.3. UMTS User Equipment

The UMTS user equipment (UE) is the mobile equipment with the UMTS subscriber identity module (USIM). The USIM is a card that inserts into the mobile equipment and identifies the subscriber to the core network [13].

The USIM card has provides the following functions [13]:

• Supports multiple user profiles on the USIM.

• Updates USIM information over the air.

• Provides security functions.

• Provides user authentication.

• Supports inclusion of payment methods.

• Supports secure downloading of new applications.

The UMTS UE can operate in one of three modes of operation [13]:

• PS/CS mode—The UE is attached to both the packet-switched (PS) and circuit switched (CS) domain, and the UE can use both PS and CS services in the same time.

• PS mode—The MS is attached to the PS domain and uses only PS services (but allows CS-like services such as voice over IP).

• CS mode—The MS is attached to the CS domain and uses only CS services.

2.1.2. UMTS protocol stack

In this research project, the PS (packet switch) domain of the UMTS network has the full attention. Being looked at from the upper layer, the PS domain functions like normal IP network [16]. As TCP and UDP are both running on top of IP, it will be shown the similar protocol stack if UDP was used instead of TCP [17]. In addition to this, RTP can be implemented to transmit voice [18].

2.1.3. Quality of Service in UMTS Networks

UMTS services will not only offer mobile services supported by 2nd generation systems such as GSM, but also expanded these services to higher rates and greater flexibility. The services evolving in the GSM platform through its Circuit Switched (CS) and Packet Switched (PS) services continued in UMTS with its new services [19].

Thus, future UMTS services will have user transmission rates from low bit up to 2 Mbps [19].

The main four classes of UMTS traffic differentiated by their delay sensitivity are conversational, streaming, interactive, and background [19]. Conversational classes have higher delay sensitivity than background classes. The first two classes correspond to real time classes, while the 2nd two to non-real time. Table 2-1[19] illustrates these classes.

Table 2-1 QoS Classes in UMTS

Traffic class	Conversational	Streaming	Interactive	Background
Characteristics and applications	Preserve time relation between information entities – low delay (e.g. voice, videotelephony	Preserve also time relation between information entities (e.g. multimedia	Request response pattern preserving data integrity. (e.g. Internet or web browsing)	Connectionless, generally packet communications preserving data integrity (e.g. ftp, email, etc.)

2.2. WiMAX

WiMAX is a wireless metropolitan access network (MAN) technology that is based on the standards defined in the IEEE 802.16 specification. WiMAX technology is a telecommunications technology that offers transmission of wireless data via a number of transmission methods; such as portable or fully mobile Internet access via point to multipoint links. The 802.16 standard can be used in a point-to-point or mesh topology using pairs of directional antennas. These antennas can be used to increase the effective range of the system relative to what can be achieved in the point-to-multipoint mode [20].

WiMAX is envisioned as a solution to the outdoor broadband wireless access that is capable of delivering high-speed streaming data. It has the capability of delivering high-speed services up to a range of 30 miles, thus posing strong competition to the existing last mile broadband access technologies, such as cable and DSL. WiMAX uses multiple channels for a single transmission and provides bandwidth of up to 100 Mbps [21].

The use of orthogonal frequency-division multiplexing (OFDM) increases the bandwidth and data capacity by spacing channels very close to each other and still avoids interference because of orthogonal channels.

2.2.1. Types of WiMAX

The WiMAX family (802.16) concentrates on two types of usage models: a fixed WiMAX and mobile WiMAX [22]. The basic element that differentiates these systems is the ground speed at which the systems are designed to manage. Based on mobility, wireless access systems are designed to operate on the move without any disruption of service; wireless access can be divided into three classes; stationary, pedestrian and vehicular.

A Mobile WiMAX network access system is one that can address the vehicular class, whereas the fixed WiMAX serves the stationary and pedestrian classes. This raises a question about the nomadic wireless access system, which is referred to as a system that works as a fixed WiMAX network access system but can change its location.

2.2.1.1. Fixed WiMAX

Because communications takes place via wireless links from WiMAX Customer Premise Equipment (WiMAX CPE) to a remote Non Line-of-sight (NLOS) WiMAX base station, requirements for link security are greater than those needed for a wireless service. The security mechanisms within the IEEE 802.16 standards are sufficient for fixed WiMAX access service [22].

Another challenge for the Fixed WiMAX access air interface is the need to set up high performance radio links capable of data rates comparable to wired broadband service, using equipment that can be self installed indoors by users, as is the case for Digital Subscriber Line (DSL) and cable modems. IEEE 802.16 standards provide advanced physical (PHY) layer techniques to achieve link margins capable of supporting high throughput in NLOS environments.

2.2.1.2. Mobile WiMAX

The 802.16a extension, refined in January 2003, uses a lower frequency of 2 to 11 GHz, enabling NLOS connections. The latest 802.16e task group is capitalizing on the new capabilities this provides by working on developing a specification to enable Mobile WiMAX clients. These clients will be able to hand off between WiMAX base stations, enabling users to roam between service areas.

2.2.2. WiMAX Architecture

WiMAX is based on IEEE standard for high layer protocol such as TCP/IP, VoIP, and SIP etc. WiMAX network is offering air link interoperability.

The Architecture of WiMAX is based on all IP platforms. The packet technology of WiMAX needs no legacy circuit telephony. Therefore it reduces the overall cost during life cycle of WiMAX deployment. The main guidelines of WiMAX Architecture are as follow [22]:

- Support structure of packet switched. WiMAX technology including IEEE 802.16 standard and its modification, suitable for IETF and Ethernet.

- Offers flexibility to accommodate a wide range of deployment such as small to large scale. WiMAX also support urban, rural radio propagation. The uses of mesh topologies make it more reliable. It is the best coexistence of various models.

- Offers various services and applications such as multimedia, Voice, mandated dogmatic services as emergency and lawful interception.

- Provides a variety of functions such as ASP, mobile telephony, interface with multi internetworking, media gateway, delivery of IP broadcasting such as MMS, SMS, WAP over IP.

- Supports roaming and Internet working. It support wireless network such as 3GPP and 3GPP2. It support wired network as ADSL.

- Supports global roaming, consistent use of AAA for billing purposes, digital certificate, subscriber module, USIM, and RUIM.
- The range is fixed, portable, nomadic, simple mobility and fully mobility.

The WiMAX architecture consists of three logical entities [22]:

- Mobile Stations (MS) used by the end user to access the network.
- The access service network (ASN), which comprises one or more base stations and one or more ASN gateways that form the radio access network at the edge.
- Connectivity service network (CSN), which provides IP connectivity and all the IP core network functions.

All three correspond to a grouping for functional entities which may be single or distributed physical device over several physical devices may be an implementation choice. The manufacturer chooses any implementation according to its choice which is may be individual or combine.

2.2.2.1. Base station (BS)

The responsibility of Base station (BS) is to provide that the air interface to the Mobile Station (MS). The other functionality of BS is micro mobility supervision functions. The handoff prompting, supervision of radio resource, classification of traffic, DHCP, keys, session, and multicast group management[22].

2.2.2.2. 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[23].

2.2.2.3. Connectivity Service Network (CSN)

CSN is a set of functions related to network offering IP services for connectivity to WiMAX clients. A CSN may include network fundamentals such as server, routers, and user database and gateway devices that support validation for the devices, services and user. The Connectivity Service Network also handled different type of task such as management of IP addresses, support roaming between different NSPs, management of location, roaming, and mobility between networks management of location, roaming, and mobility between ASNs [22].

The WiMAX architecture framework allows for the flexible decomposition and /or combination of functional entities. For example the ASN may be decomposed into base station transceivers (BST), base station controllers (BSC), and ASNGW analogues to the GSM model of BTS, BSC, and Serving GPRS Support Node (SGSN) [22].

2.2.3. The MAC layer of WiMAX

WiMAX offers some flexible features that can potentially be exploited for delivering real time services. In particular, although the MAC layer of WiMAX has been standardized, there are certain features that can be tuned and made application and/or channel specific [24]. For example, the MAC layer does not restrict itself to fixed size frames but allows variable-sized frames to be constructed and transmitted.

The MAC layer of WiMAX is comprised of three sub layers which interact with each other through the service access points (SAPs), as shown in Figure 2-5. The service specific convergence sub layer provides the transformation or mapping of external network data with the help of the SAP. The MAC common part sub layer receives this information in the form of MAC service data units (MSDUs), which are packed into the payload fields to form MPDUs. The privacy sub layer provides authentication, secure key exchange, and encryption on the MPDUs and passes them over to the physical layer. Of the three sub layers, the common part sub layer is the core functional layer which provides bandwidth and establishes and maintains connection.

Moreover, as the WiMAX MAC provides a connection-oriented service to the subscriber stations, the common part sub layer also provides a connection identifier (CID) to identify which connection the MPDU is servicing.

The sub layer controls the on-air timing based on consecutive frames that are divided into time slots. The size of these frames and the size of the individual slots within these frames can be varied on a frame-by-frame basis. This allows effective allocation of on-air resources which can be applied to the MPDUs to be transmitted. Depending on the feedback received from the receiver and on-air physical layer slots, the size of the MPDU can be optimized [20].

2.2.4. WiMAX Physical Layer

Physical layer sets up the connection between the communicating devices and is responsible for transmitting the bit sequence. It also defines the type of modulation and demodulation as well as transmission power. WiMAX 802.16 PHY-layer considers two types of transmission techniques OFDM and OFDMA. Both of these techniques have frequency band below 11 GHz and use TDD and FDD as its duplexing technology. WiMAX physical layer is based-on the orthogonal frequency division multiplexing (OFDM). OFDM is a good choice of high-speed data transmission, multimedia communication and digital video services. It even can maintain very fast data rate in a non-line of sight condition and multipath environment. The role of the PHY-layer is to encode the binary digits that represent MAC frames into signals and to transmit and receive these signals across the communication media. The WiMAX PHY layer is based on OFDM; which is used to enable high-speed data, video, and multimedia communications and is used by a variety of commercial broadband systems [20].

2.2.5. Quality of service in IEEE 802.16

Originally, four different service types were supported in the 802.16 standard: UGS, rtPS, nrtPS and BE. The UGS (Unsolicited Grant Service) is similar to the CBR (Constant Bit Rate) service in ATM, which generates a fixed size burst periodically. This service can be used to replace T1/E1 wired line or a constant rate service. It also can be used to support real time applications such as VoIP or streaming applications. Even though the UGS is simple, it may not be the best choice for the VoIP in that it can waste bandwidth during the off period of voice calls. The rtPS (real-time polling service) is for a variable bit rate real-time service such as VoIP. Every polling interval, BS polls a mobile and the polled mobile transmits bwrequest (bandwidth request) if it has data to transmit [20].

The BS grants the data burst using UL-MAP-IE upon its reception. The nrtPS (non-real-time polling service) is very similar to the rtPS except that it allows contention based polling. The BE (Best Effort) service can be used for applications such as e-mail or FTP, in which there is no strict latency requirement. The allocation mechanism is contention based using the ranging channel. Another service type called ertPS (Extended rtPS) [10] was introduced to support variable rate real-time services such as VoIP and video streaming. It has an advantage over UGS and rtPS for VoIP applications because it carries lower overhead than UGS and rtPS.

2.3. Comparison between UMTS and WiMAX

UMTS is proposed to converge packet-switched and circuit switched networks. Its IP Multimedia System (IMS) is used for multimedia communications. IMS was originally defined by the Third Generation Partnership Project (3GPP) for the next generation mobile networking applications and uses SIP as the signaling protocol. With the availability of UMTS, more and more phones can use different wireless networks other than Wi-Fi. One example is Fring [25] that uses VoIP over UMTS. So far, four service types have been proposed and incorporated into the QoS model of UMTS:

- Conversational class - for voice/video telephony, with low end-to-end delay and low jitter, two-way
- Streaming class - for streaming video, with low jitter, one-way
- Interactive class - for web browsing, with low loss/error rate, two-way
- Background class (Best Effort) - for email and background download, with low loss/error rate, one-way

The WiMAX wireless technology is called the last-mile solution for wireless broadband access. One of the features of the MAC (Media Access Control) layer of WiMAX is that it is designed to differentiate services among traffic categories with different multimedia requirements [26]. WiMAX offers some flexible features that can potentially be exploited for delivering real-time services. Though the MAC layer of WiMAX has been standardized, there are certain features that can be tuned for specific applications and channels. For example, the MAC layer does not restrict itself to fixed-sized frames, but allows variable-sized frames to be constructed and transmitted. This is very useful for framing VoIP packets [27].

So far, five service types have been proposed and incorporated into the QoS model of WiMAX:

- Unsolicited Grant Service (UGS) - Supports real-time data streams.
- Real-time Polling Service (rtPS) - Supports real-time data streams.
- Non real-time Polling Service (nrtPS) - Supports delay tolerant data with variable packet sizes.
- Best Effort (BE) - Supports data streams where no minimum data rate is required and packets are handled based on available bandwidth.
- Extended real-time Polling Service (ertPS): Scheduling algorithm for VoIP services with variable data rates and silence suppression.

Both WiMAX and UMTS have advantages and disadvantages compared to each other. WiMAX is the first truly open mobile standard (IEEE802.16e) governed by the IEEE's fair licensing practices and open to participation. This is in fact revolutionary since 3GPP and 3GPP2 are consortium and do not allow open participation. This open process should lead to greater innovation and hence a better performance when moving forward and can potentially reduce intellectual property licensing fees, because it provides for a quicker improvement of the technology compared to existing mobile technologies. WiMAX is also the first major mobile standard to offer all-IP network. UMTS will get there in subsequent releases but it still employs a complicated and ultimately expensive core network [29]. Table 2-2 shows comparison of the two networks [28].

Table 2-2: COMPARISON OF WIMAX AND UMTS

Paramater	WIMAX	UMTS
Peak down-link data rate	46Mbps with 3:1 DL-to-UL ratio TDD; 32Mbps with 1:1	14.4Mbps using all 15 codes; 7.2Mbps with 10 codes
Peak up-link data rate	7Mbps in 10MHz using 3:1 DL-to-UL ratio; 4Mbps using 1:1	1.4Mbps initially; 5.8Mbps later
Bandwidth	3.5MHz, 7MHz, 5MHz, 10MHz, and 8.75MHz initially	5MHz
Modulation	QPSK, 16 QAM, 64 QAM	QPSK, 16 QAM
Multiplexing	TDM/OFDMA	TDM/CDMA
Duplexing	TDD initially	FDD
Frequency	2.3GHz, 2.5GHz, and 3.5GHz initially	00/900/1,800/1,900/ 2,100MHz

2.4. IP network and real-time applications

As a voice communication tool, VoIP is a real-time application which is very delay sensitive. For this research subject following a study of the IP networks for real-time applications.

IP networks are "best-effort networks", which look like they are not suitable for Real- Time applications [30]. Although IP networks have had some success in supporting real-time applications as benefit of the widely roll out of broadband Internet access. Real-Time IP applications such as voice, video, and interactive gaming are becoming the wave of growth for IP networks. These bandwidth-intensive applications are referred to be as Real-Time applications because, unlike best-effort applications, they must be transported through a network with minimal delay or latency.

IP networks are highly dynamic. Any outage or change in the network will cause routing path re-calculating for the entire network [30]. This characteristic comes from the original design of the IP networks, which are designed for best-effort applications. It reduces the need for high reliability in the individual network elements (routers) or links, because after finding any outage or change, the network can simply re-converge and find an alternative route. This dynamic and self re converge network model is perfectly suited to best effort applications, such as email, web browsing and non-critical data transmitting, which are not time-sensitive and tolerant to some packet loss.

In contrast, real-time applications, such as voice/video calls, are very delay-sensitive and must be supported by a highly stable network. To support real-time applications, some improvements are required for the IP networks: reliability, stability and faster convergence [31]. With the reliability of the network elements (routers and links) and the stability of the network, fewer interruptions would occur, and when interruptions occur, the faster convergence can reduce the impact. By addressing these three basic requirements, the IP networks can achieve the reliability and stability required to support new Real-Time services.

2.4.1. IP networks to support voice

In order for consumers to accept VoIP, the quality of VoIP calls should be equal to the traditional PSTN voice services. But since VoIP shares the same network with data transmitting either in the Internet or Intranet, it must compete with other applications for the limited network bandwidth. That brings up some requirements that VoIP needs to run over IP networks. VoIP, as a real-time, delay-sensitive application has special performance needs for bandwidth, delay, jitter and packet loss.

2.4.1.1. Bandwidth requirement

Telecommunication companies' quality voice requires sampling at 8 KHz. The bandwidth then depends on the level of quantization. With Linear quantization at 8 bits/sample or at 16 bits/sample, the bandwidth is either 64 Kbps or 128 Kbps.

In order to get VoIP calls at a Telco quality, two different approaches can be attempted. One is to transmit voice in the highest quality, which needs unrestricted bandwidth. Another approach is to transmit voice at a certain quality, which is competitive with PSTN call quality. By using the second approach, the required bandwidth can be reduced to a reasonable low level. Some source data are highly redundant, for example, a digital signal contains many strings of zeroes (or ones), and it will be economical to transmit a code indicating that a string of zero (or one) follows along with the length of the string. Compression and decompression (CODEC) of digital signals is a means of reducing the required bandwidth or transmission bit rate. Many different algorithms for compression and decompression of digital codes have been constructed. Pulse code modulation (PCM) and adaptive differential PCM (ADPCM) are examples of "waveform" CODEC techniques. Waveform CODECs are compression techniques that exploit the redundant characteristics of the waveform itself. In addition to waveform CODECs, there are source CODECs that compress speech by sending only simplified parametric information about voice transmission; these CODECs require less bandwidth. Source CODECs include linear predictive coding (LPC), code-excited linear prediction (CELP) and multipulse-multilevel quantization (MP-MLQ). Coding techniques for telephony and voice packet are standardized by the ITU-T in its G-series recommendations. Some algorithms for voice compression and decompression are given in the Table 2-2.

Table 2-3 Algorithms for voice compression and decompression

Input Range	Transmission Rate	Standard
Linear Predictive Coding Algorithm	64 Kbps	LPC-10, G.711
Code Excited Linear Prediction (CELP)	8 Kbps	G.729, G.729A
32 Kbps Adaptive Differential Pulse Code Modulation (ADPCM)	32 Kbps	G.721

In order to transmit the voice information over IP networks, besides the actual bandwidth the voice data used, some extra bandwidth is required to cover adding the heads for each packet transmitted. These headers are IP, UDP and RTP. An IPv4 header is 20 octets; a UDP header is 8 octets and an RTP header is 12 octets. The total length of this header information is 40 octets (bytes), or 320 bits. These headers are sent each time a packet, containing voice samples, is transmitted. The additional bandwidth occupied by this header information is determined by the number of packets, which are sent each second. For example, if one packet carries the voice samples representing 20 milliseconds, the 50 such samples are required to be transmitted in every second. Each sample carries an IP/UDP/RTP header overhead of 320 bits (Perkins and Crowcroft, 2000). Therefore, in each second, 16,000 header bits are sent, which means an extra of 16 Kbps bandwidth is required. Islam (Islam et al., 2005) provides a detailed description of how to calculate the bandwidth requirements for VoIP over IP network transmission. Table 2-3 shows the IP bandwidth requirements for the most common coding algorithms.

Table 2-4 IP bandwidth requirements for the most common coding algorithms

Coding algorithm		Bandwidth	Sample	IP bandwidth
G.711	PCM	64kbps	0.125ms	80kbps
G.723.1	ACELP	5.6kbps	30ms	16.27kbps
	MP-MLQ	6.4kbps		17.07kbps
G.726	ADPCM	32kbps	0.125ms	48kbps
G.728	LD-CELP	16kbps	0.625ms	32kbps
G.729(A)	CS-ACELP	8kbps	10ms	24kbps

2.4.1.2. Delay

Delay is a vital element for VoIP, because Voice traffic is real-time traffic and if there is too long of a delay in voice packet delivery, speech will be unrecognizable and unacceptable. An acceptable delay is less than 200 milliseconds. Delays are caused by a number of different factors. There are basically two kinds of delay in VoIP networks: Propagation delay and Handling delay.

Propagation delay is caused by the characteristics of the speed of light/electrical signal travelling via a fiber-optic or copper medium of the physical layer of the network. Much cannot be done about the propagation delay, but sometimes it is a big part of the whole delay. It takes light about 100 ms to travel around the Earth, so a call to someone on the other side of the Earth would cause propagation delay of about 50 ms, which is a quart of the maximum acceptable delay.

Handling delay is caused by the devices that handle voice information and have a significant impact on voice quality in a packet network. One big part of the handling delay comes from the network. This delay is an accumulation of queuing delay in network routers and switches.

Handling delay also includes the time a system takes to generate a voice packet, it may take 5ms to 20ms to generate a frame depending on the system, and usually one or more frames are placed in one voice packet. Another contribution to this delay is the time taken to move the packet to the output buffer and the time the packet waiting in the output buffer before being processed. Also, CODECs induce delay as well. There are various coding schemes available. Table 2-4 shows the best and worst case coding delays for the common CODECs. G711 is not listed in the table, because it does not compress the PCM sample and therefore, it does not experience a codec delay [30].

Table 2-5 Codec delay

Codec	Rate	Minimum Sample block	Worst Case Codec Delay
ADPCM, G.726	32 Kbps	10 ms	10 ms
CS-ACELP, G.729A	8.0 Kbps	10 ms	10 ms
MP-MLQ, G.723.1	6.3 Kbps	30 ms	20 ms
MP-ACELP, G.723.1	5.3 Kbps	30 ms	20 ms

2.4.1.3. Jitter

Jitter is defined as a variation in delay of VoIP packets reaching the receiver. The VoIP receiver expects packet flows to arrive at equal intervals of time, so it can play out a continuous voice stream. Any variation in that arrival of a packet creates jitter. Normally jitter can be compensated by using a jitter buffer for playing out the audio smoothly, but this way introduces some extra delay [30].

2.4.1.4. Reliability

Although IP networks are best-effort networks, the traditional data communication provides reliable end-to-end communication between two users by using mechanisms in up-layer protocols (TCP is a good example of this). It uses checksum and sequence numbering for error control and some form of negative acknowledgement with a packet retransmission handshake for error recovery. The negative acknowledgement with subsequent re-transmission handshake, introduces more than a round trip delay to transmission. For real-time applications, especially for VoIP, retransmitted packets might be entirely useless, so VoIP networks should leave the proper error control and error recovery scheme to higher communication layers. They can, thus, provide the level of reliability required, taking into account the impact of the delay characteristics. Although TCP/IP provides reliable connection, it is at the cost of packet delay or higher network latency. On the other hand, UDP is faster compared to TCP. Therefore, VoIP uses UDP as the transport level protocol. Reliability is built into higher layers. RTP over UDP/IP is usually used for voice and video communication [30].

2.4.1.5. Interoperability

In order to communicate using different products from different vendors, standards have to be confirmed. Currently the main standards are the H.323 and SIP. H.323 from ITU-T was the first set of agreed-upon standards, so it is quite popular. However, SIP from IEFT is becoming more acceptable. It is relatively lightweight and easily scalable, so almost all the vendors are developing products on it [30].

2.4.1.6. Integration with PSTN

In the world that PSTN is still the most used telephony, VoIP needs to be integrated with PSTN to be more accepted and deployed. This will make PSTN and IP telephony networks appear as a single network to the end users and has been achieved through the use of gateways between the Internet on one hand and PSTN on the other [30].

2.5. Voice over IP (VOIP)

In modern society, it is considered to be essential for any business to have Internet access and Intranet. More and more communications are in digital forms and transported via packet networks such as IP, ATM, and Frame Relay. Since data traffic is growing dramatically, there comes a lot of interest in transporting voice over the data networks. Transporting voice communications using Internet Protocol (IP) is usually called "Voice over IP" or VoIP, and has became very attractive by giving the low cost. Because the VoIP is used as a real time application to make the comparison between the UMTS an WiMAX QoS this chapter included an overview of VoIP application which is discussed in the following section [24].

2.5.1. An introduction of VoIP Systems

In the 1990s, a number of individuals in research environments, both in educational and corporate institutions, started researching the passing of voice over IP networks, especially corporate intranets and the Internet. This technology is commonly referred to today as VoIP. In simple terms, it digitizes the audio stream and breaks it into small chunks, then transmits those chunks over an IP network to the receiver. These chunks are collected and reassembled. This process converts a two people audio communication stream into a normal telephone call.

VoIP brings significant change in the way that people communicate. It gave users an option of using pure IP-based phones, including desktop computers, wireless phones and laptops, in addition to the telephones we have today. In Figure 2-5[32] shows how VoIP network can be connected and communicate with the normal PSTN network and Mobile network.

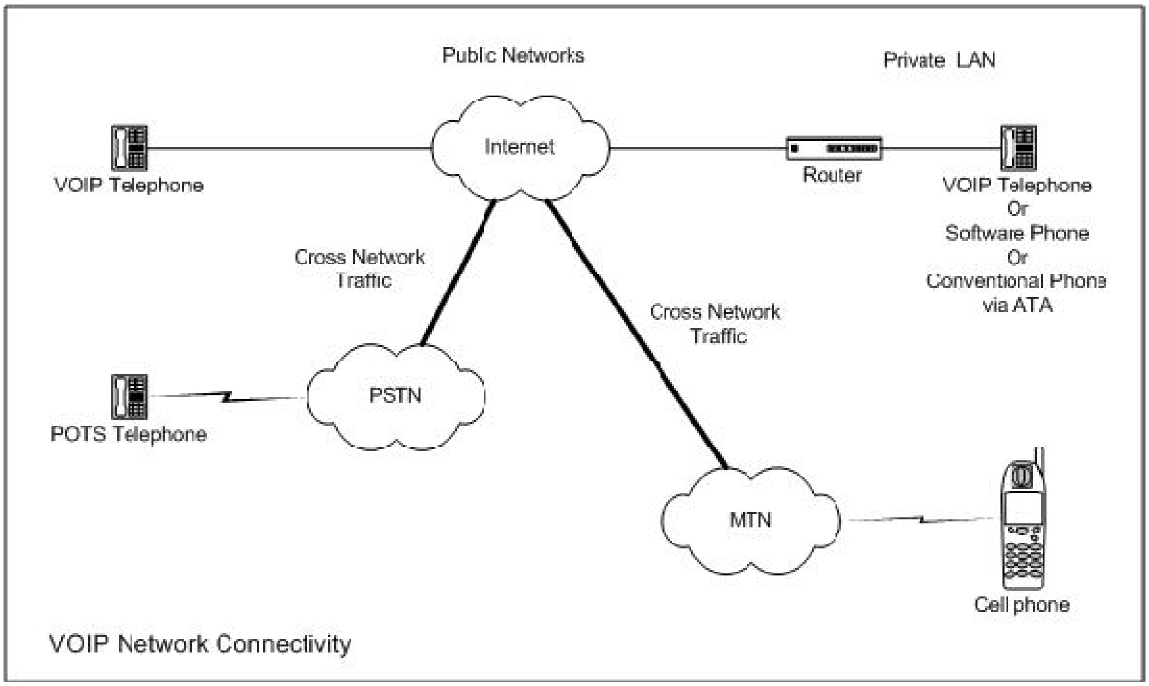


Figure 2-4: VoIP Network Connectivity [32]

2.5.2. Importance and benefits of VoIP

First of all, VoIP offers significant cost savings relative to the PSTN. Remote branches and users can use VoIP to bypass the long-distance carriers, which charge them per minute. What they need to pay is a monthly Internet access fee. Another aspect of VoIP is integration. VoIP gives us the ability to integrate a standalone telephone with the personal computer. People can use a computer for voice communications (soft phones). VoIP also allows something else: the ability to use a single high-speed Internet connection for both voice and data communications. This idea is commonly referred to as convergence and is one of the primary drivers for corporate interest in the technology. The benefit of convergence should be fairly obvious: by using a single data network for all communications, it is possible to reduce the overall maintenance and deployment costs. In short, VoIP enables people to communicate in more ways and with more choices [24].

2.5.3. VoIP architectures

A whitepaper [32] suggested that to design and implement VoIP architectures, product developers are face challenges in 5 areas:

1. Voice quality should be comparable to what is available using the PSTN, even over networks having variable levels of QoS.

2. The underlying IP network must meet strict performance criteria including minimizing call refusals, network latency, packet loss, and disconnects. This is required even during congestion conditions or when multiple users must share network resources.

3. Call control (signaling) must make the telephone calling process transparent, so that the callers need not know what technology is actually implementing the service.

4. PSTN/VoIP service inter-networking (and equipment interoperability) involves gateways between the voice and data network environments.

5. System management, security, addressing (directories, dial plans) and accounting must be provided, preferably consolidated with the PSTN operation support systems. Among many industry solutions, AT&T's Common VoIP Architecture and Cisco's AVVID are becoming outstanding. Cisco's AVVID (Architecture for Voice, Video and Integrated Data) is a well-known architecture and worthy to discuss. Cisco presents a typical IP telephony solution employing the Cisco AVVID network infrastructure [13]. The Cisco AVVID IP Telephony solution is the leading converged network telephony solution for organizations that want to increase productivity and reduce costs associated with managing and maintaining separate voice and data networks. The flexibility and sophisticated functionality of the Cisco AVVID Network

Infrastructure provides the framework that permits rapid deployment of emerging applications, such as desktop IP telephony, unified messaging, desktop collaboration, enterprise application integration with IP phone displays, and collaborative IP contact centres. These applications enhance productivity and increase enterprise revenues. Figure 2-6 [13]illustrate a very typical VoIP solution.

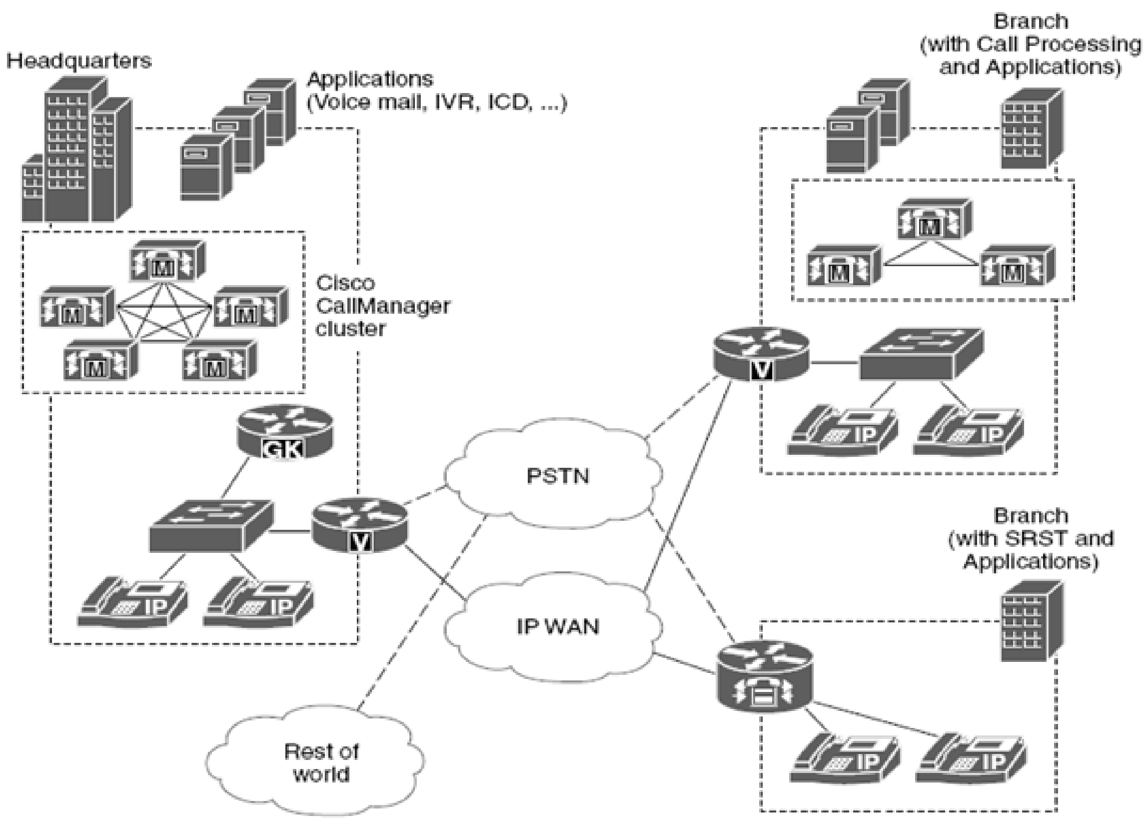


Figure 2-5: Typical IP Telephony Solution [13]

2.5.4. VoIP standards and protocols

The most popular and accepted VoIP standards are H.323 from ITU-T and SIP from IETF.

2.5.4.1. H.323 standard

H.323 is an ITU recommendation for multimedia communications over connectionless networks that do not guarantee Quality of Service (QoS), such as IP networks. The standard covers point-to-point communications and multipoint conferences. It addresses call control, multimedia management, bandwidth management, and interfaces between LANs and other networks. The elements of H.323 architecture are User Terminals, GateWays (GWs), Multipoint Control Units (MCUs,) and GateKeeper (GKs).

User terminals are normally IP phones or soft phones. They provide real-time two-way communications. Gateways are used for integration with the PSTN network. They perform the translation of the signaling and media streaming exchanged between VoIP and PSTN networks. Multipoint control units are used for conferencing. All terminals participating in the conference establish a connection with the MCU. Gatekeepers are responsible for all authorization, address resolution and bandwidth management. Terminals, gateways and MCUs are generally called “Endpoints”.

There are many protocols involved in a H.323 system. Figure 2-7 [33] shows the H.323 protocols in relation to the OSI model.

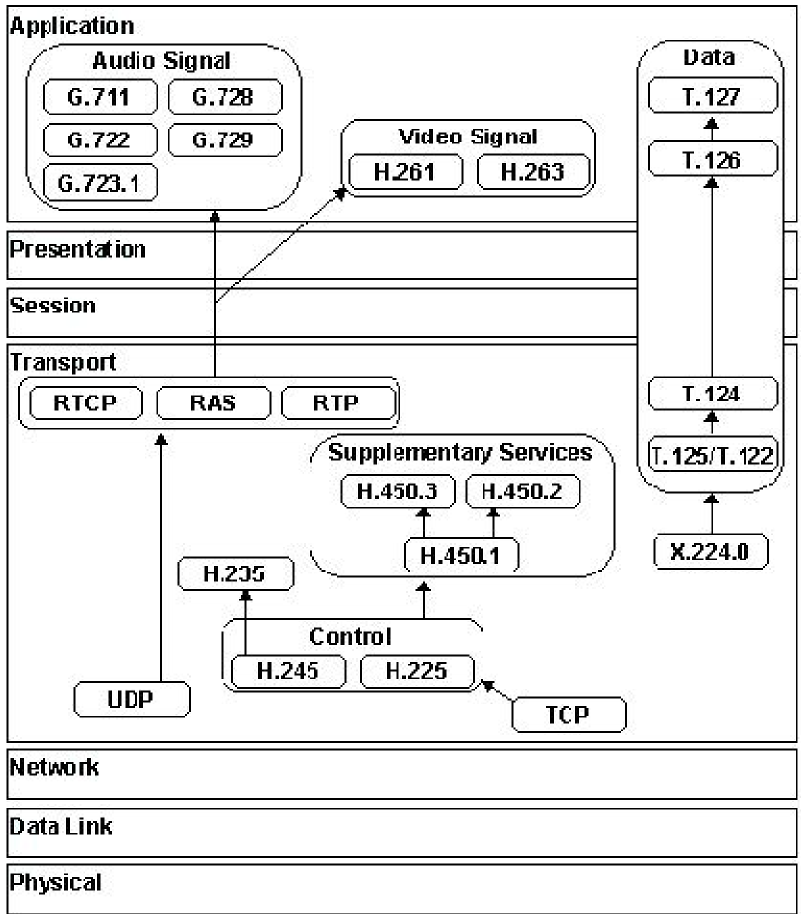


Figure 2-6: The H.323 suite in relation to the OSI model [33]

The Registration Admission Status (RAS) protocol is the key protocol for GKs. When an endpoint joins the network, it sends the GK a RAS registration request (RRQ), which contains information about itself such as the endpoint address and user alias. If the GK accepts the registration, it sends a Registration Confirm message (RFC). Otherwise, it sends a Registration Reject message (RRJ). RAS messages are carried in UDP packets. In an H.323 call setup life cycle, four key protocols are used: RAS, H.225, H.245 and RTP/RTCP. When an H.323 endpoint wants to make a call, it asks the GK for permission by sending a RAS Admission Request (ARQ) message, which contains information of the destination endpoint. The GK may reject the request by sending back an Admission Reject (ARJ) message with a variety of reasons such as "not enough bandwidth” or “cannot find destination”. More commonly, the GK will grant permission for the call. The GK resolves the address (either locally, by consulting another GK, or by querying some other network service) and then sends back an Admission Confirm (ACF) message containing the actual destination address, alias, etc. Once the address of the remote endpoint is resolved, the endpoint will use H.225 Call Signaling, in order to establish communication with the remote endpoint. Many H.225 messages are used to establish the call, including Setup and Setup acknowledge, Call Proceeding, Connect, Alerting, Information, Release Complete, Facility, Progress, Status and Status Inquiry and Notify. Endpoints must notify their gatekeeper that they are in a call [33].

Once a call has concluded, a device will send a Release Complete message. Endpoints are then required to notify their gatekeeper that the call has ended. As soon as the call has initialized the two endpoints start using H.245 call control protocol. H.245 provides capabilities such as capability negotiation, master/slave determination, flow control, etc. The two endpoints agree on the nature of the information that will be exchanged through the media channel and its format (compression, encryption, etc.). After these procedures, the Real Time Protocol/Real Time Control Protocol (RTP/RTCP) starts to transfer the media data according to the endpoints’ capabilities. The actual media communication starts. Figure 2-8 [34] shows the whole process of establishing and releasing an H.323 voice call. Some particular types of message are not discussed here.

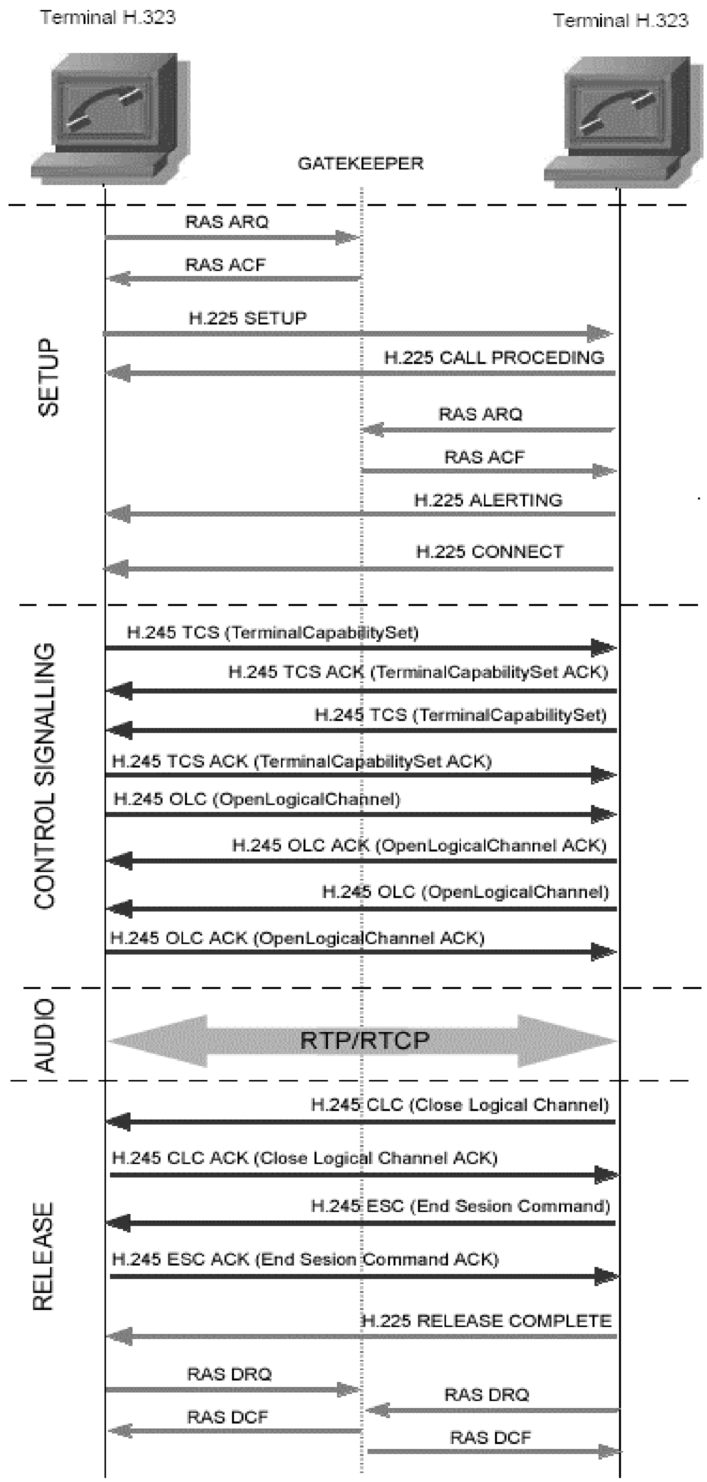


Figure 2-7: A typical H.323 call setup life cycle [34]

2.5.4.2. Session Initiated Protocol (SIP) standard

The Session Initiation Protocol (SIP) is an Internet Engineering Task Force (IETF) standard protocol. SIP is a signaling protocol that can set up and tear down media communications such as voice/video calls over Internet. Although SIP was introduced later than H.323, it is playing a major role in VoIP.

The SIP architecture identifies two basic components: SIP users, normally called SIP User Agents (UA), and SIP Servers. A user agent can be a SIP enabled IP phone or just a soft phone. It can function in one of the following roles:

User-agent client (UAC): A client application that initiates the SIP call request.

User-agent server (UAS): A server application that contacts the user when a SIP request is received and that returns a response on behalf of the user.

A SIP endpoint normally can function as both a UAC and a UAS, but it only can function as one in a conversation. The endpoint that initiated the SIP call request becomes the UAC and the remote endpoint becomes the UAS.

In SIP architecture there are three different server groups:

SIP Registrar Server: A server receives and processes registration message from UACs regarding their current user location. Registrar servers are often co-located with a redirect or proxy server.

SIP Proxy Server: A server forwards the SIP messages to multiple proxy servers, in order for the SIP messages to reach their destinations. Proxy servers can provide functions such as authentication, network access control, routing, etc.

Redirect Server: A server helps endpoints to find the desired address by redirecting them to another server.

SIP uses Session Description Protocol (SDP) to describe the sessions that will be setup. A SIP INVITE message includes a SDP message as a payload, describing the capabilities of calling agent and then both parties negotiate on the capabilities of the session will be set up. Figure 2-9 [13] indicates the protocols involved in a SIP voice call and their relationship.

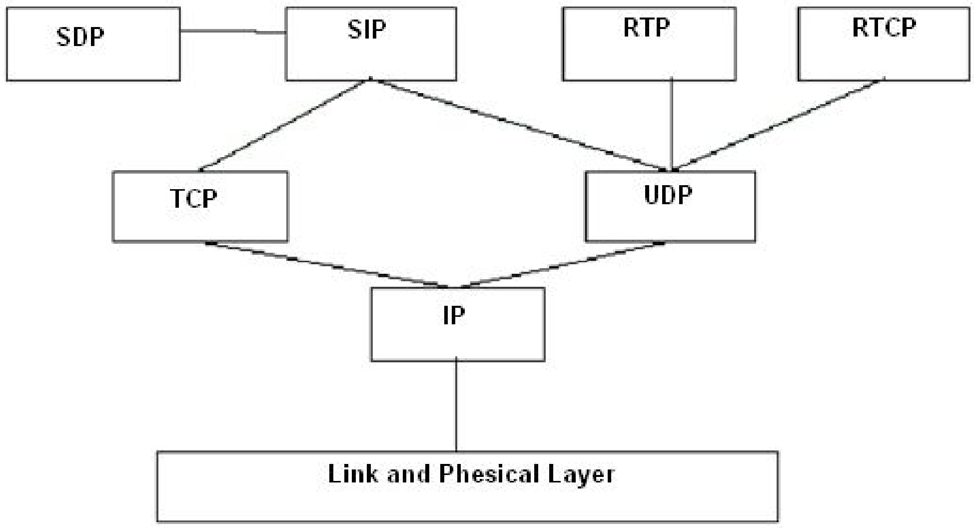


Figure 2-9: SIP protocol stack [13]

The procedures of establishing a SIP call under different circumstances are slightly different. In some scenarios, UA needs to contact with Servers to be able to find out and connect with remote endpoint. Figure 2-11 [35] shows a typical SIP call process without involve of any servers.

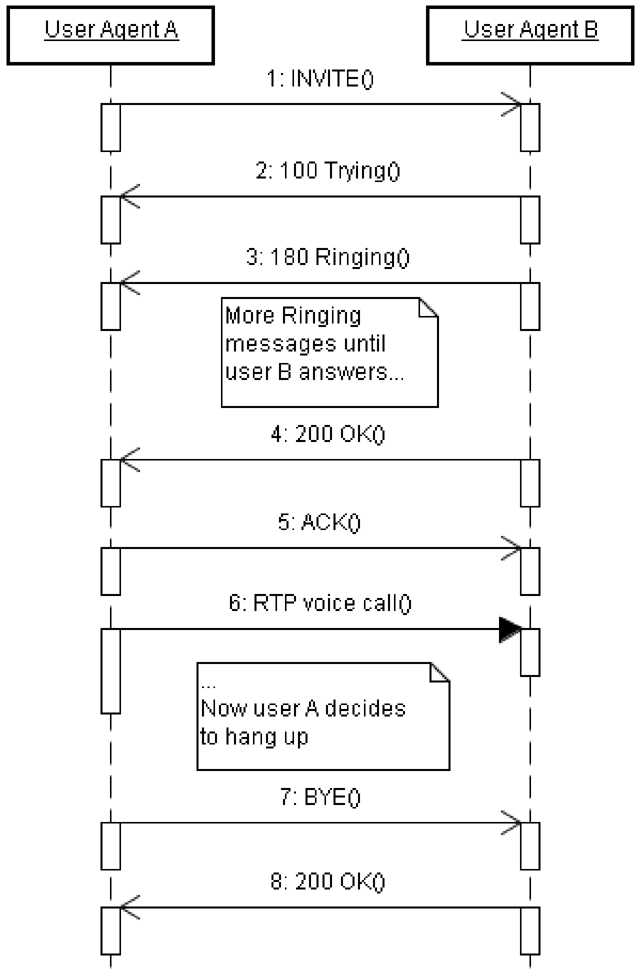


Figure 2-10: A typical SIP call [35]

User Agent A (UA A) sends a SIP request "INVITE" to User Agent B (UA B) to indicate its wish to talk to UA B. This request contains the details of the voice streaming protocol. The Session Description Protocol (SDP) is used in the payload for this purpose. The SDP message contains a list of all media CODECs supported by UA A. After UA B received the request, it confirms the receiving with UA A. While the phone rings, UA B sends provisional messages (ringing) to UA A. When UA B accepts the call, it sends an OK response to UA A. In the payload of the response, there's another SDP message. It contains a set of media CODECs that are supported by both user agents. At this point both parties are officially in the call. All types of SIP requests are accepted using

200-type responses. UA A finally confirms with an ACK message. Both user agents are now connected using the method selected in the last SDP message. At the end of the communication session, one of the users hangs up. At this point this UA A sends a new request, BYE. The other user's user agent accepts the request and replies with an OK message. The call is disconnected.

2.5.4.3. RTP and RTCP

In both SIP and H.323 standards, Real-time Transport Protocol (RTP) and Real-time Transport Control Protocol (RTCP) are used to transport the voice over the Internet. RTP/RTCP were designed for providing end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or uncast network services [36]. RTP is the one to carry the real-time data, while RTCP is to monitor the transmission QoS and generate the statistic information for the participants in the session.

As called Real-time transport protocol, RTP itself even does not provide guaranteed timely delivery. Also, it does not have any mechanism to ensure the QoS, such as in order delivery and packet re-delivery. Despite this, RTP, along with RTCP, is one of the foundational VoIP protocols.

RTP is build on top of the User Datagram Protocol (UDP), which is running in transport layer and also only provides unreliable best-effort data transmission. Every RTP packet has a fix-long header which contains the packet information. Figure 2-12 shows the structure of a RTP packet. As we can see in the figure, the header includes [36]:

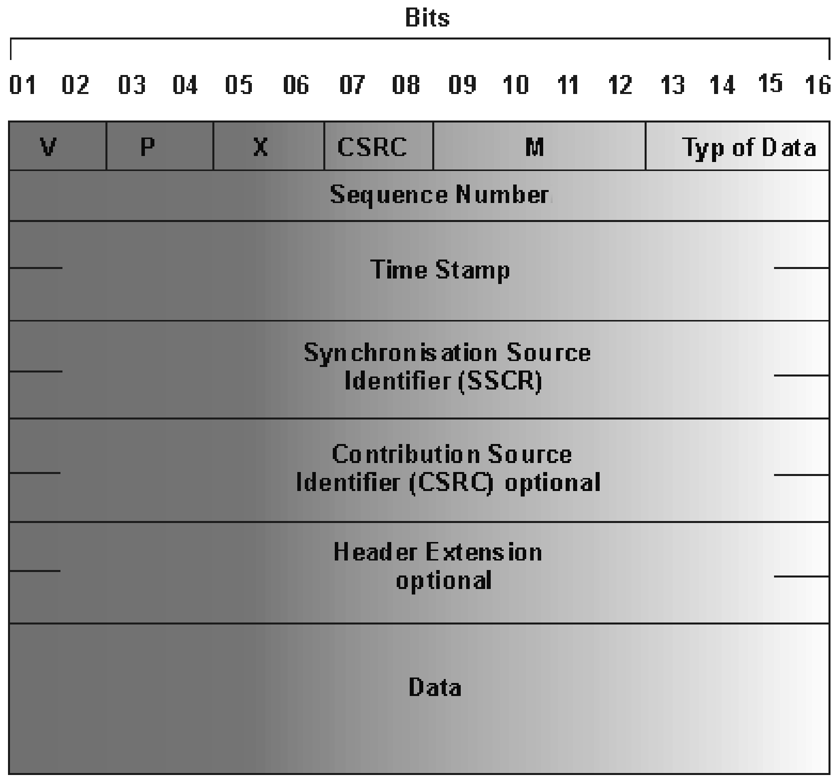


Figure 2-11: RTP packet structure [36]

• Version (V): 2 bits, identifies the version of RTP.

• Padding (P): 1 bit, used to identify if there are any extra padding octets at the end RTP packets.

• Extension (X): 1 bit, identifies if a header extension is used after the fixed header.

• CSRC count (CC): 4 bits, contains the number of CSRC identifiers that follow the fixed header.

• Marker (M): 1 bit, defined by a profile. It is intended to allow significant events such as frame boundaries to be marked in the packet stream

• Payload type (PT): 7 bits, indicates the format of the payload and determines its interpretation by the application.

• Sequence number: 16 bits, increments by one for each RTP data packet sent, and may be used by the receiver to detect packet loss and to restore packet sequence. The initial value of the sequence number is random.

• Timestamp: 32 bits, reflects the sampling instant of the first octet in the RTP data packet.
• SSRC: 32 bits, indicates the synchronization source.
• CSRC list: 0 to 15 items, 32 bits each. The CSRC list identifies the

RTP uses a fixed packet header, while RTCP has several different packet types. These different packets carry variety of control information [37]:
SR: Sender report, for transmission and reception statistics from participants that are active senders
RR: Receiver report, for reception statistics from participants that are not active senders and in combination with SR for active senders reporting on more than 31 sources
SDES: Source description items
BYE: Indicates end of participation
APP: Application-specific functions
RTCP is based on periodic transmission of control packets to all the participants of a particular session. All participants in the session send RTCP packets. The control packets are distributed in the same way as the data packets. Each RTCP packet includes, a sender and/or receiver reports that report statistics, such as number of contributing sources for the payload contained in this packet. The number of identifiers is given by the CC field.
• Extension header: Indicates the length of the extension in 32bit units, excluding the 32bits of the extension header.

2.5.4.4. VoIP CODECs

When making a call over the Internet, the software (soft-phone) or hardware needs to use a codec to send/receive information in a certain format and convert it to the sounds end user can hear. Generally, a codec with higher bandwidth requirements provides better voice quality (If the internet connection is fast enough to support the codec). Most VoIP providers/hardware/licensed software will support G.711 and G.729. The G.711 codec requires a connection almost 3 times faster than that required by the G.729 codec. In the case of the free soft-phone, then G.729 may not be available; however, the GSM codec should be, and will give similar call quality to that of a mobile phone.

Chapter Three

Quality of Service Parameters

The research objective is to implement a VoIP application in a UMTS and WiMAX networks by simulation scenarios and then to compare the simulation results for each network model to analyze the VoIP QoS in each network case. This chapter includes a theoretical discussion of VoIP QoS parameters that used in this research.

3.1. Quality of Service QoS

Quality of service (QoS) is the overall performance of a telephony or computer network, particularly the performance seen by the users of the network. To quantitatively measure quality of service, several related aspects of the network service are often considered, such as error rates, bandwidth, throughput, transmission delay, availability, jitter, etc.

Quality of service is particularly important for the transport of traffic with special requirements. In particular, much technology has been developed to allow computer networks to become as useful as telephone networks for audio conversations, as well as supporting new applications with even stricter service demands.

In PSTN network, quality of service for every phone call is guaranteed by the constant available bandwidth. While packet networks work the opposite way, when bandwidth availability drops, data can still be transmitted through but in a slows transmission speed. It would be critical to the real-time applications. The idea of QoS is to meet the requirements of the real-time applications by applying some mechanism to control and provide better service to them [39]. It is commonly applied in the situations where VoIP is available.

In this research work the parameters used for QoS of VoIP are MOS, end-to end delay, jitter and packet loss.

3.1.1. Mean Opinion Score (MOS)

A common benchmark used to determine the quality of speech is the Mean Opinion Score (MOS). To get a MOS of a speech, a group of listeners rate the quality of sample speech on a scale of 1 to 5, with 5 being the best quality. The averaged score is taken as the MOS of that speech [40].

Each CODEC provides a certain quality of speech. Table 3-1 [41] lists the MOS for the common used CODECs.

Table 3-1: MOS for common CODECs

CODEC	G.711	G.723.1	G.726	G.728	G.729
MOS	4.5	3.6	4.2	4.2	4.2

MOS provides a numerical measure of the quality of human speech in voice telecommunications, with value ranging from 1 to 5 where 1 is the worst quality and 5 is the best quality. In this simulation work, MOS through a non-linear mapping has been computed from R-factor as in:

$$\mathrm{MOS} = 1 + 0.035\mathrm{R} + 7 * 10^{-6}\mathrm{R}(\mathrm{R} - 60)(100 - \mathrm{R}) \quad (3\text{-}1)$$

Where:

$$R = 100 - I_s - I_e - I_d + A \qquad (3 - 2)$$

I_s: is the effect of impairments that occur with the voice signal;

I_e: is the impairments caused by different types of losses occurred due to codecs and network, and

I_d : represents the impairment caused by delay particularly mouth-to ear delay.

Using the default setting for I_s and A Eqn (3-1) can be reduced to:

$$R = 94.2 I_e I_d$$

3.1.2. Delay

The total end-to-end delay is the sum of a packet assembly at the source, a network delay and receiver delay.

ITU G.114 [42] recommends:

• 0-150 ms, acceptable for most applications.

• 150-400 ms, acceptable but has impact.

• Above 400 ms, unacceptable.

While a more common and used limit for delay in VoIP is that the acceptable delay should be less than 200 milliseconds [42].

Delay in transporting a voice packet over the IP network causes two main problems: Echo and Talker Overlap. When the round trip delay through the network becomes greater than 50 ms, echo becomes a problem. Echo cancellers are used to avoid the problem. Since the IP networks normally have higher end-to-end delay, echo cancellation becomes an essential requirement for VoIP. In addition, with the increasing of delay, the Talker Overlap becomes more serious and it can be extremely annoying. When the delay gets more than 250 ms, the connection sounds like half duplex and cannot be claimed as an interactive session [42].

3.1.2.1. Packet end-to-end delay

Delay in VoIP networks is caused by propagation delay and processing delay. Propagation delay is a characteristic of the transmission medium which may be fibre optic, copper medium or radio frequency. Processing delay is caused by the network devices that handle VoIP traffic. Processing delay may be affected by the network device capacity and also the current network traffic at each device along a transmission path [53].

End to-end delay means the time required for a packet to be traversed from source to destination in the network and is measured in seconds. Generally, in VoIP network there are three types of delays occurring during the packet transverse. They are: sender delays when packets are transverse from source node, network delay and receiver delay [49].

Packet end-to-end delay is the total voice packet delay and is calculated as

$$D_{e2e} = D_n + D_e + D_d + D_c + D_{de} \qquad (3-3)$$

WhereD_n, $D_{e,}$ D_d, $D_{c,}$ and D_{de}represent the network, encoding, decoding, compression and decompression delay, respectively [42].

3.1.2.2. Packet delay variation (PDV)

PDV in OPNET is defined as the variance of the packet delay, which is computed as follows:

$$PDV = \frac{\sum_{i=1}^{n}([t'(n) - t(n)] - u)^2}{n} \qquad (3-4)$$

Where u is the average delay of the n selected packets [42].

3.2.3. Jitter

Jitter is the variance of packet arrival time. It is defined to be the mean deviation of the packet spacing change between the sender and the receiver [43]. It is measured in milliseconds. In an IP network, it is not guaranteed that the packets arrive at the receiver with the same and equal intervals as they are sent at the sender. That is where jitter comes from.

In order to get high quality voice calls, in VoIP applications, Jitter buffers are used. The incoming packets are held for a specified amount of time to allow the slowest packets to arrive before they are used to produce the voice stream. Jitter buffers introduce additional delay [43].

Let $t(i)$ and $t'(i)$ be the time transmitted at the transmitter and the time received at the receiver, respectively. Jitter is calculated as follows:

$$jitter = max_{i=1}^{n}([t'(n) - t'(n-1)] - [t(n) - t(n-1)]) \quad (3-5)$$

Where n is the number of transmitted packets.

3.2.4. Packet loss

Packet loss is a very important parameter of QoS. Generally speaking, a packet loss rate of 5% will annoy users. Packet loss usually occurs when there is congestion on the packets path, which causes the router buffers to overflow. The stability of the network heavily influences the packet loss rate [43].

A lot of research has been done in relation to the packet loss and VoIP quality. Different CODECs can be affected by packet loss differently. G.711's highest MOS score is about 4.4, while G.729A has only got highest MOS of 3.6 [44]. When packet loss occurs, G.711 can be less affected than G.729A, especially when the packet loss rate is more than 5%. Figure 3-1 [44] shows the different effect of packet loss for G.711 and G.729A. Also, different type of packet loss will introduce different results for VoIP quality [45]. In [46], the effects of random and burst packet loss have been investigated.

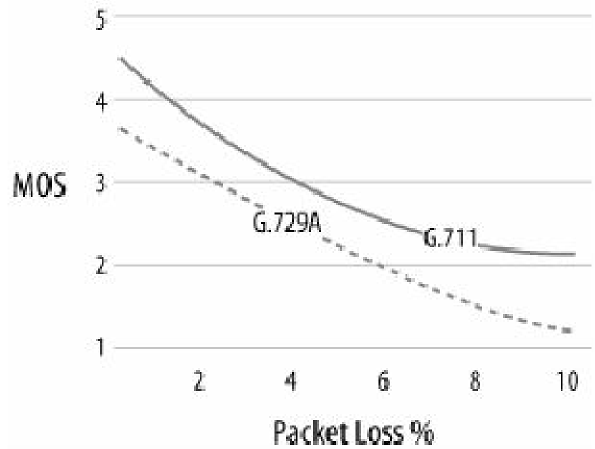


Figure 3-1:MOS rating with packet loss [44]

Chapter 4

Simulation Implementation

This chapter provides a detailed review of the research work carried out and identifies the results that are to be analyzed in the next chapter. The research work contains three important parts:

- VoIP over UMTS network Modeling.
- VoIP over WiMAX network Modeling.
- Simulation and analysis.

This chapter begins with introduction about the simulation as a method to study the QoS of the real time application including a brief overview about OPNET modeler as the simulation software used within this study. After that a UMTS network and implementation of the VoIP service across the UMTS network model has been developed and the same happened for the WiMAX network. Finally, different test scenarios will be discussed, and simulations will be done to generate statistical data from each of the scenarios.

4.1. Introduction to OPNET

OPNET stands for Optimized Network Engineering Tools, and was created by OPNET Technologies, Inc., which was founded in 1986. OPNET is a network simulation tool set; its products and solutions address the following aspects of communications networks [47]:

• Application performance management.

• Planning.

• Engineering.

• Operations.

• Research and development.

This tool set is powerful and can create and test large network environments via software. To address each of these aspects, OPNET provides corresponding product modules throughout its product line.

OPNET Modeler is the foremost commercial product that provides network modeling and simulation software solution among the OPNET product family. It is used widely by researchers, engineers, university students, and the US military. OPNET Modeler is a dynamic discrete event simulator with a user-friendly graphic user interface (GUI), supported by object-oriented and hierarchical modeling, debugging, and analysis. OPNET Modeler is a discrete event simulator that has evolved to support hybrid simulation, analytical simulation, and 32-bit and 64-bit fully parallel simulation, as well as providing many other features. It has grid computing support for distributed simulation. Its System in- the-Loop interface allows simulation with live systems which feed real-world data and information into the simulation environment. It provides an open interface for integrating external object files, libraries, and other simulators. It incorporates a broad suite of protocols and technologies, and includes a development environment to enable modeling of a very wide range of network types and technologies. With the ongoing release of updated versions, OPNET Modeler incorporates more and more features in order to keep up with the evolution of communication networks, devices, protocols, and applications. Hundreds of protocols and vendor device models with source code are already incorporated in the modeler. OPNET Modeler accelerates the research and development (R&D) process for analyzing and designing communication networks, devices, protocols, and applications [47]. OPNET Modeler GUI makes it user-friendly, and makes it easy for users to begin learning about it and working with it. However, when trying to progress beyond this initial phase, its full-featured functionalities and powerful programming interfaces make it difficult for people to grasp.

OPNET Modeler provides a comprehensive development environment with a full set of tools including model design, simulation, data collection, and data analysis and supporting the modeling of communication networks and distributed systems. OPNET Modeler can be used as a platform to develop models of a wide range of systems. These applications include: standard-based local area network (LAN) and wide area network (WAN) performance modeling, hierarchical internetwork planning, R&D of protocols and communication network architecture, mobile network, sensor network and satellite network. Other applications include resource sizing, outage and failure recovery, and so on.

OPNET Modeler is used in the UMTS and WiMAX networks QoS studies throughout this research.

4.2. Implementation on Opnet

The Opnet Modeler simulation environment utilizes two mechanisms to manage simulations called Project and Scenario. Projects are always the top level in any Opnet Modeler simulation. Every project contains one or more scenarios, which represent particular network configurations to be simulated. Generally, a scenario is a network configuration including the topology, protocols, applications, services and traffic flow.

A project may contain more than one scenario. Each scenario uses a network that is slightly different or the same with varied parameters. In this section, a UMTS network was created with two VoIP end-points. This scenario was duplicated with changes to the network and parameters for each simulation. For some of the scenarios, more IP traffic was added to show varying levels of congestion. The use of scenarios has permitted results to be obtained from a number of similar networks with minor variations to the network or parameters and the results have been used in the analysis and comparison described later.

To evaluate the performance of WiMAX and UMTS for VoIP traffic, the WiMAX and UMTS simulation modules have been implemented in OPNET network simulator based on OPNET's discrete event simulation model library. To model the UMTS and WiMAX networks, a new project and scenario was created using the Opnet Modeler "start-up wizard". As shown in Figure 4-1, the project was named VoIPUMTS and the default scenario name was used. After the names have been entered, the start-up wizard starts.

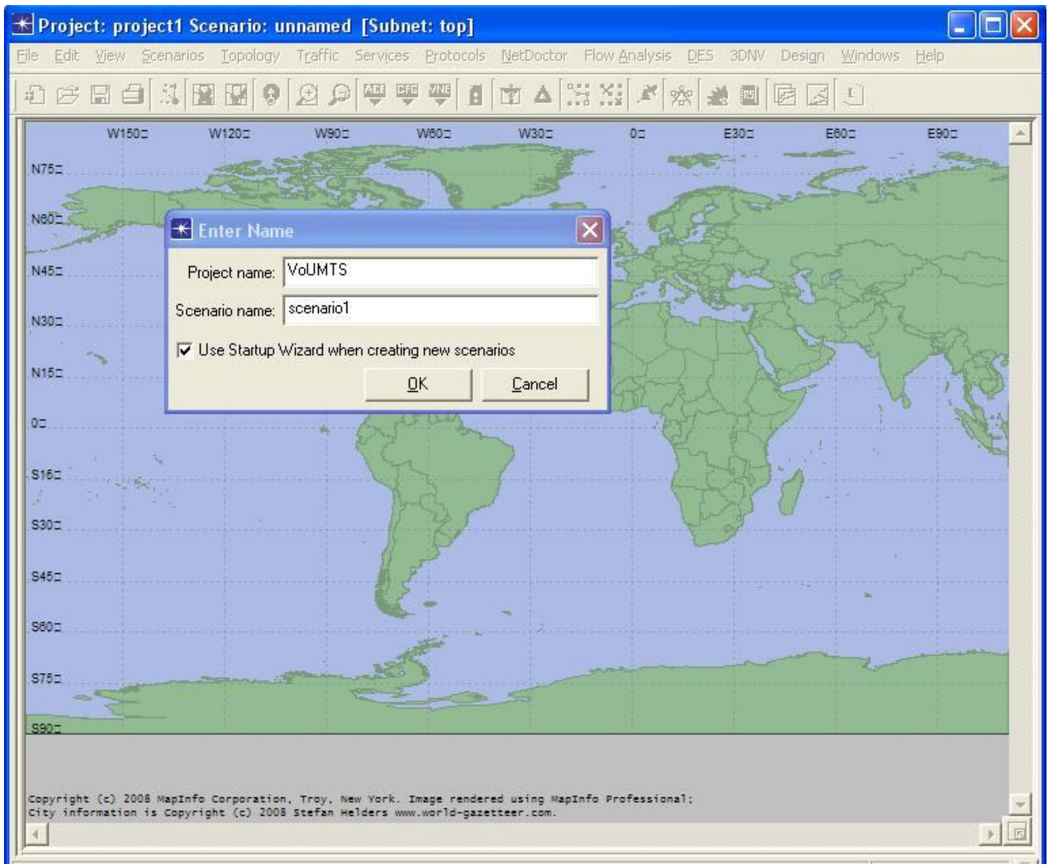


Figure 4-1: Starting a New Project – UMTS Project and Scenario names

The first step of the start-up wizard is to choose the initial network topology. The wizard gives some options to create the initial scenario as shown in Figure 4-2. The topology can either be imported from some specially formatted files or created from scratch manually. In this research, there is no pre-made network topology available that can be used, so the network topology was created from scratch manually. In Figure 4-2, "Create empty scenario" has been chosen.

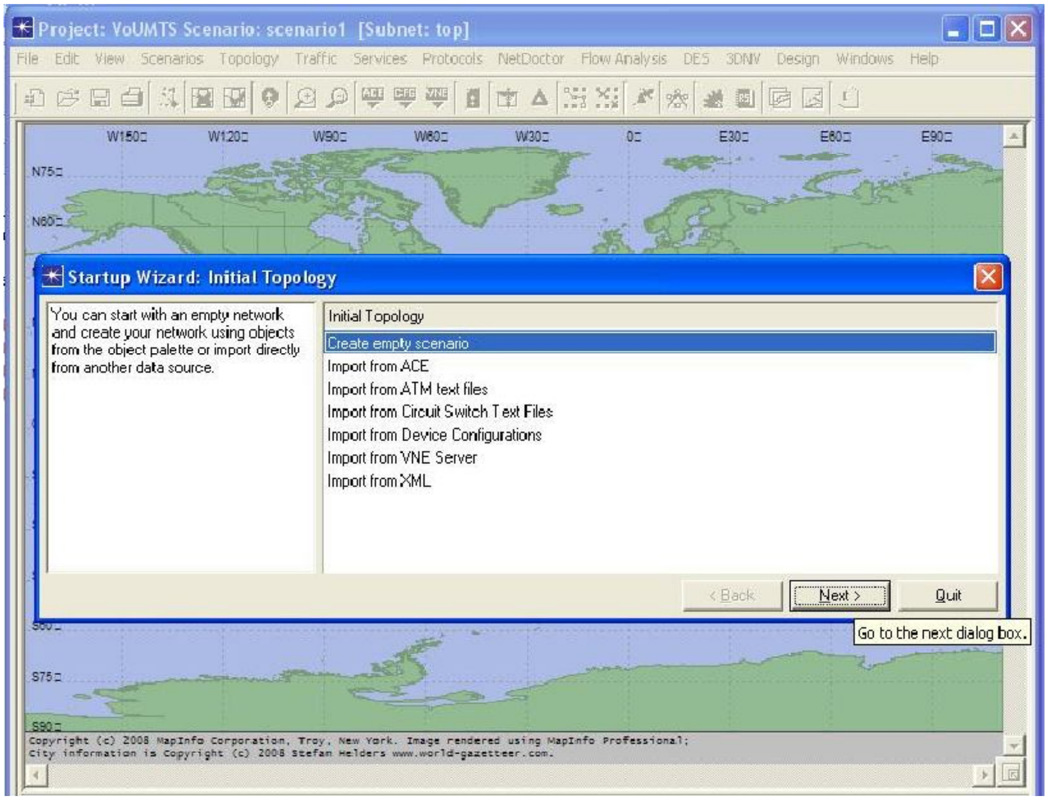


Figure 4-2: Start-up Wizard – UMTS Initial Topology

The next step is to choose the network scale and size. In this research project, a basic UMTS network has to be simulated. It includes all the UMTS elements such as base stations, RNCs, SGSNs and GGSNs. To include all this equipment, "Campus" has been selected as the network scale. The size of the campus network is specified as shown in Figure 4-3.

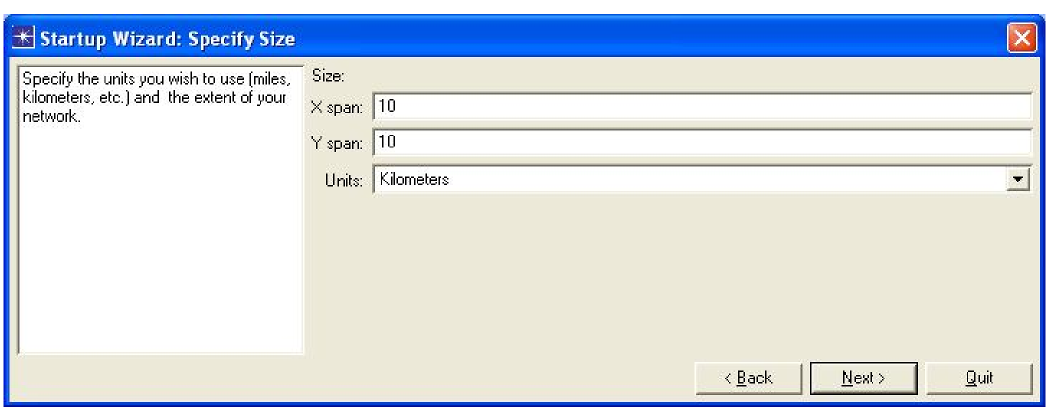


Figure 4-3: Startup Wizard - campus network size

After the network scale and size have been chosen, the next step is to select the technologies that will be used in the network. This selection will be used to create the Object Palette, which holds items that are used often in the project editor. All of the models that were included in the selected technologies will be put into the Object Palette, so they can be easily placed into the project. As this research project is to simulate VoIP over UMTS network and WiMAX network, UMTS and WiMAX technologies have been selected as shown in Figure 4-4.

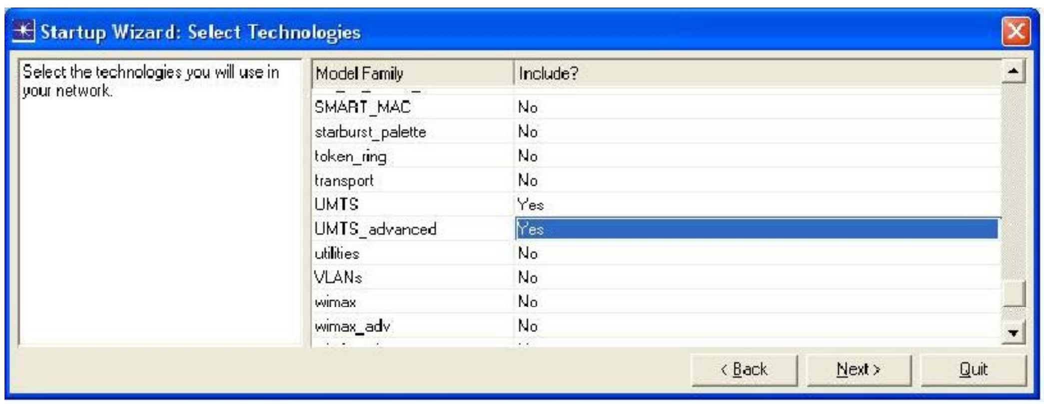


Figure 4-4: Start-up Wizard - Select Technology

Before the Start-up Wizard finishes, it gives a review of the scenario summary. Figure 4-5 shows the summary of the first scenario that is going to be created.

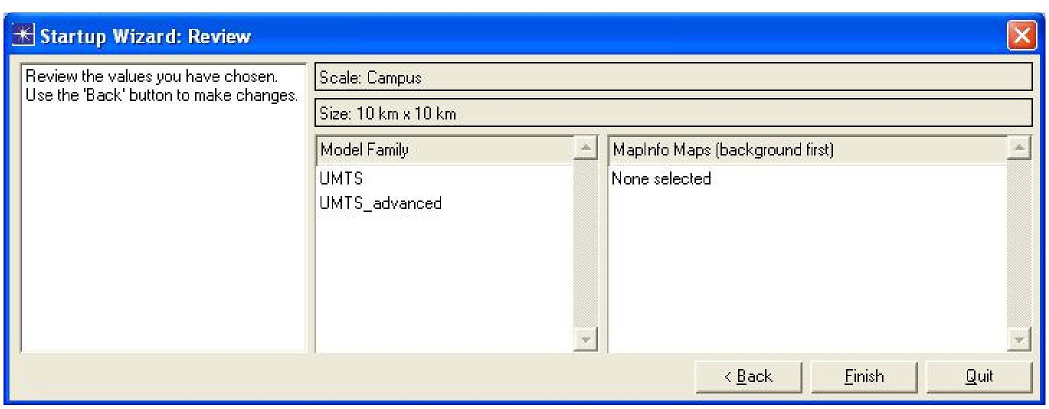


Figure 4-5: Start-up Wizard – Summarize

Finally, after finishing the Start-up Wizard, the Object Palette pops up with all the models that maybe needed to create the scenario network. Figure 4-6 shows the Object Palette created for one of the project scenarios. At this point, the network simulation environment has been setup and it is ready to be used for network Modeling and simulation.

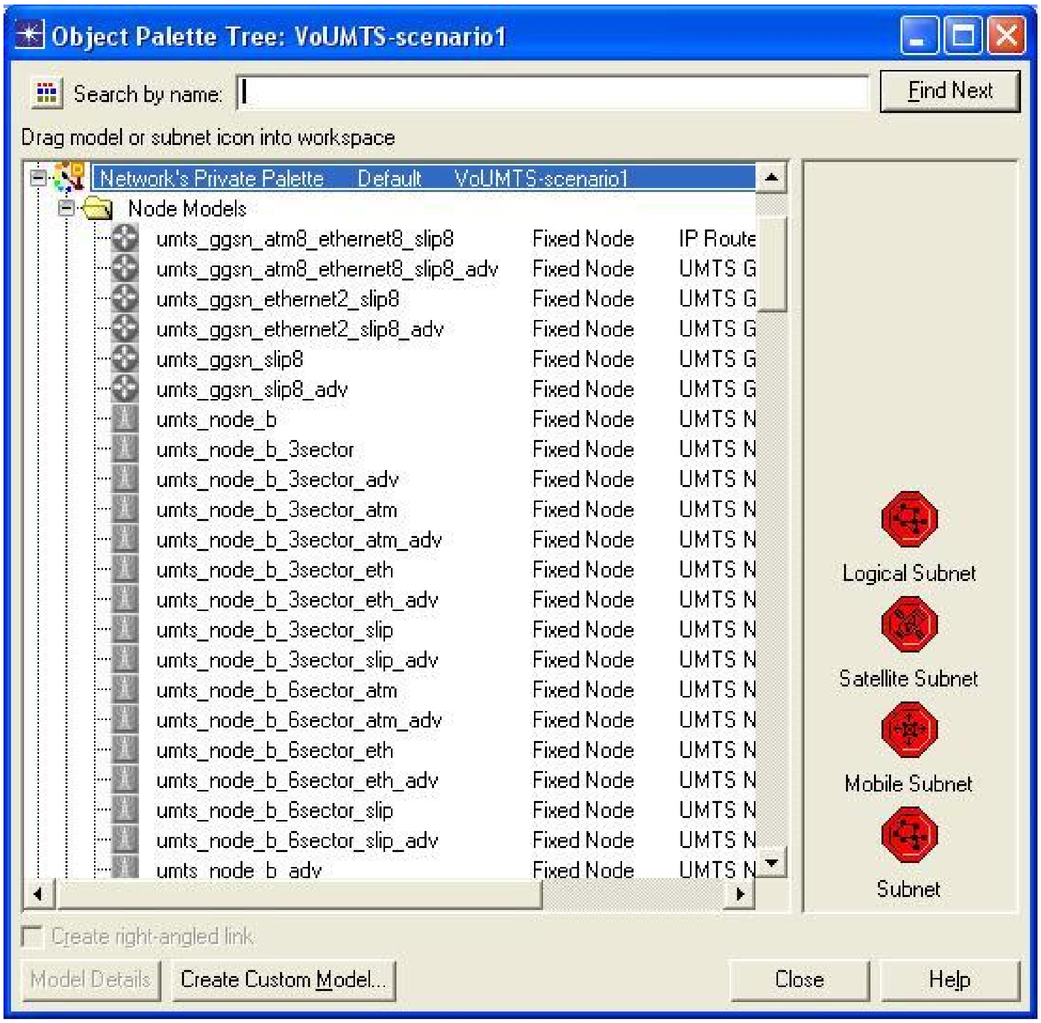


Figure 4-6: Start-up Wizard - Object Palette

4.3. UMTS network Modeling

In the next section the UMTS network implementation has been discussed.

4.3.1. Configure the Topology

In this research, a simple UMTS network is modeled with twelve end users connected to the UMTS network where two of these end users communicating using VoIP and the rest of end users using three other IP applications (web browsing, email exchanging, and file transfer) which represent the background traffic for the UMTS network.

The VoIP end users are connected into the UMTS network via different base stations. All the end users also have access to the internet, so they can access servers outside the UMTS network, such as Web server, FTP server and E-mail server. Figure 4-7 shows the sketch topology that this research used.

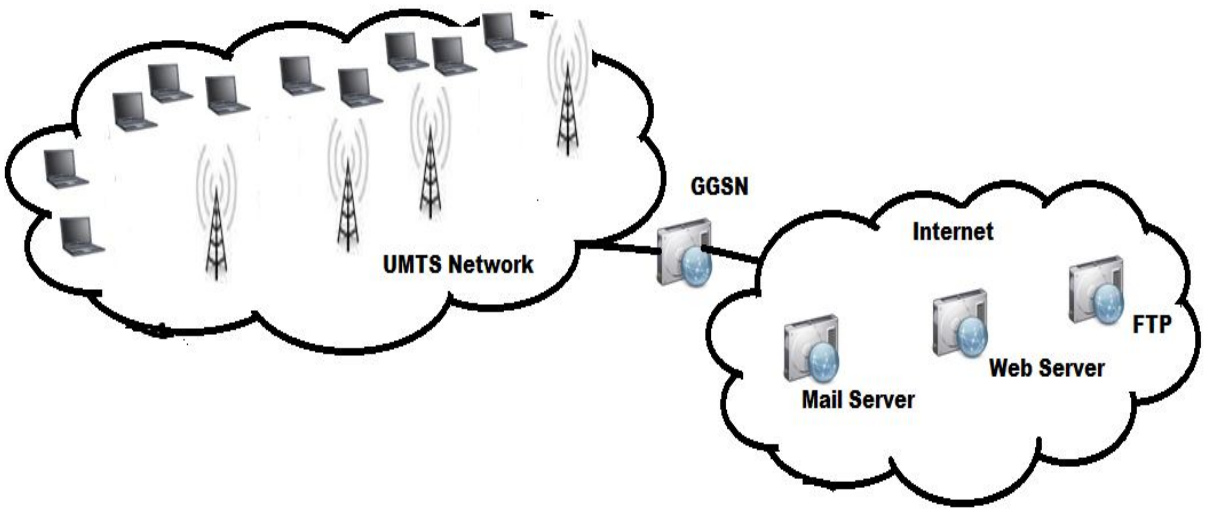


Figure 4-7: VoIP over UMTS sketch topology

As discussed in Chapter 2, a UMTS network consists of three domains: Core Network (CN), UMTS Terrestrial Radio Access Network (UTRAN) and User Equipment (UE). To model a UMTS network, all the UMTS essential models should be added into the scenario and connected properly.

UMTS networks contain both circuit switched and packet switched networks. In this research project, only a packet switched network is concerned, so there is no need to model the circuit switched elements like Mobile services Switching Centre (MSC), Visitor location register (VLR) and Gateway MSC. The two key packet switched elements are Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN). They are essential for the UMTS core network and must be modeled in this project. To model the UMTS core network, we start with the GGSN in the core network.

Figure 4-8: GGSN node model

After setup the GGSN, the SGSN models can be put into the project. The numbers of the SGSN models depend on the scale of the network. In this case, only two are required and they are connected to the IP backbone using PPP_DS3 links. Figure 4-9 shows the core network after SGSNs are added.

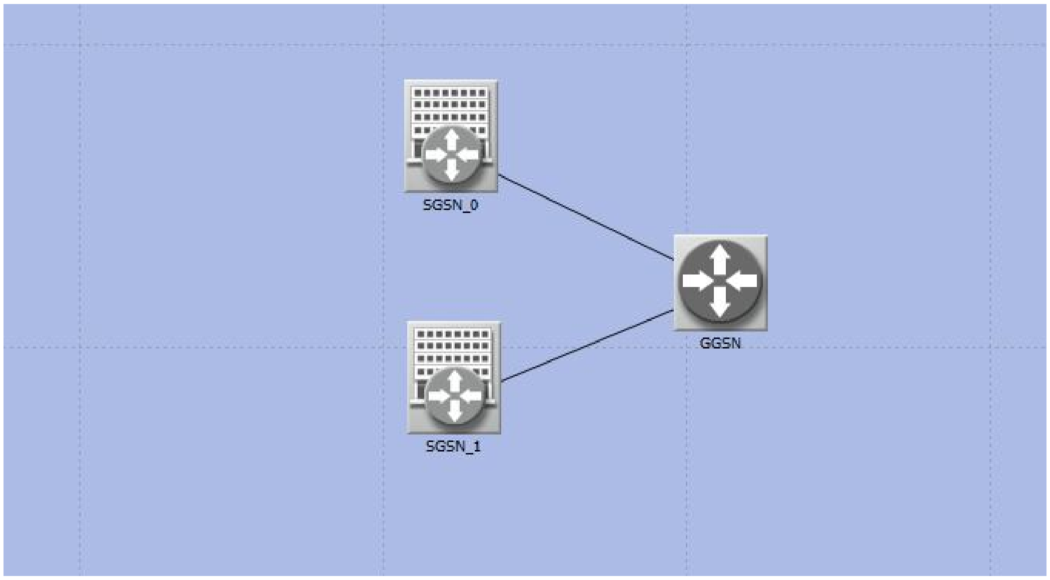


Figure 4-9: core network after SGSNs are added

The next step is to add the UMTS Terrestrial Radio Access Network (UTRAM) models into the network. It includes two different network models: the radio network controller (RNC) model and the Node B (also known as base station) model. One or more RNCs can be connected to a SGSN and one or more base stations can be connected to a RNC. ATM_OC3 links are used to connect the base stations to the RNCs and the RNCs to the SGSNs. In this research, a simple UMTS network with only four base stations is used. Figure 4-10 shows the topology and connections for this research.

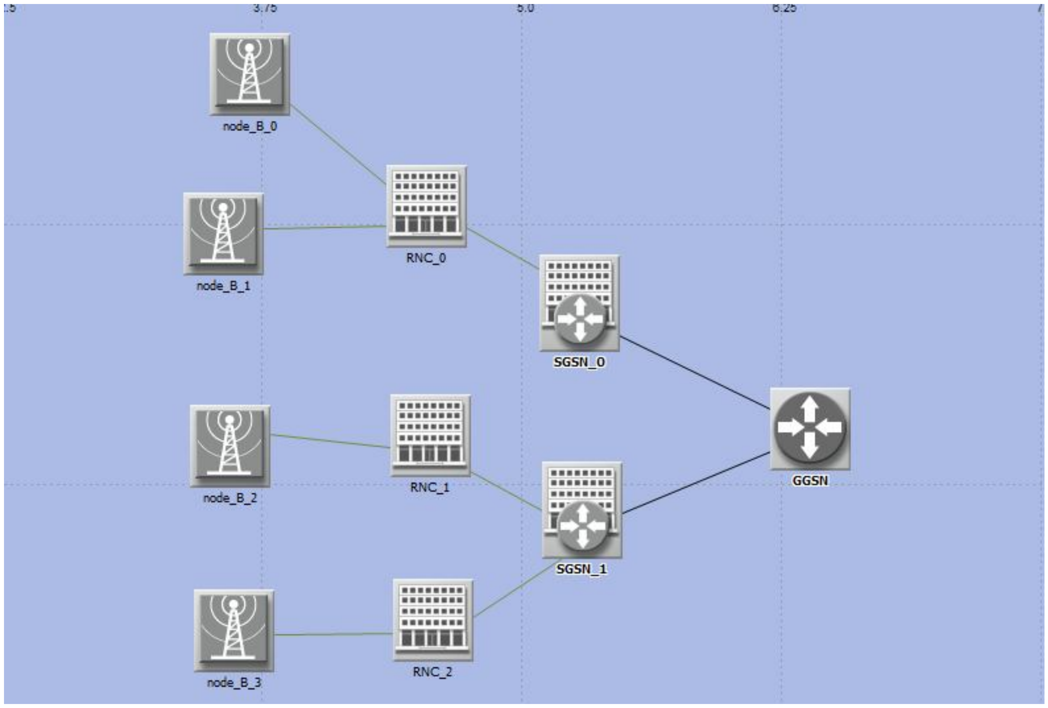


Figure 4-10: UMTS Modeling - UTRAM added

Opnet Modeler includes a UMTS workstation model, which can be used as the UMTS user equipment (UE). The UMTS workstation model can simulate a normal IP workstation. They can be used as VoIP end users and other IP service clients. In this research, twelve UMTS workstation models have been added (as shown in Figure 4-11) to simulate VoIP calls and other IP activities.

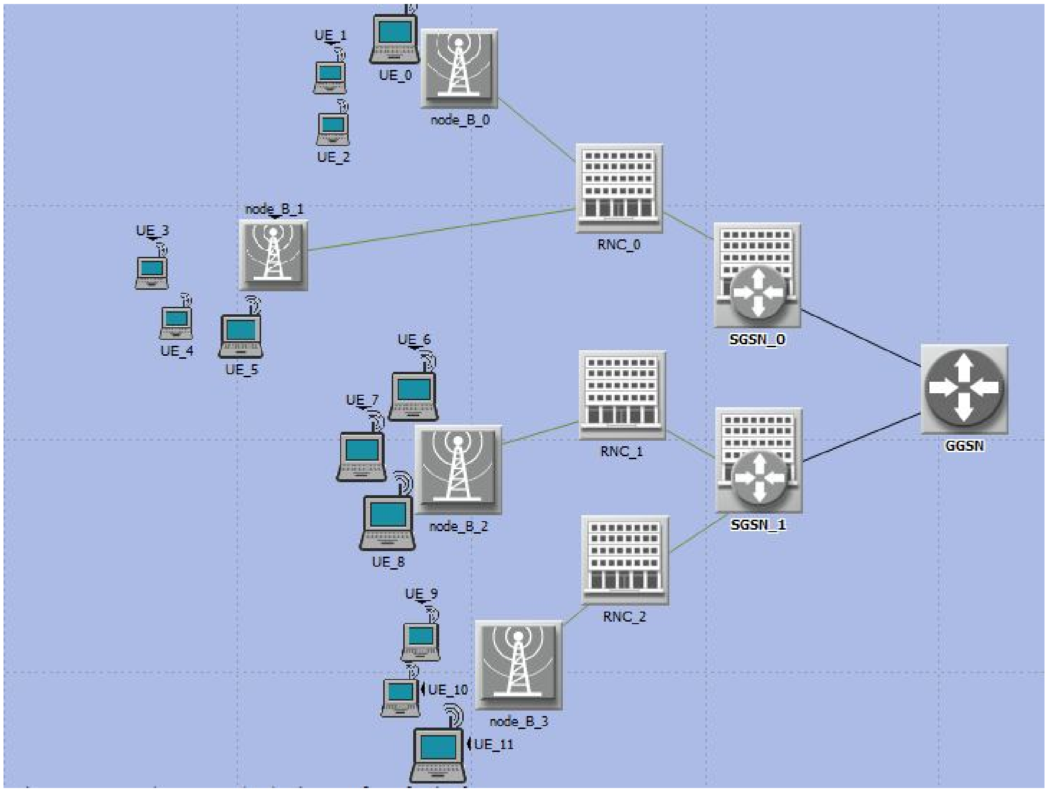


Figure 4-11:UMTS Modeling - End users added

To make the UMTS workstation models connect to the UMTS network in Opnet, the workstations have to be manually configured to use the right SGSN. In this instance, open the attributes dialog SGSN_1 by right clicking and selecting “Edit Attributes (Advanced)”, under “UMTS parameters”. There is a parameter called “SGSN ID” (shown in Figure 4-12). All the UMTS workstations that use the base stations connected to this SGSN have to set the same ID as their SGSN serving ID. After opening the UE_5’s attributes dialog, the “UE Serving SGSN ID” setting can be found under “UMTS” (shown in Figure 4-13). By default, the “UE Serving SGSN ID” is set to “0”. In this case, we change it to “1” to match the SGSN ID of SGSN_1. The same setting, also, has been done to other user equipments.

Here, a base UMTS network has been modeled.

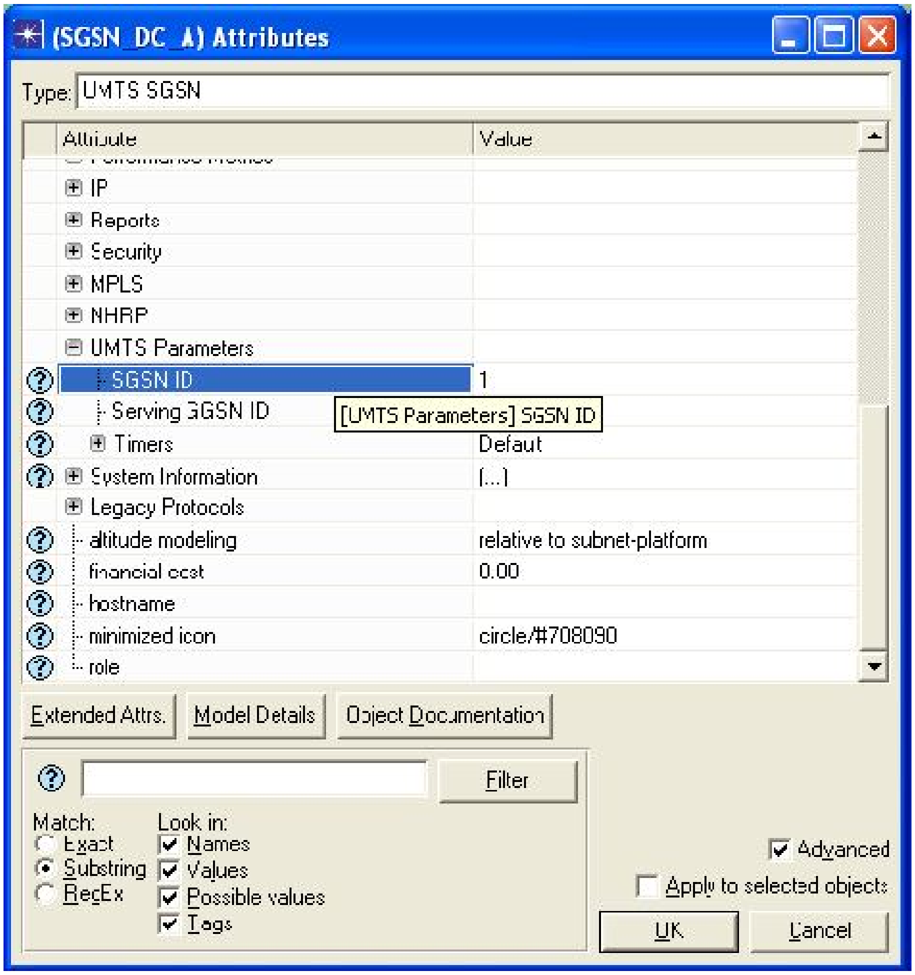


Figure 4-12: UMTS Modeling - UMTS parameters configuration for SGSN

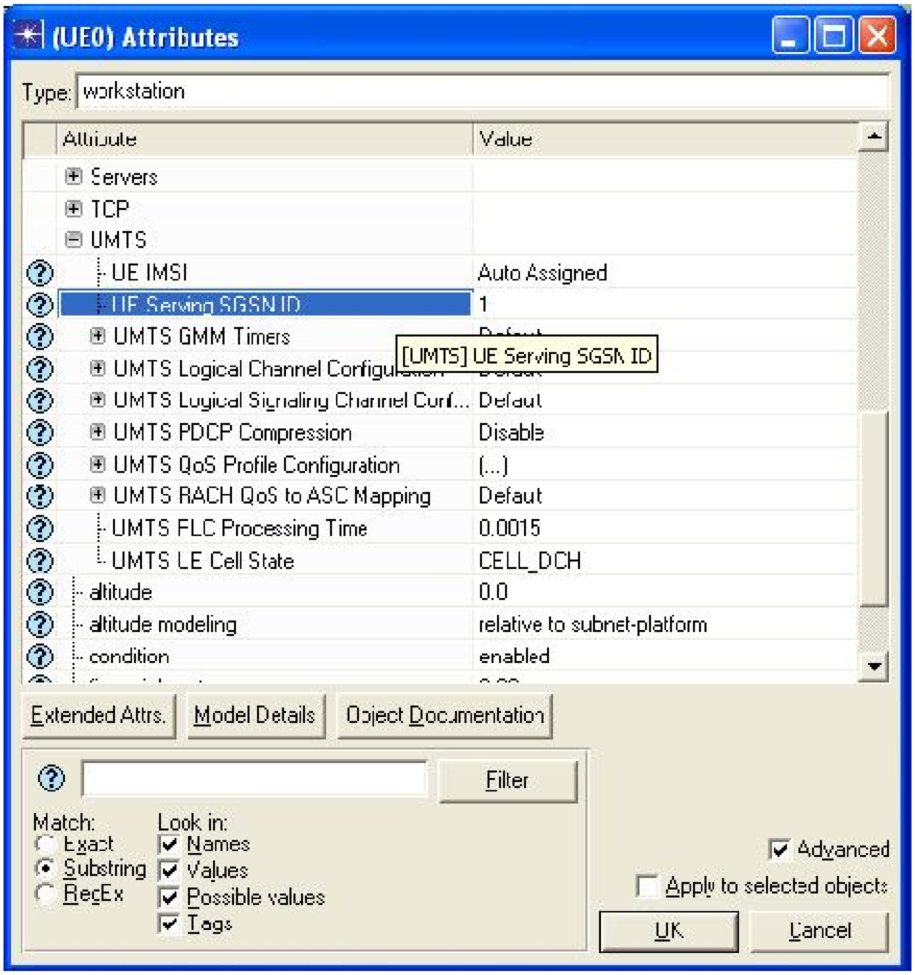


Figure 4-13: Hybrid UMTS Modeling - UMTS parameters configuration for UE

To add Internet access to the UMTS network, a IP Internet model has to be added in the UMTS core network. Opnet also have an IP Internet model IP32_Cloud to represent the Internet. PPP_DS3 links are used to connect the GGSN to the UMTS IP backbone and the Internet (as shown in Figure 4-14).

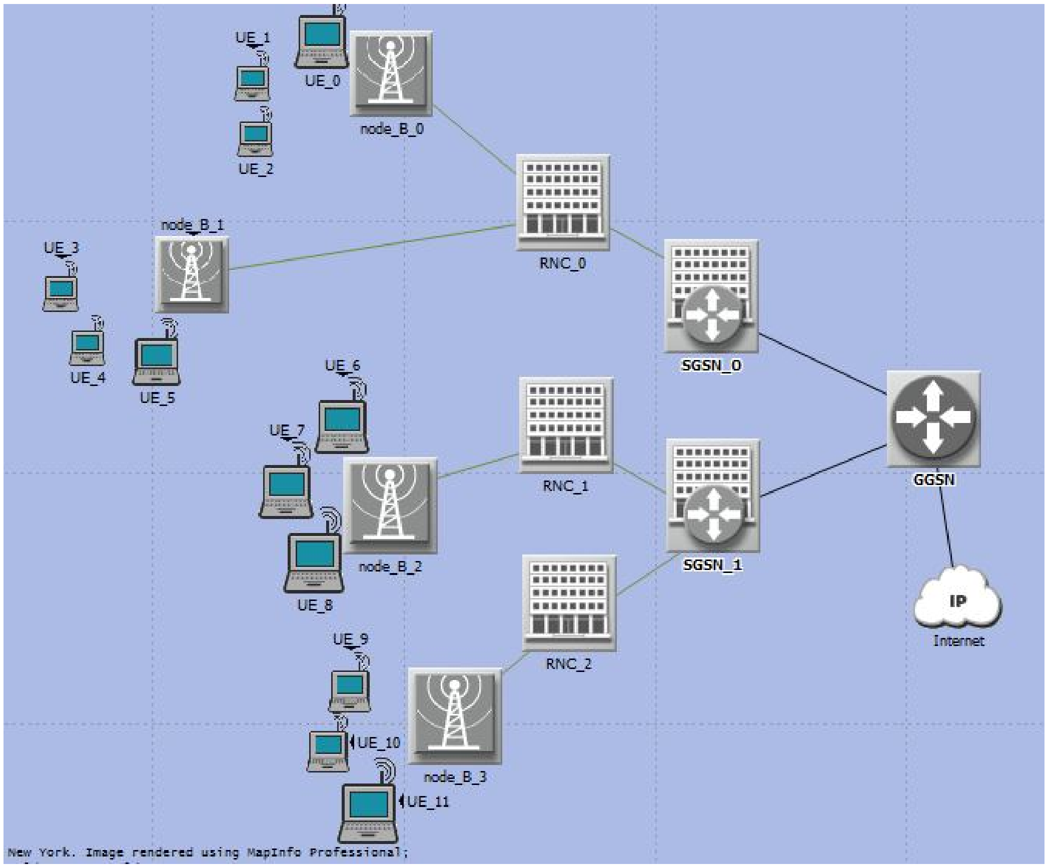


Figure 4-14: UMTS Modeling - connecting to Internet

In this research scenario, the UMTS end users should be able to access some servers inside the Internet, so other network services can be provided to them. To simulate this, all the servers can be connected to another side of the Internet cloud via a router. Figure 4-15 shows, in other side of the Internet, a router model has been connected using a PPP_DS3 link. This router is used to connect the three servers into the internet. Each serve is connected to the router suing a 10baseT link.

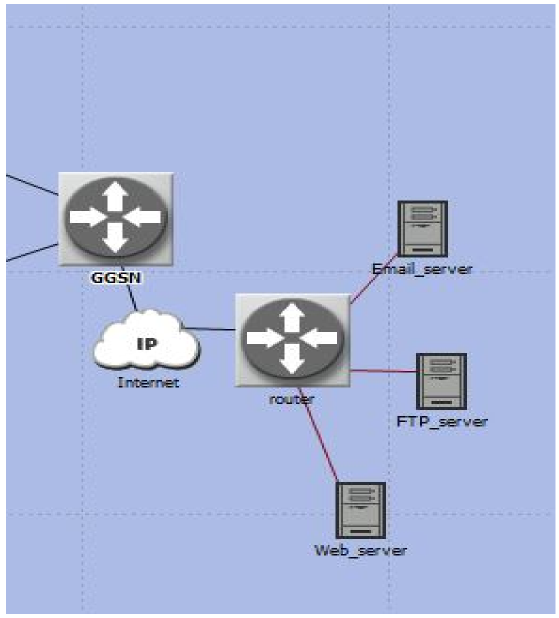


Figure 4-15: UMTS Modeling - With IP network connected

After the simulation network topology has been setup, before the simulation can be run, the important step is to configure application definitions and profile definitions. All the applications that can be used in this simulation scenario are configured in application definitions. Profiles define groups of applications that might be used by a certain group of users and also define the activities of these applications used by the group. Opnet has the nodes call “application config” and “profile config”. They can be added into the scenario. Figure 4-16 shows the final simulation topology for the UMTS network . The configuration of the configure application definitions and profile definitions will be discussed before running the simulation.

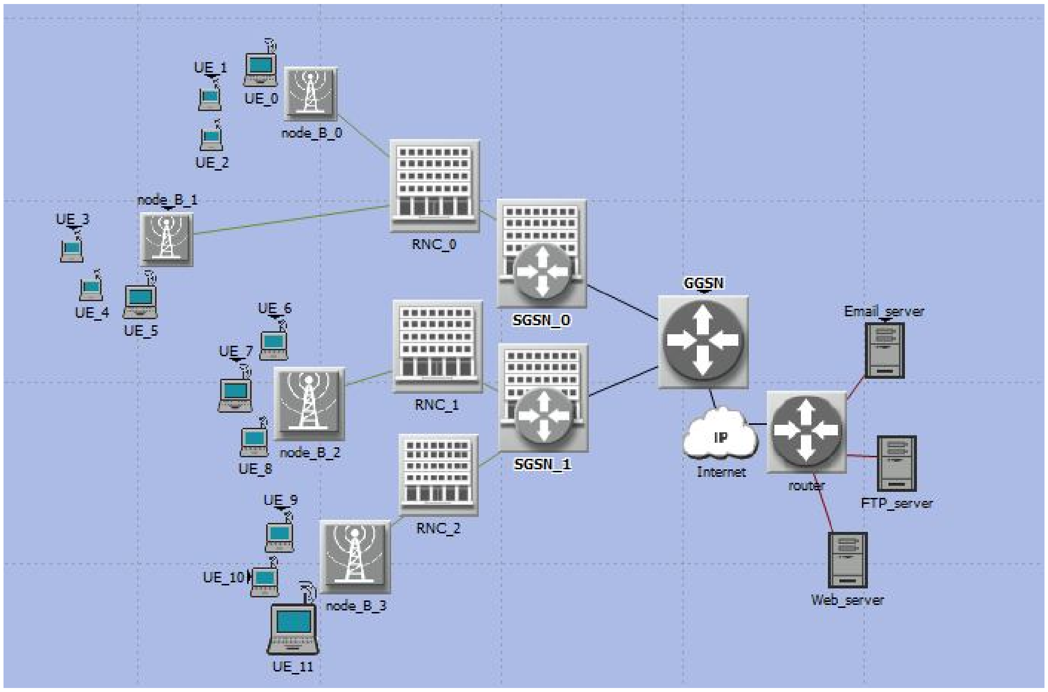


Figure 4-16: UMTS Modeling – Final simulation topology

4.3.2. Configure the simulation

After the simulation network environment has been setup, the next step is to configure applications. Opnet has some most common applications predefined. What needs to be done is to add these applications into the Applications Definitions and configure their behavior. In this research, VoIP and some other applications are to be added and configured. Before we start this, firstly we change the scenario name from “scenario 1” to “UMTSGSM quality” to represent that in this scenario, only GSM quality VoIP calls, which use GSM FR codec, will be made, no other applications will be running. After this scenario is configured, it can be duplicated and more applications can be added in.

In the Attributes Dialog of Applications Definitions, under the “Application Definitions” setting, the value of “Number of Row” represents the number of Applications defined (As shown in Figure 4-17).

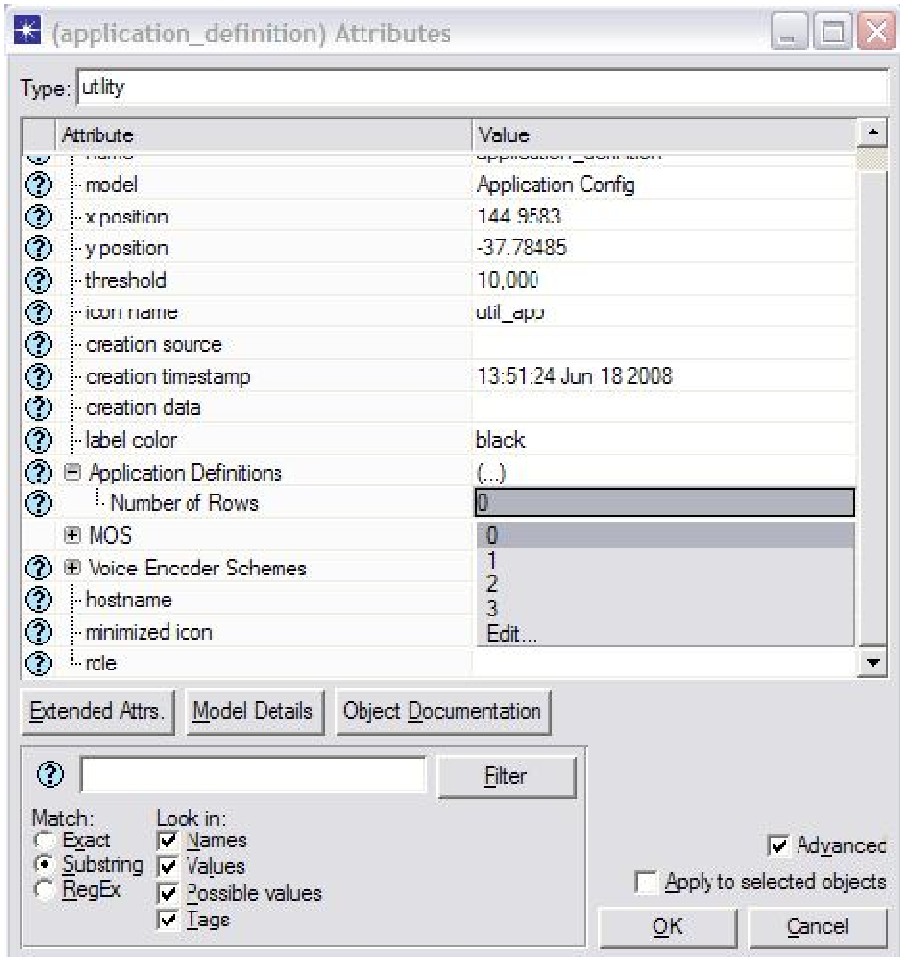


Figure 4-17: Application Definitions - Number of Applications

Start with one application. After a new Row of Applications Definition has been added, a name should be given to the application. In this case, “VoIP (GSM Quality)” has been given. If we open the “Description” setting, there are options including the basic predefined applications and a custom application which can be edited to simulate some special applications such as Email, HTTP, Ftp and Voice. Figure 4-18 shows all the options available for the application definitions.

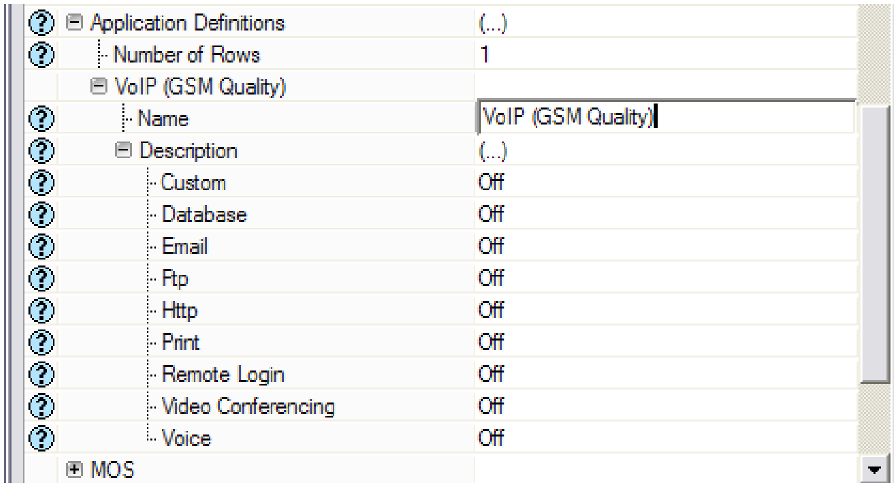


Figure 4-18: Application Definitions - Predefined Applications

In this case, the Voice application is going to be selected and configured. Opnet also have some predefined voice application settings. The options will drop down when the “value” cell of the Voice setting is clicked (shown as Figure 4-19). One of these can be selected, also the “Edit” option can be clicked to custom the application. In this scenario, “GSM Quality and Silence Suppressed” is selected.

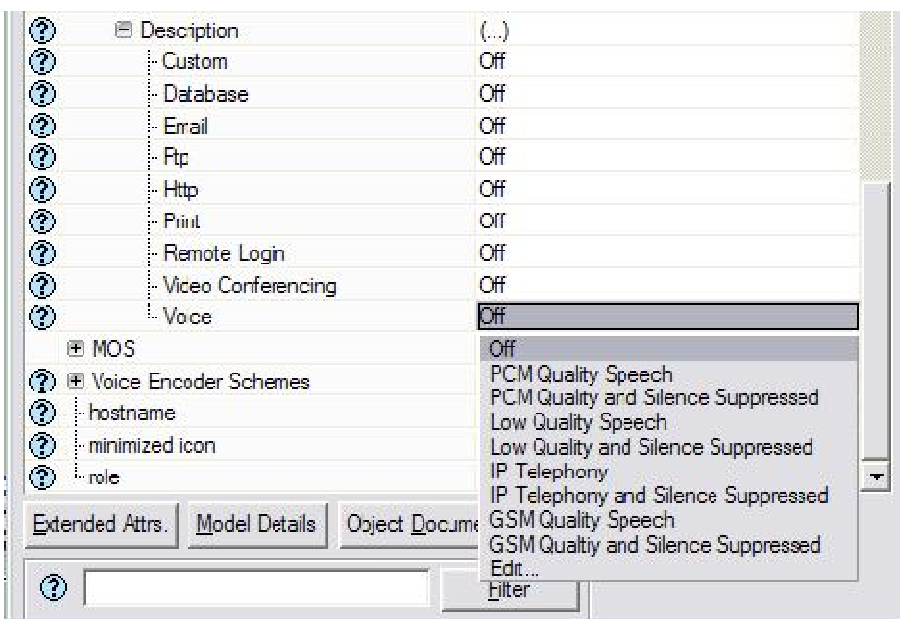
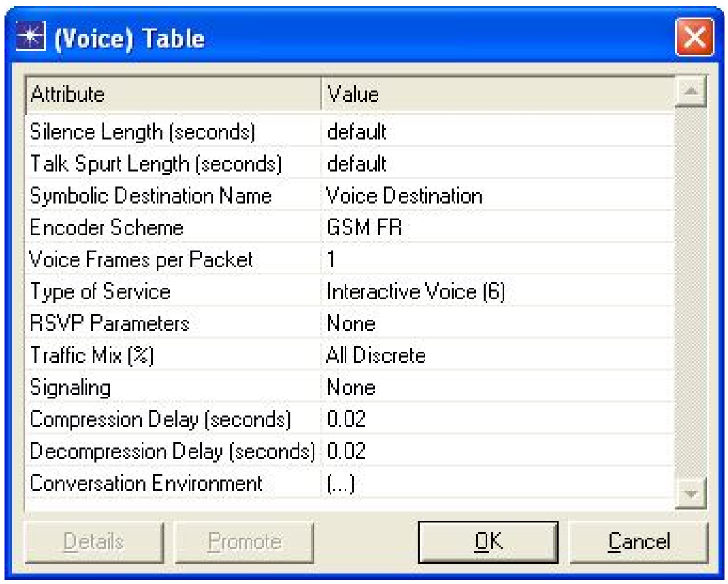


Figure 4-19: Application Definitions - Voice Application Options

After the "GSM Quality and Silence Suppressed" is selected, all the settings of this particular application can be edited by clicking the "Edit…" option. Figure 4-20 shows the settings table for the voice application. In this case, all these settings are kept as default values.



Attribute	Value
Silence Length (seconds)	default
Talk Spurt Length (seconds)	default
Symbolic Destination Name	Voice Destination
Encoder Scheme	GSM FR
Voice Frames per Packet	1
Type of Service	Interactive Voice (6)
RSVP Parameters	None
Traffic Mix (%)	All Discrete
Signaling	None
Compression Delay (seconds)	0.02
Decompression Delay (seconds)	0.02
Conversation Environment	(...)

Figure 4-20: Application Definitions - Voice settings Table

In the same way, more applications that will be used in this project can be defined. In this research project, six application definitions have been configured including three VoIP applications (using different CODECs), Email, FTP and Web browsing. Figure 4-21 lists the six application definitions. These application definitions then can be used to configure profiles which eventually will be used in the network models.

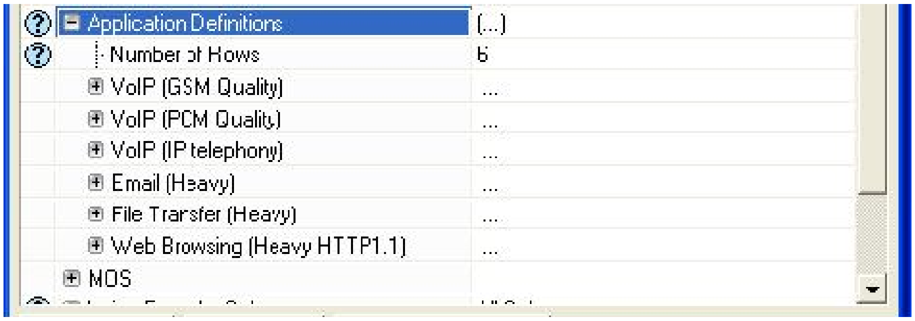


Figure 4-21: Application Definitions

A profile is used to model typical application usage of a user or workstation. It normally contains a set of applications, which can be specified when, how long, and how often these applications are typically used. In this research project, only two profiles are configured for the twelve UMTS workstations. To configure the profile definitions, open the Profile Definition attributes Editor. Figure 4-22 shows how to add one or more profiles into the Profile Definition.

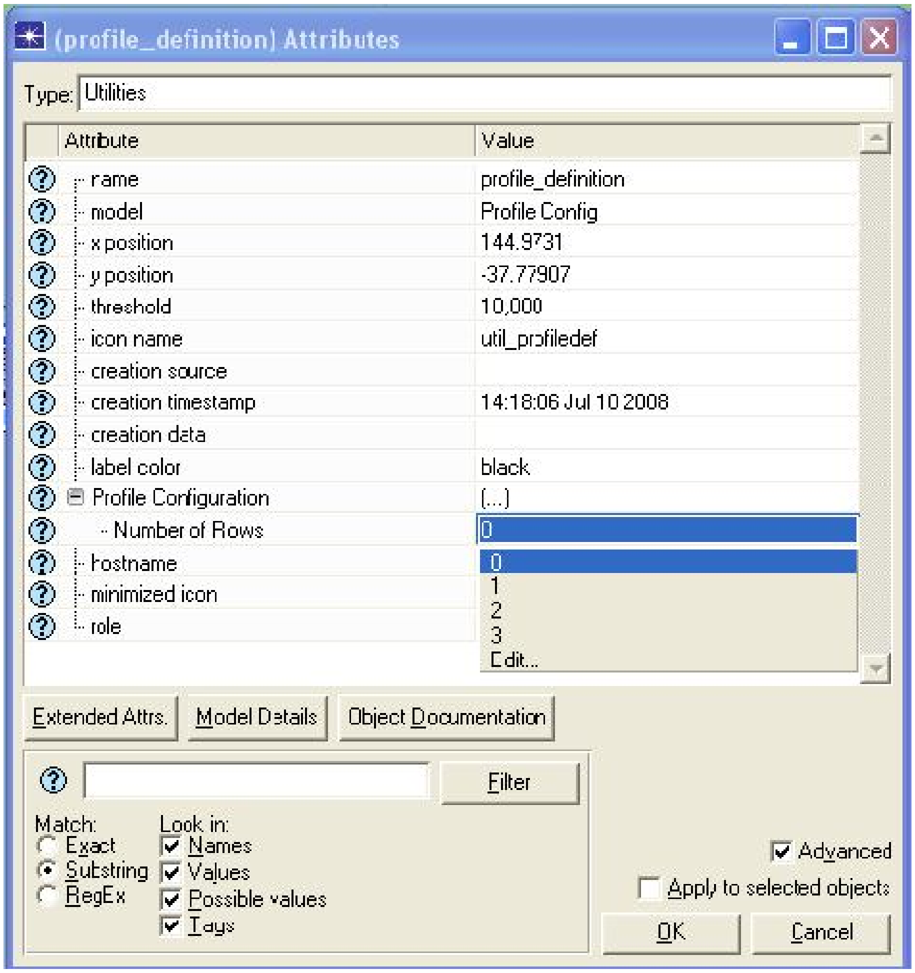


Figure 4-22: Profile Definitions - Adding a new profile

The first profile is named VoiceEndUser to present the users of VoIP applications, and the second one will be named InternetEndUser profile, to present the users for the other IP applications.

Figure 4-23 also lists the profile's time-related attributes. Every profile can be repeatedly run during a simulation. The time-related attributes define how the profile will be run in a simulation. In this research project, each profile just runs once in every simulation, so the "Repeatability" attribute is set to "Once at Start Time" and the "duration" is set to "End of Simulation".

It may take some time for the whole network to get converged. Before that, the applications should not be started. The "Start Time" attribute is used to control the start time of the profile in which the applications are. In this case, it is set to “uniform (120, 140)”, which will start the profile between 120 seconds and 140 seconds after the simulation started.

The “Operation Mode” attributes determines how the applications in a profile will be run when there is more than one application in it. They can be run either at the same time (Simultaneous Mode) or one by one (Serial Model). In this case, only one application is added in the profile project, so the profile “Operation Mode” can be set to either “Simultaneous” or “Serial”. In this research project, more applications will be added in different scenarios which will be created by duplicating this scenario. The applications will be run at the same time to investigate the impact on the VoIP applications from other applications. For that reason, the “Operation Mode” is set to “Simultaneous”.

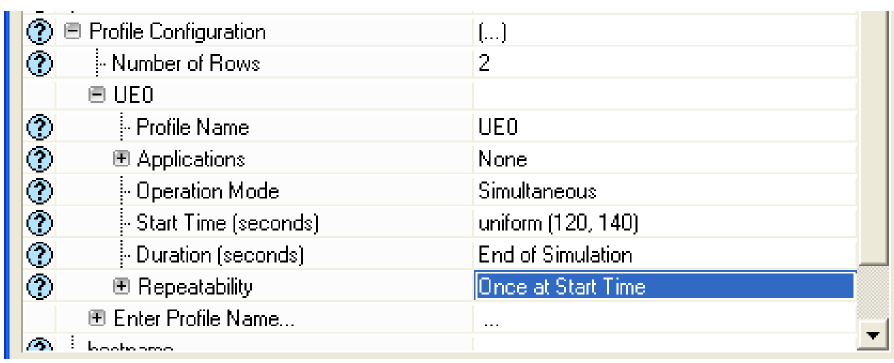


Figure 4-23: Profile Definitions - Profile time-related attributes

Inside a profile, one or more applications can be configured. In this scenario only one application is added (more applications will be added in other scenarios). These applications can only be chosen from the application definitions that have been previously configured.

Similar to the profiles, every application also has some time-related attributes (as shown in Figure 4-26). In this case, the profile is in “Simultaneous” mode, so the “Start Time Offset” refers to the offset of the first

instance of the application, from the start of the profile. It is set to "constant (0)", so the application can start as soon as the profile starts.
In this scenario, many VoIP calls will be simulated. Each call is set to 2 minutes long and the interval between each call is set to 5 minutes. The application time-related attributes are set accordingly.

To run all these defined applications in a simulation, they have to be deployed in the actual network models.

Most applications here are server-client applications, while the VoIP application is a little bit different. The calling and called party in a VoIP application are in the same position except in the call establish stage. In Opnet, VoIP applications are configured in the same way, like server-client applications. The "calling party" is configured like a client and the "called party" is treated as a server. In this project, the UE_0 is considered as the calling party and the UE_11 is going to be the called party.

In this project, three servers need to be configured. To configure a server, open the server's "attribute editor" dialog, under the "applications" group, the "application: supported service" is the where applications can be added to indicate that the server should provide these services. Similar methods can be used to configure the other servers. The FTP server is configured to support FTP service; the Web server is configured to support Web service; the Email server is configured to support Email service; and the UE_11 is configured to support all the three VoIP applications that have been defined in the Application Definitions.

Finally, the UMTS workstations should be configured to start the applications in the simulation. Again, in the "Attributes editor" dialog, under the "Applications", add a new profile in "Application: Supported Profiles". UE_0 and UE_11 are configured with their own profiles in "Profile Definitions". Applications can be added in later on in other scenarios, which are duplicated from this scenario. In this way, only the Profile Definitions need to be changed in other scenarios.

4.3.3. Running the simulation

The final step before running the simulation is to choose DES (Discrete Event Simulation) statistics. The DES "choose results" dialog opened by clicking the "choose individual statistics…" in the "DES" menu. There are three different statistics: Global Statistics, Node Statistics and Link Statistics. There are more detailed statistics available for selection. In this project, selected global statistics is Voice. The selected nodes statistics are Voice called party and Voice calling party.

After that, all the configurations for the scenario have been completed. The simulation has been executed and the results have been collected.

The initial simulation was duplicated and different CODECs schemes were used for the VoIP calls. Three CODECs (GSM FR, G.711 and G.729A) were used because of their impact on the statistics of the delay. Simulations were run for each CODEC with the frame size varied to 4 msec, 10 msec, and 20 msec. Additional simulations were run with the number of voice frames per packet varied from 1 to 6 voice frames per packet. The same network was used for each of the different simulations in case of background traffic and in case of no background traffic to get the impact of the background traffic on the QoS of the VoIP.

4.4. WiMAX network Modeling

In the next section the WiMAX network implementation has been discussed.

4.4.1. Configure the Topology

In case of WiMAX, a simple WiMAX network is modeled with twelve end users connected to the WiMAX network where two of these end users communicating using VoIP and the rest of end users using three other IP applications (web browsing, email exchanging, and file transfer) which represent the background traffic for the WiMAX network.

The VoIP end users are connected into the WiMAX network via different base stations. All the end users also have access to the internet, so they can access servers outside the WiMAX network, such as Web server, FTP server and E-mail server. Figure 4-24 shows the sketch topology that this research used.

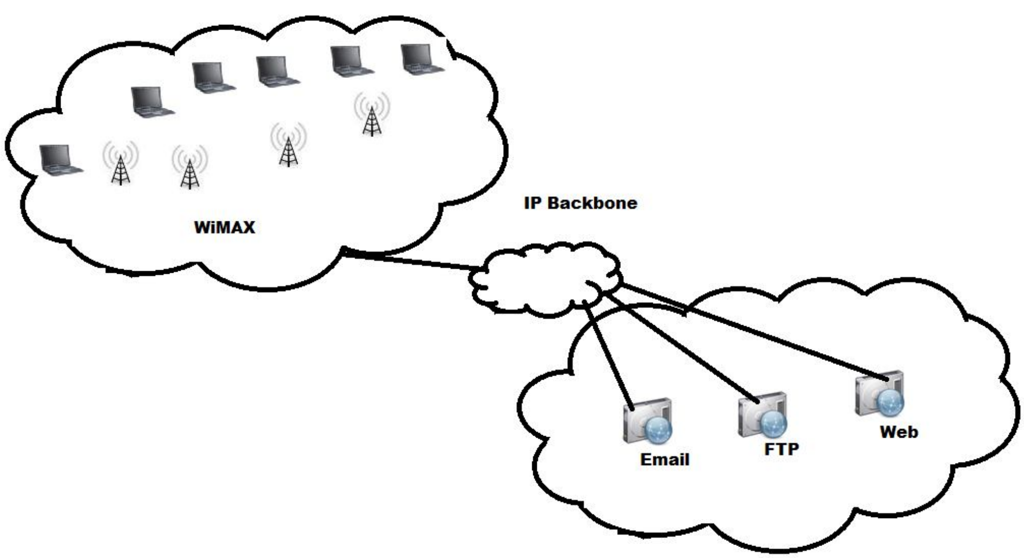


Figure 4-24: WiMAX Network sketch topology

The WiMAX network, consisting of 4 Base Stations and an IP backbone has been created by using the wireless network deployment wizard.

The same campus size for UMTS network has been used for the WiMAX network. a service class Platinum with UGS allocation to reserve the bandwidth for the network has been created.

The same steps as the UMTS network topology have been followed to complete the WiMAX topology as it shown in figure 4-25.

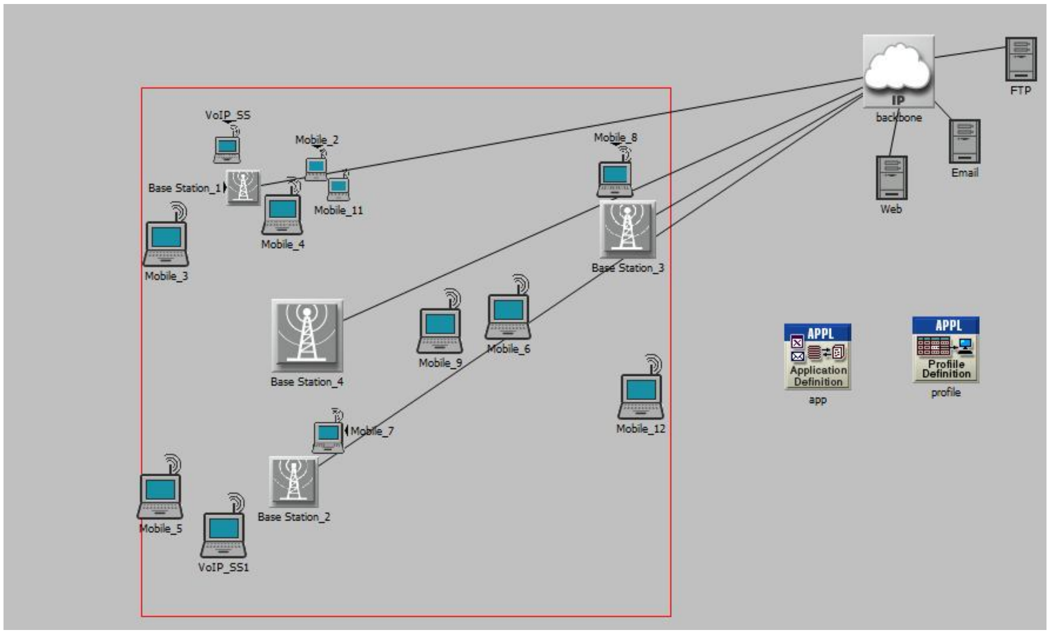


Figure 4-25: the WiMAX Simulation Model Topology

Each Subscriber station (SS) MAC Service Class Definitions has been configured as follow:

- Service Class Name Platinum
- Scheduling Type to UGS
- Maximum Sustained Traffic Rate (bps) is 2500000
- Minimum Reserved Traffic Rate (bps) is 2500000

And for the Gold service class Maximum Sustained Traffic Rate (bps) is 64000 and the Minimum Reserved Traffic Rate (bps) is 64000

And the Downlink Service configuration as follow:

- Service Class Name is Gold
- Initial Modulation is QPSK
- Initial Coding Rate is 1/2

And the same configuration for the Uplink Service has been configured.

In the case of the **Classifier Definitions the configuration are as follow:**

- Type of SAP is IP

- Traffic Characteristics are IP ToS, Equals, Interactive Voice (6)
- Service Class Name is Gold

And the Base Stations have been configured as follow:

- Type of SAP is IP
- Traffic Characteristics are IP ToS, Equals, Interactive Voice (6)
- Service Class Name is Gold

Three CODECs (GSM FR, G.711 and G.729A) were used because of their impact on the statistics of the delay. Simulations were run for each CODEC with the frame size varied to 4 msec, 10 msec, and 20 msec.

For the two networks(UMTS and WiMAX) the following parameters have been adopted :

Table 4-1: WIMAX AND UMTS NETWORK PARAMETERS

Attribute	Values
Symbolic Destination Name	Voice Destination
Encoder Scheme	GSM FR, G.711 and G.729A
Voice Frames per Packet	From1 to6
Type of Service	Gold
RSVP Parameters	None
Traffic Mix	All Discrete
Signaling	None
Compression Delay (seconds)	0.02
Decompression Delay (seconds)	0.02

4.4.2. Running the simulation

The final step before running the simulation is to choose DES (Discrete Event Simulation) statistics. The DES “choose results” dialog opened by clicking the “choose individual statistics…” in the “DES” menu. There are three different statistics: Global Statistics, Node Statistics and Link Statistics. There are more detailed statistics available for selection. In this project, selected global statistics is Voice. The selected nodes statistics are Voice called party and Voice calling party.

After that, all the configurations for the scenario have been completed. The simulation has been executed and the results have been collected.

The initial simulation was duplicated and different CODECs schemes were used for the VoIP calls. Three CODECs (GSM FR, G.711 and G.729A) were used because of their impact on the statistics of the delay. Simulations were run for each CODEC with the frame size varied to 4 msec, 10 msec, and 20 msec. Additional simulations were run with the number of voice frames per packet varied from 1 to 6 voice frames per packet. The same network was used for each of the different simulations in case of background traffic and in case of no background traffic to get the impact of the background traffic on the QoS of the VoIP.

Chapter 5

Simulation Results and analysis

In this chapter a comparison of the performance of VoIP in WiMAX and UMTS through extensive simulations was done.

Three different CODECs were used in this research. For each CODEC, 4ms, 10ms and 20ms frame sizes were used and for each frame size the Frames per Packet is set from one to six. Totally, fifty-four simulations were carried out for each network model and the results were collected.

To effectively analyze the performance, in this research work a measure of the four metrics presented in chapter 3 was done over the 54 simulation scenarios, and all end users in the same simulation use the same configurations. The Simulation duration was one hour for all calls traffic between the two end users.

5.1. Packet End-to-End Delay

Packet end-to-end delay is one of the most important performances metric in VoIP. In the case of UMTS network For G.711 CODEC, very high packet-loss rates and end-to-end delay were found. Figure 5-6 shows the results of one G.711 CODEC simulation for UMTS network. It shows that the packet loss rate is about 50 percent and this is unexpected resulte. The possible reason is that G.711 is a much higher bit rate (64Kbps) than G.729 and GSM-FR and so it need to be use with a very high speed communecation network. In general UMTS network gives worst quality of voice while using G.711 [51].

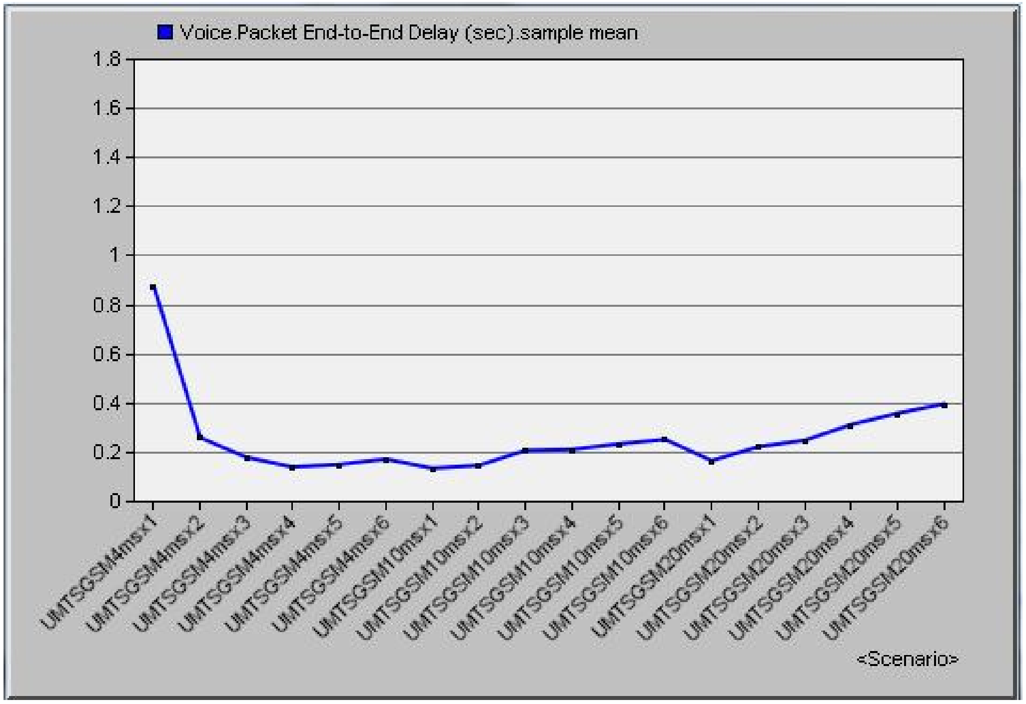


Figure 5-1: End-to-End delay for the GSM-FR and UMTS Network

Figure 5-1 shows the end-to-end delay for all of the GSM-FR CODEC schemes for UMTS network. In this figure, the sample mean values of end-to-end delay are used. From this figure, GSM-FR with 4msec frame size and 6 frames per packet have the least end-to-end delay where in the case of 4msec frame size the more voice frames in each VoIP IP packet, the less End-to-End Delay. For GSM-FR with 10msec or 20 msec frame sizes, the more frames in one packet, the more end-to-end delay.

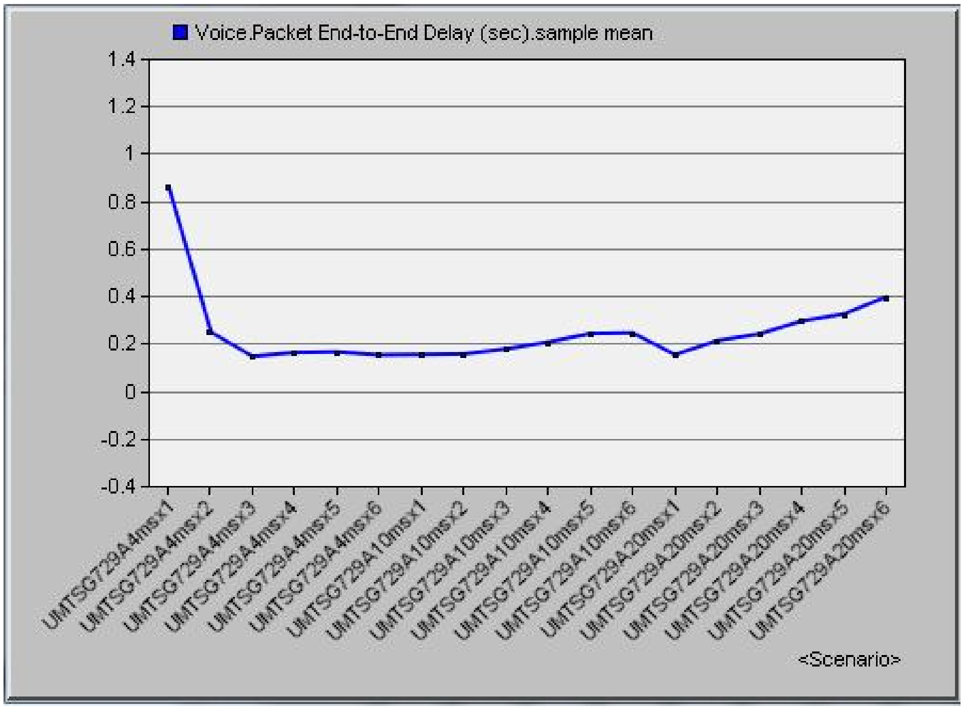


Figure 5-2: End-to-End delay for the G.729A and UMTS Network

For G.729A CODEC, the results are very similar to the results from GSM-FR for UMTS network. Figure 5-2 shows the end-to-end delay for all the G.729A CODEC schemes. In figure 5-2 the sample mean values of end-to-end delay are also used.

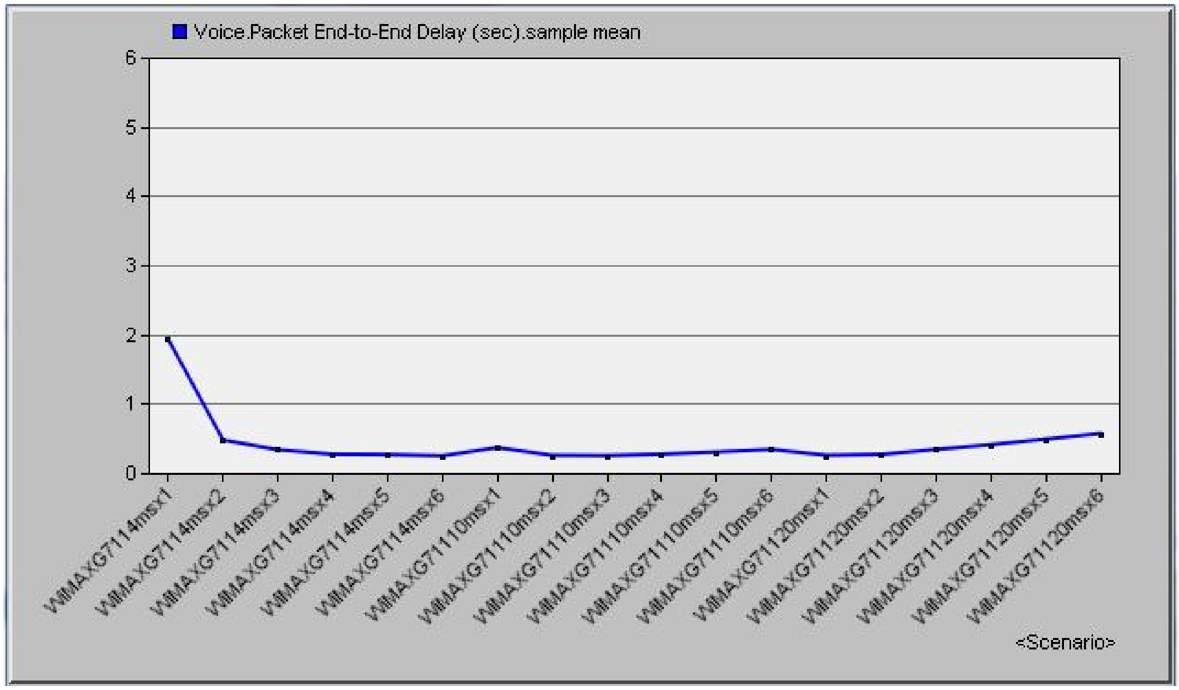


Figure 5-3: End-to-End delay for G.711 codec and WiMAX Network

Figure 5-3 shows the End-to-End delay for G.711 codec and WiMAX network. It's obvious that in the case of WiMAX network the codec G.711 can be used for VoIP application because the packet loose ratio is less than 2% as can be seen from figure 5-4. In figure 5-3 the sample mean values of end-to-end delay for all G.711 codec and WiMAX network scenarios are used. As seen from the figure, G711 codec with 4msec frame size and 6 frames per packet have the least end-to-end delay so in the case of 4msec frame size the more voice frames in each VoIP IP packet, the less End-to-End Delay. For G.711 with 10msec the end to end delay is very low for all amounts of voice frames per packet where in the case of 20msec frame sizes, the more frames in one packet, the more end-to-end delay.

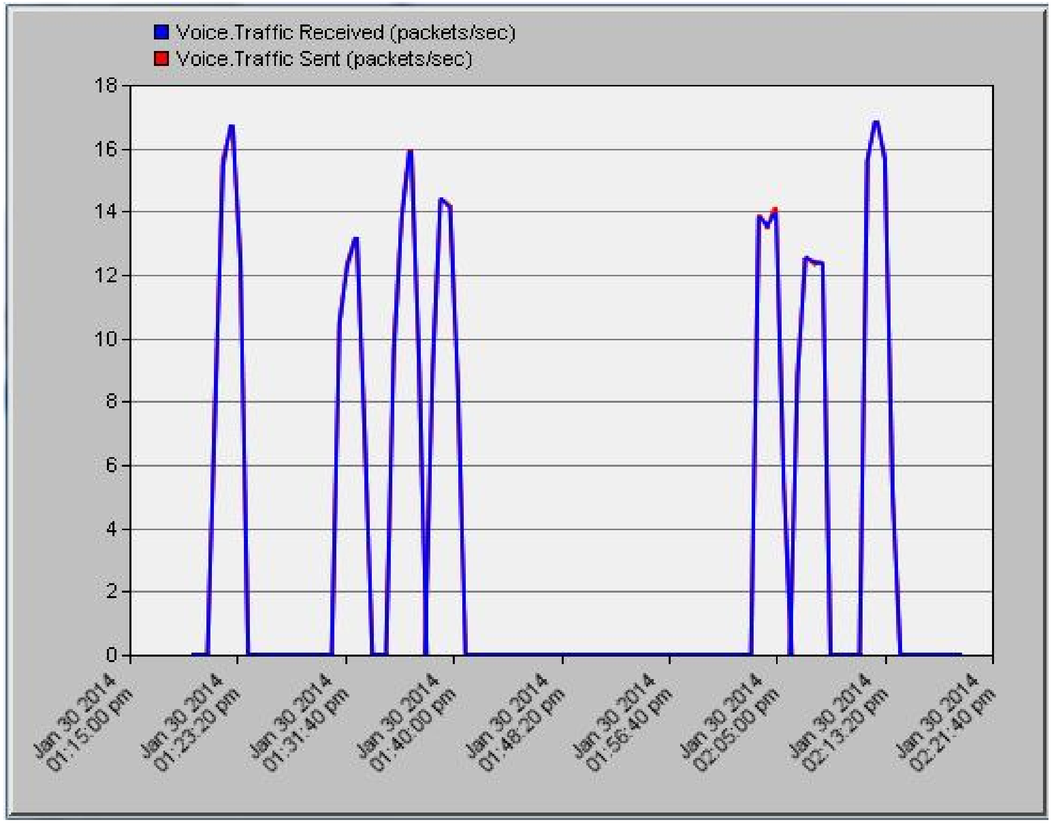


Figure 5-4: Packet loss for G.711 in WiMAX Network

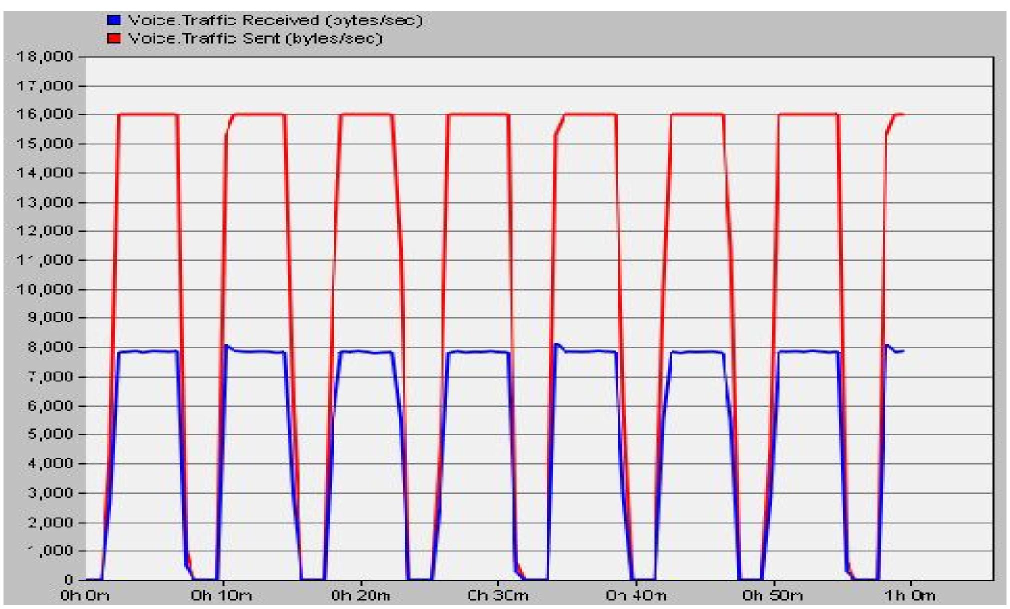


Figure 5-6: Packet loss for G.711 in UMTS Network

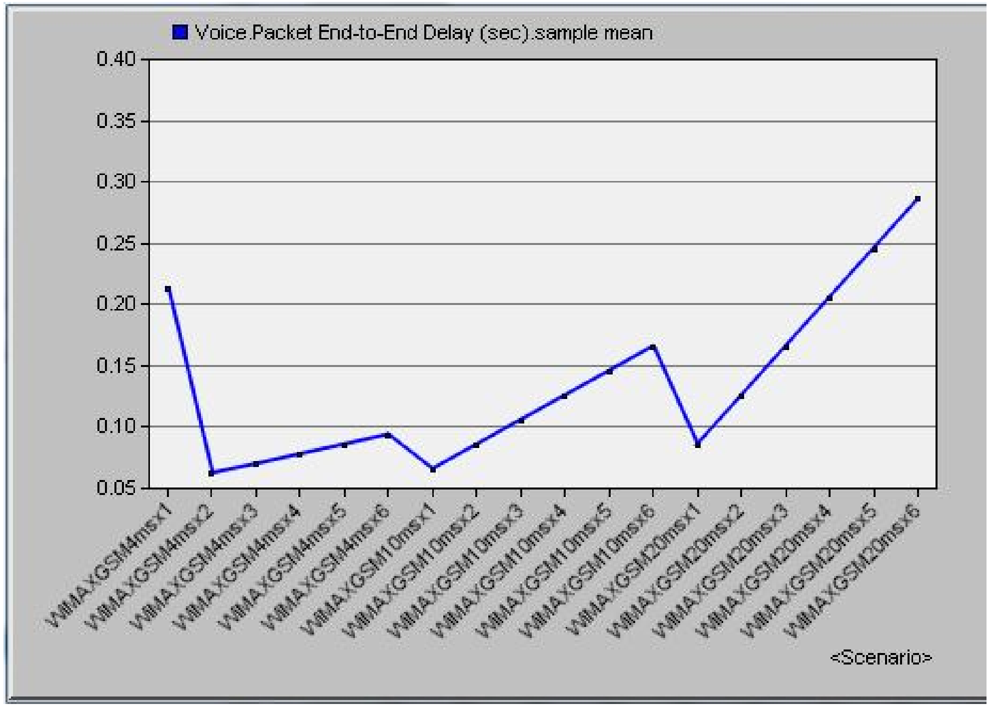


Figure 5-7: End-to-End delay for GSM-FR and WiMAX Network

Figure 5-7 shows the end-to-end delay for all of GSM-FR CODEC schemes for WiMAX network. In this figure, the sample mean values of end-to-end delay are used. And it can be observed from the figure, that the end-to-end delay value increases with the increase of the number of voice frames per VoIP packets in the case of 20msec and 10msec frame size where the end-to-end delay value also increases when the number of voice frames 2 to 6 with 4msec frame size while with one voice frame per packet the end-to end delay has its largest value for the case of 4msec frame size.

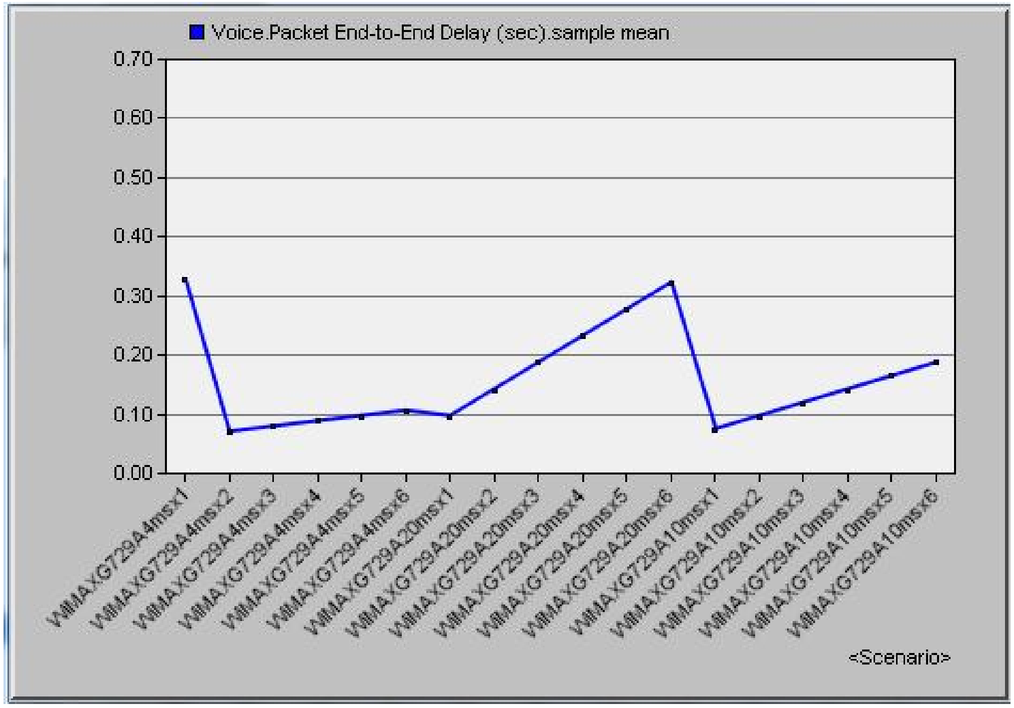


Figure 5-8: End-to-End delay for G.729A and WiMAX Network

Figure 5-8 shows the end-to-end delay for all of G.729A CODEC schemes for WiMAX network. In this figure, the sample mean values of end-to-end delay are used. And it can be observed from the figure, that the end-to-end delay value increases with the increase of the number of voice frames per VoIP packets in the case of 20msec and 10msec frame size where the end-to-end delay value also increases when the number of voice frames 2 to 6 with 4msec frame size while with one voice frame per packet the end-to end delay has its largest value.

These simulation results indicate that WiMAX can provide better VoIP services in terms of end-to-end packet delay. The reason is that WiMAX is an all-IP network [52], whereas UMTS is still a combination of circuit and packet switched technologies. However the selection of codec can affect on UMTS network where G.711 codec gives very high end to end delay and packet loose ratio. Another reason can be because of that the packet end-to-end delay is the sum of packetization delay (which is the amount of time being used for waiting for many sample frames) and network delay. Ideally, small packetization is desirable so as to minimize packetization delay. However, this results in a huge network bandwidth requirement where in UMTS systems, the conditions of the wireless channel are changing over time because of radio propagation effects or fading. On rate-controlled dedicated channels (DCHs), this results in a slow adaptation of the bandwidth currently assigned to the user [48].

It's important to mention that G.711 codec technologies apply the Pulse Code Modulation (PCM) samples method for signals of voice frequencies sampled at 8000 G.711 encoder will create a 64Kbps bitstream where reflect the need to a very high speed network and this can be the main reason behind the high packet loose in when using G.711 with UMTS network. These simulation results indicate that G.711 and G.729 can provide better VoIP services in terms of end-to-end packet delays when using WiMAX network.

The main reason behind the packet end-to-end findings can be clear when see the equation (3-3) in chapter 3 that the main deference in the results can be seen in the network delay that the network delay in the case of UMTS network is larger than the network delay in the case of WiMAX network. The reason is that WiMAX is an all-IP network [52].

5.2. Jitter Results

Jitter is the variations of time between packets transmit from source to reach destination, and it can be computed as the signed maximum difference in one way delay of the packets over a particular time interval. Jitter refers to how variable latency is in a network.

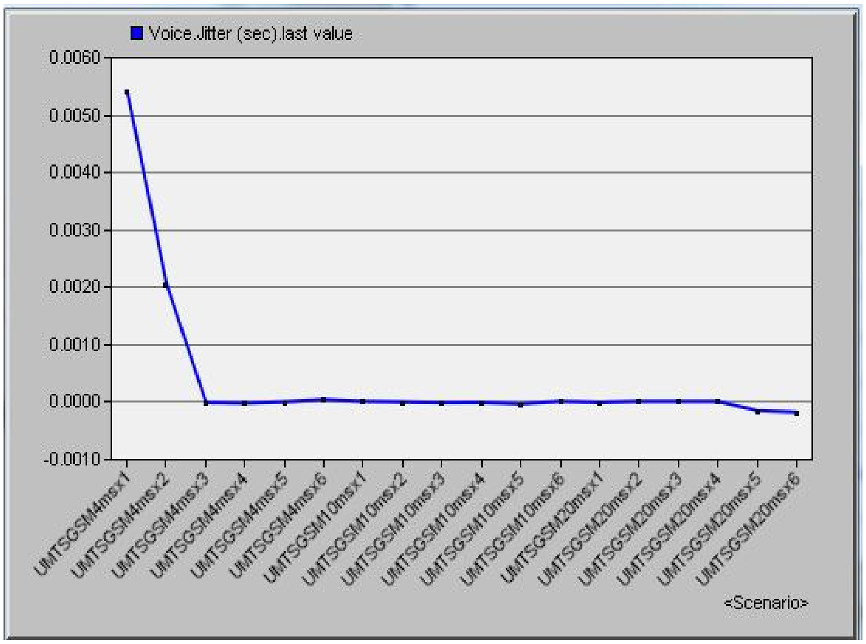


Figure 5-9: Jitter for GSM-FR and UMTS Network

Figure 5-9 shows the Jitter of the GSM-FR CODEC schemes for the UMTS network. In this figure, the last Jitter value from each GSMFR scenario is used. It can be seen that UMTS has a large range of jitter variation for the voice frame size 4msec; in the other hand the jitter is constant for the rest of scenarios. According to figure 5-9 the jitter among GSM-FR scenarios ranging from -0.0005 to 0.0055.

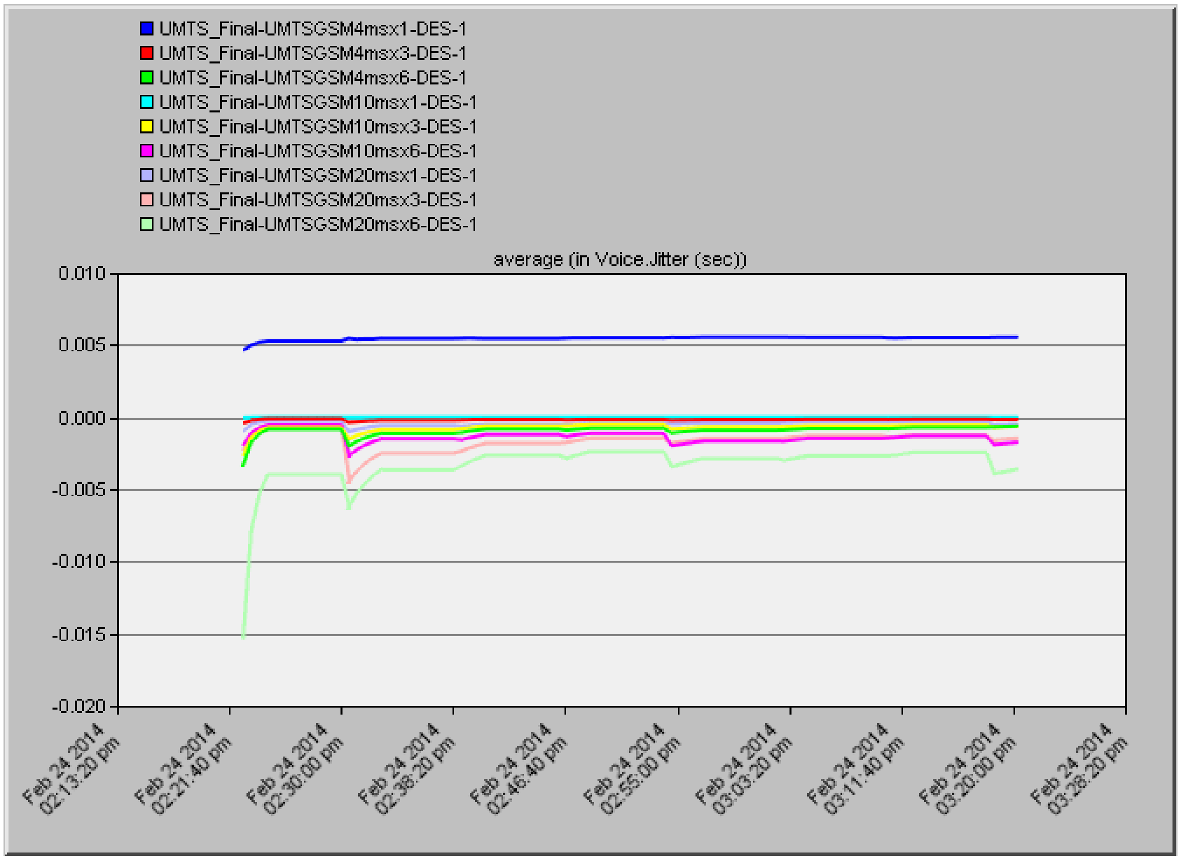


Figure 5-10: Average Jitter for GSM-FR in UMTS Network

Figure 5-10 shows a more clear view for the Jitter in the GSM-FR codec and UMTS network. The average jitter is measured. It can be seen that in the case of 20msec frame size the jitter takes longer time to converge to the stable stage when compared to 10msec and 4msec frame sizes this means that UMTS network packets transmission affected by the large voice frame sizes.

As can be seen from figure 5-10 for the frame size of 20msec with the number of frames per packet equal to 6 frames the jitter result is negative. According to equation (3-5) in chapter 3 the jitter value can be negative which means that the time difference between the packets at the destination is less than that at the source [49] and this can be due to the slow packet scheduling in UMTS network when using GSM-FR codec [24].

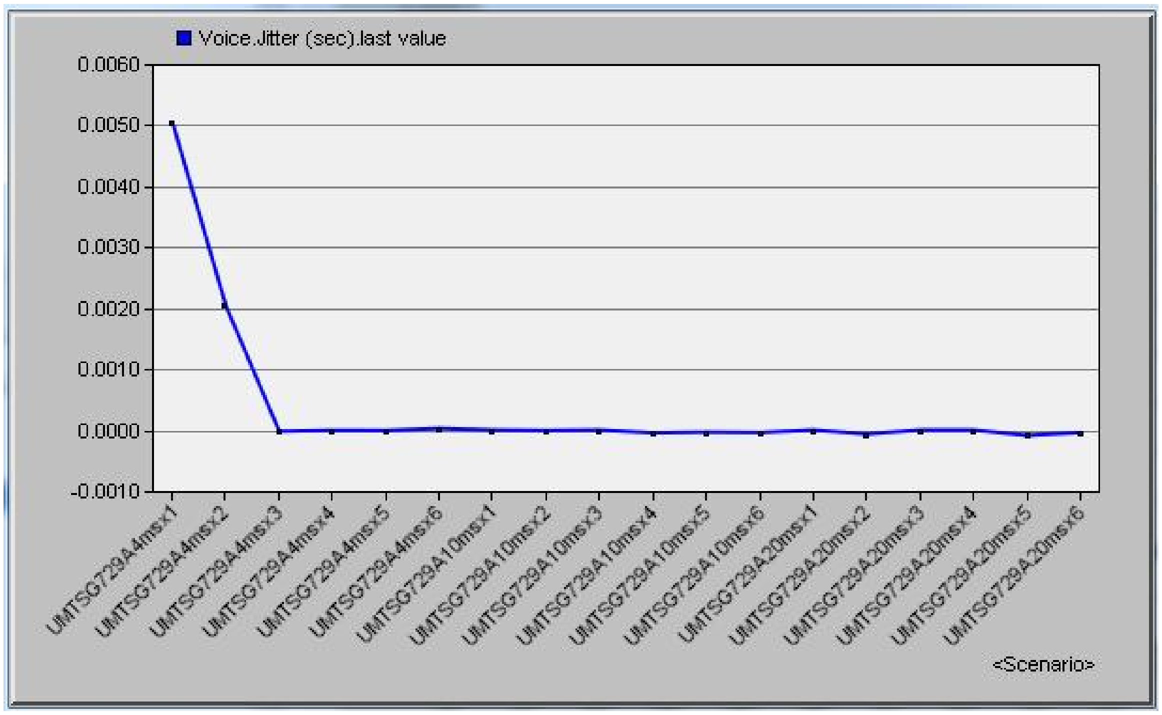


Figure 5-11: Jitter for G.729A and UMTS Network

For G.729A CODEC, also the Jitter results are very similar to the results from GSM-FR. Figure 5-11 shows the Jitter for all the G.729A CODEC schemes for the UMTS network.

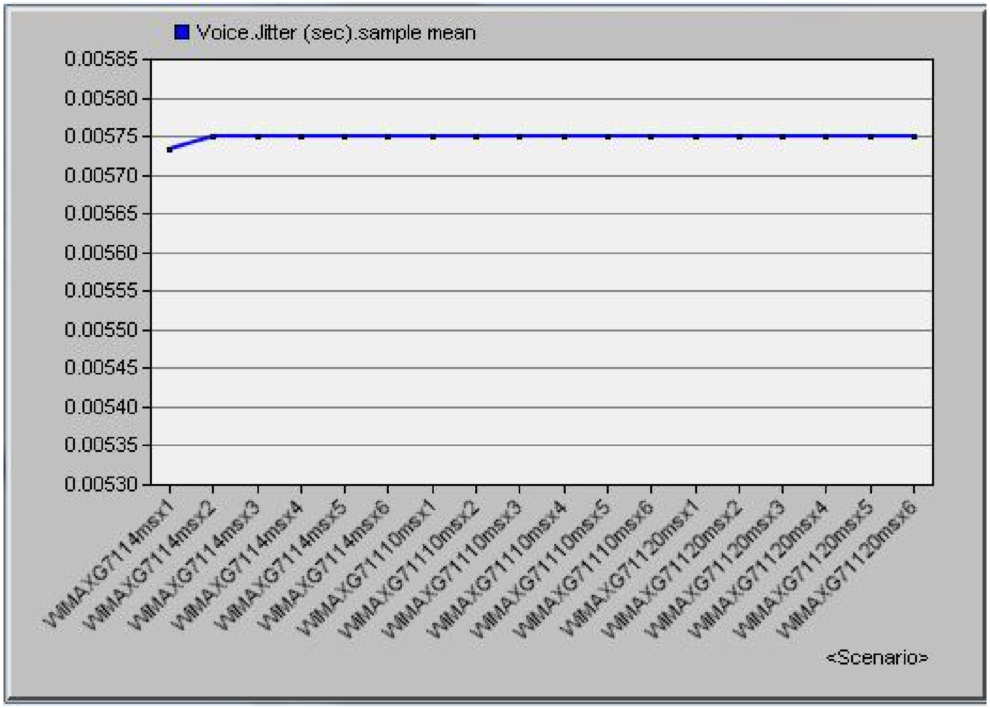


Figure 5-12: Jitter for G.711 Codec and WiMAX Network

The sample mean of Jitter values for G.711 codec and WiMAX scenarios can seen in figure 5-12. As seen from the figure the jitter value is steady for all G.711 scenarios and equal to 0.0575 except the case of 4msec frame size where the jitter range from 0.00572 to 0.00575 where equal to 0.5% of that for UMTS.

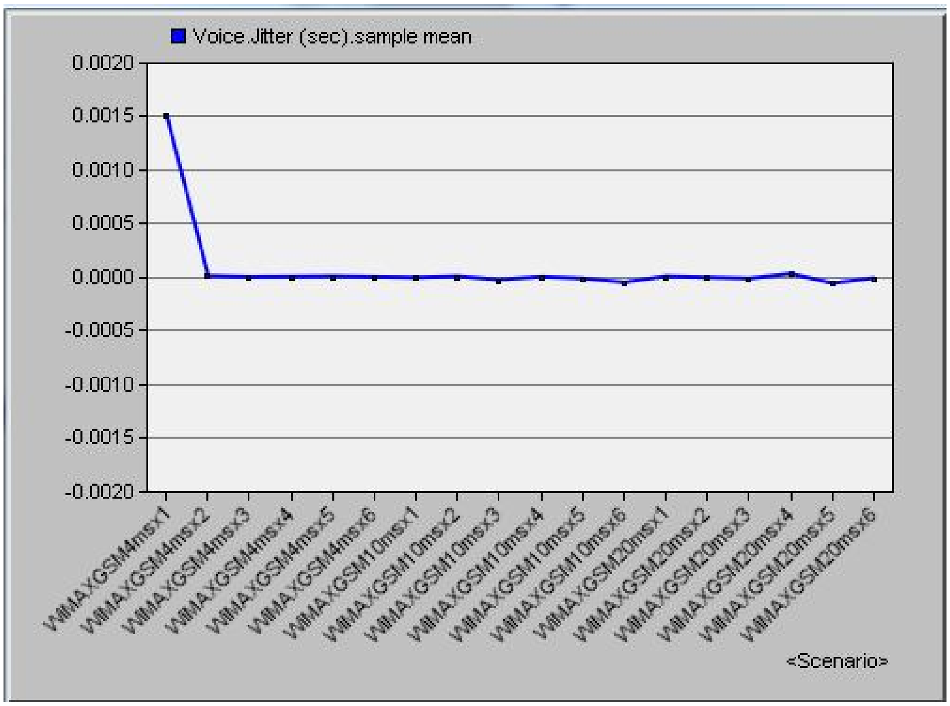


Figure 5-13: Jitter for GSMFR Codec and WiMAX Network

The sample mean of Jitter values for GSMFR codec and WiMAX scenarios can seen in figure 5-13. As seen from the figure the jitter value is steady almost for all GSMFR scenarios.

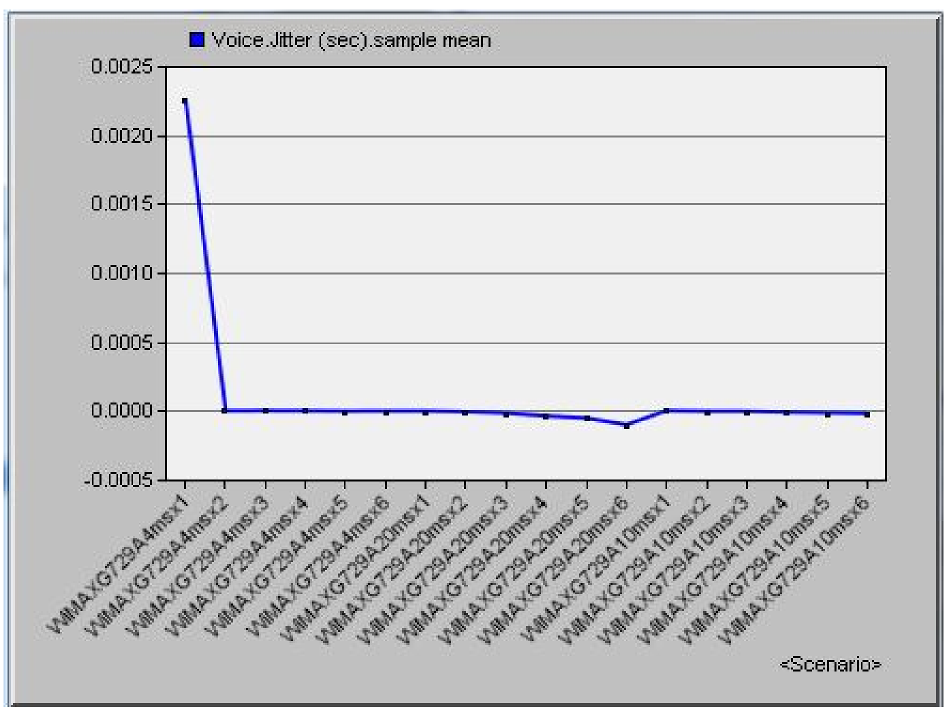


Figure 5-14: Jitter for G.729A Codec and WiMAX Network

The sample mean of Jitter values for G.729A codec and WiMAX scenarios can seen in figure 5-5. As seen from the figure the jitter value is steady for most of G.729A scenarios and the jitter is equal to 0.000 except the case of 4msec frame size with one frame per packet where the jitter is equal to 0.0021 this can be due to that in this case the network need to process more VoIP packets than in the other cases because each voice frame needs one VoIP packet which means a very large packets density.

From the above results it can be seen that UMTS has a large range of jitter variation, and takes longer time to converge to the stable stage in many simulation scenarios. For WiMAX, it has a narrow range. Moreover, it has a fast convergence to the stable state. This phenomenon can be explained as follows: as the number of IP packets increases in UMTS, the congestion in the system also increases due to the slow packet scheduling [50].

5.3. Packet Delay Variation Results

Packet Delay Variation is the variance of the packet delay. Packet delay variation plays a crucial role in the network performance degradation and affects the user-perceptual quality. Higher packet delay variation results in congestion of the packets, which can results in the network overhead [1].

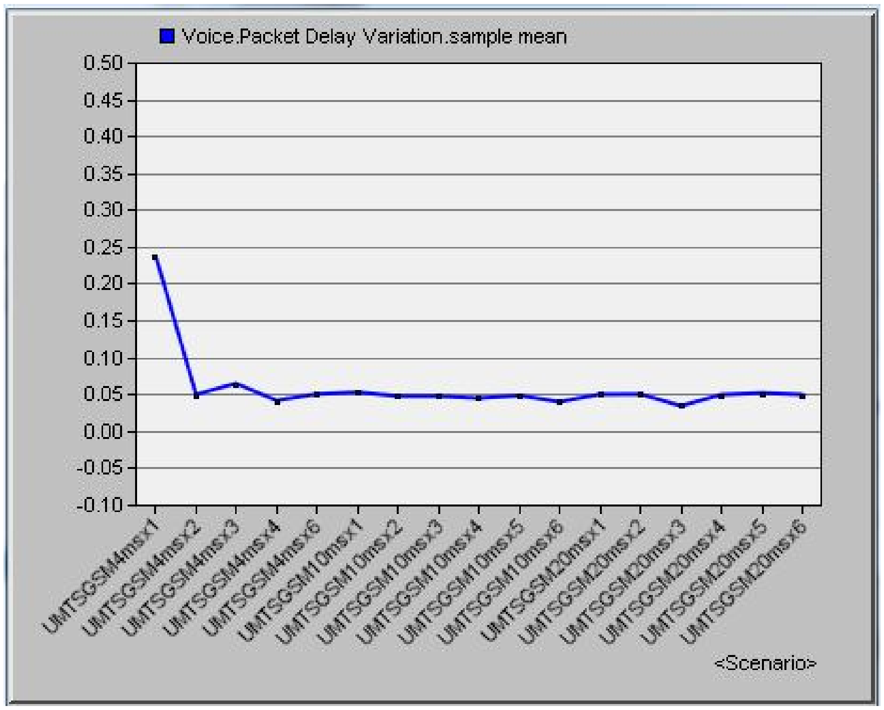


Figure 5-15: PDV for GSMFR Codec and UMTS Network

Figure 5-15 shows the Packet Delay Variation for all of the GSM-FR CODEC scenarios for the UMTS network. In this figure, the sample mean values of Packet Delay Variation are used. From this figure, in the case of GSM-FR with 4msec frame size, the more voice frames in each VoIP IP packet, the less Packet Delay.

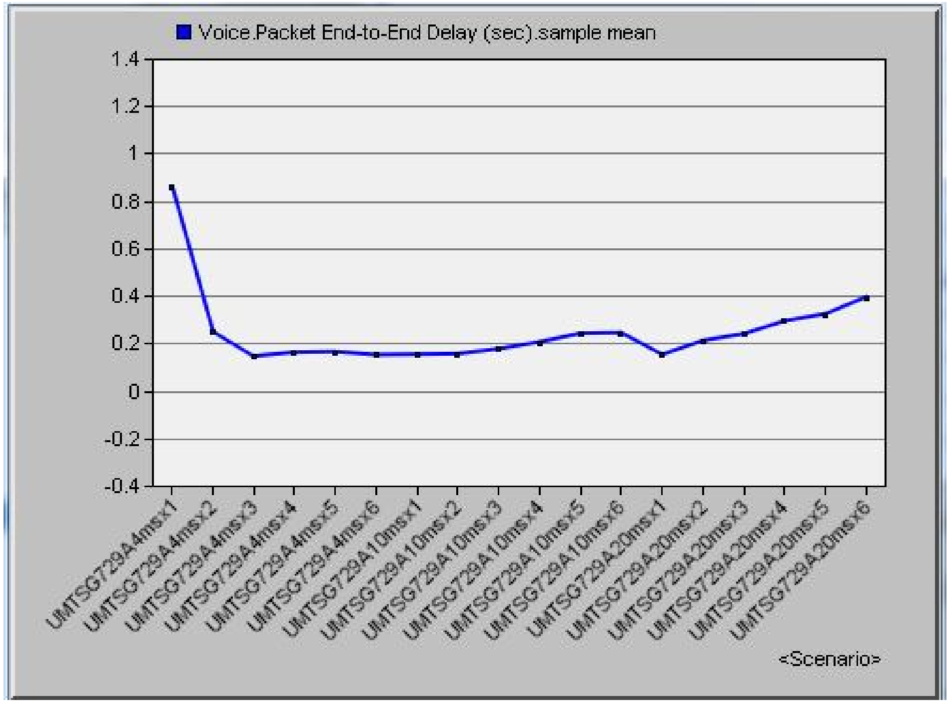


Figure 5-16: PDV for G.729A Codec and UMTS Network

For G.729A CODEC, also the Packet Delay Variation results are ranging between 0.19 and 0.23 for most simulations except 4msec frame size with one frame per packet where the PDV has its largest value and 20msec frame size with 6 frames per packet where the PDV is equal to 0.4.genarally the packet delay variation for UMTS network with G.729A codec found larger than in the case of GSM-FR codec. This can be due to that the sampling rate in the case of GSM-FR codec is larger than sampling rate in the case of G.729A where GSM-FR sampling rate is equal to 13kbps and for G.729A it’s equal to 8kbps.

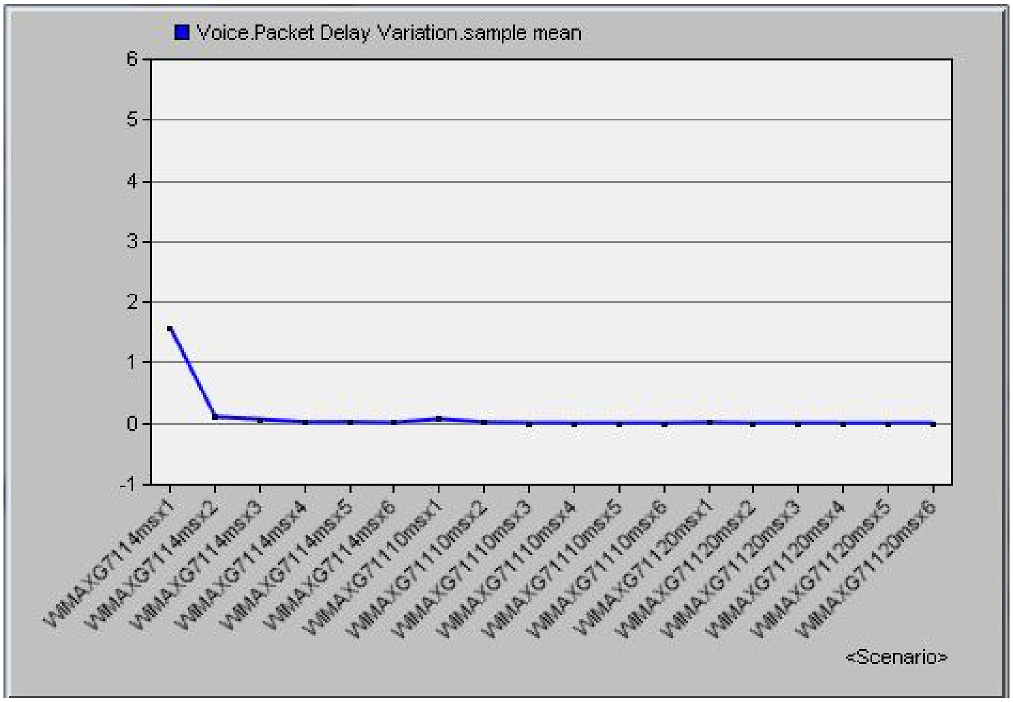


Figure 5-17: PDV for G.711 Codec and WiMAX Network

The sample mean for PDV values of G.711 codec and WiMAX network is shown in figure 5-17. As seen from the figure the PDV equal to 0.000 for almost all G.711 scenarios. The PDV has its largest value in the case of 4msec frame size and one frame per packet and equal to 1.5 and this can be due to the amount of IP packets within the network.

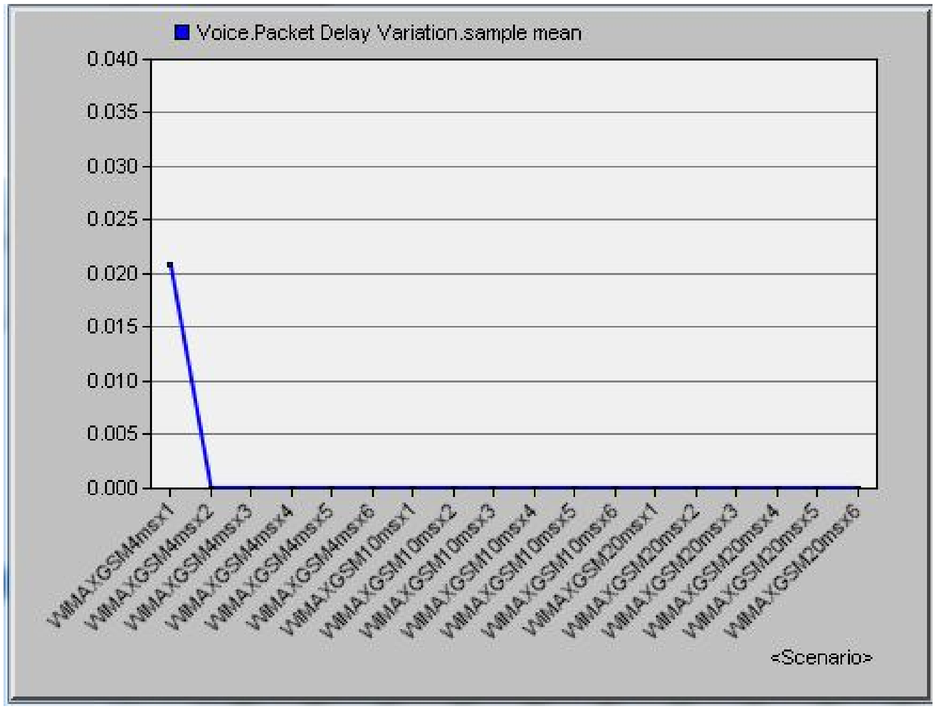


Figure 5-18: PDV for GSM-FR Codec and WiMAX Network

The sample mean for PDV values of GSM-FR codec and WiMAX network is shown in figure 5-18. As seen from the figure the PDV equal to 0.000 for almost all GSM-FR scenarios. The PDV has its largest value in the case of 4msec frame size and one frame per packet and equal to 0.021.

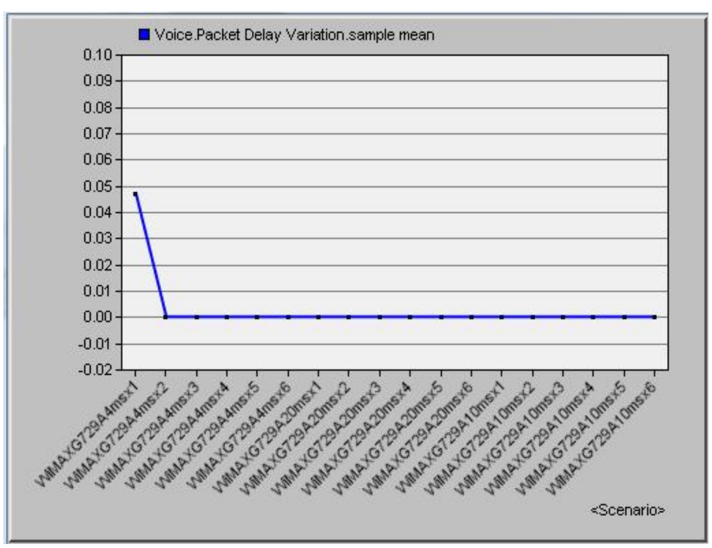


Figure 5-19: PDV for G.729A Codec and WiMAX Network

Figure 5-19 shows the sample mean for PDV values of G.729A codec. The result here is close to the result in the case of GSM-FR.

5.4. MOS

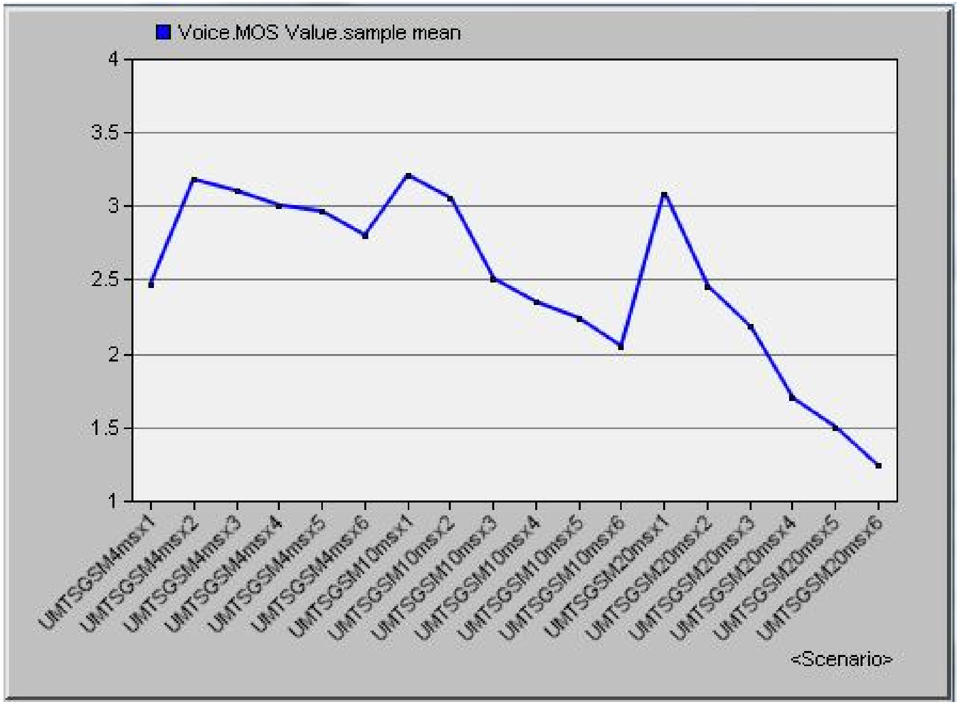


Figure 5-20: MOS Values for GSM-FR Codec and UMTS Network

Figure 5-20 shows the sample mean for MOS values for GSM-FR codec and UMTS network scenarios. As seen from the figure for the 4msec frame the best MOS has been observed for 2 frames per voice packet. For all frame sizes (4msec, 10msec, and 20msec) the MOS value decreases with the increase of the number of voice frames. The minimum MOS value has observed for the 20 msec frame size and 6 frames per packet. This result can be due to Jitter value for UMTS with GSM-FR codec scenarios.

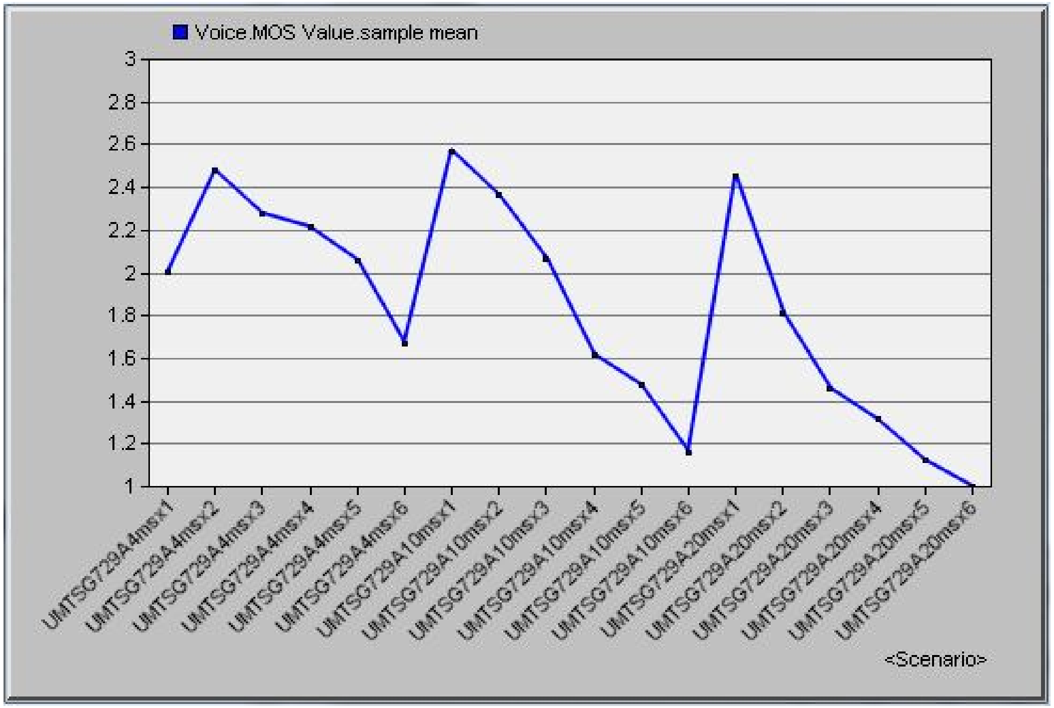


Figure 5-21: MOS Values for G.729A Codec and UMTS Network

For G.729A CODEC, also the MOS values are very similar to the results from GSM-FR. Figure 5-21 shows the MOS for all the G.729A CODEC schemes for the UMTS network.

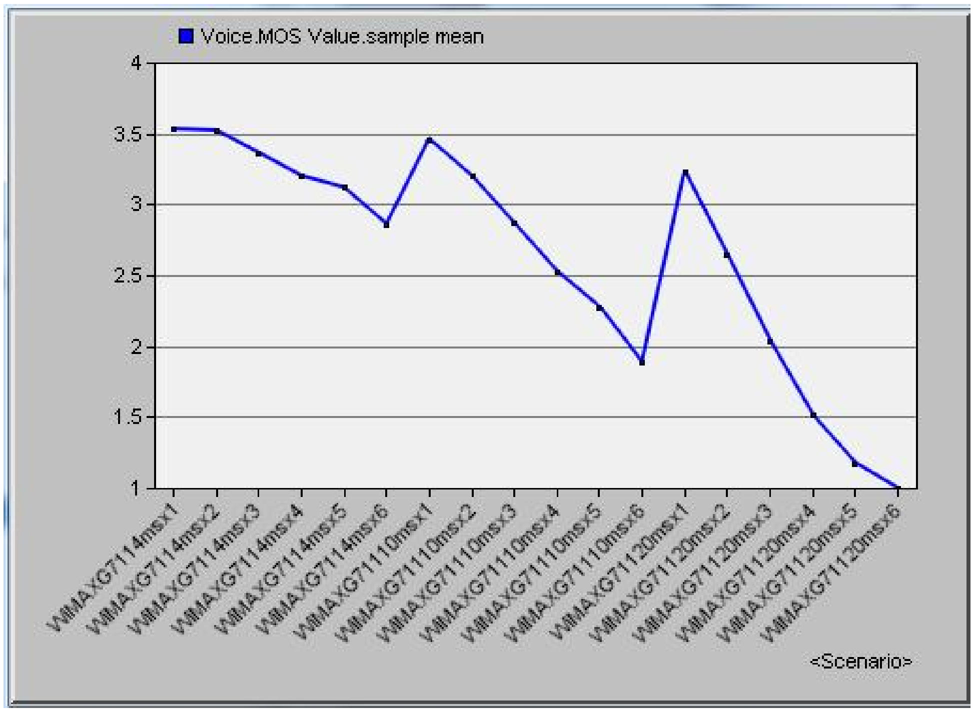


Figure 5-22: MOS Values for G.711 Codec and WiMAX Network

Figure 5-22 shows the sample mean of MOS values of G.711 codec and WiMAX network. As seen from the figure the largest MOS value has been obtained in the case of one voice frame in the VoIP packets when the frame size has a value of 4msec. for all frame sizes (4msec, 10msec, and 20msec) the MOS value decreases with the increase of the number of voice frames.

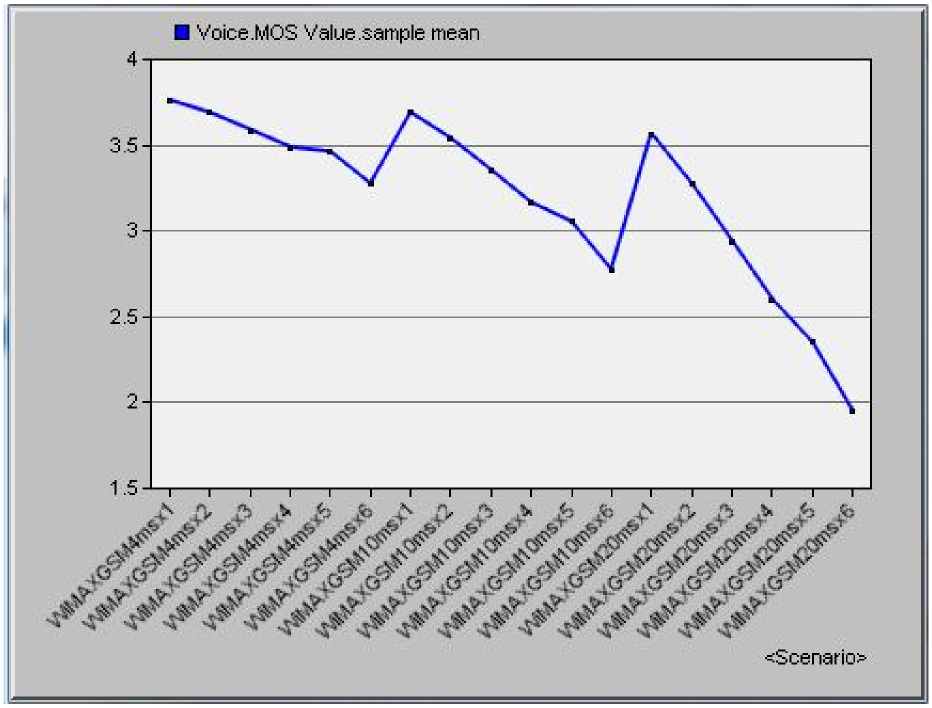


Figure 5-23: MOS Values for GSMFR Codec and WiMAX Network

Figure 5-23 shows the sample mean of MOS values of GSMFR codec and WiMAX network. as seen from the figure the MOS result that can be obtained in the case of GSMFR are larger than the MOS values in the case of G.711.

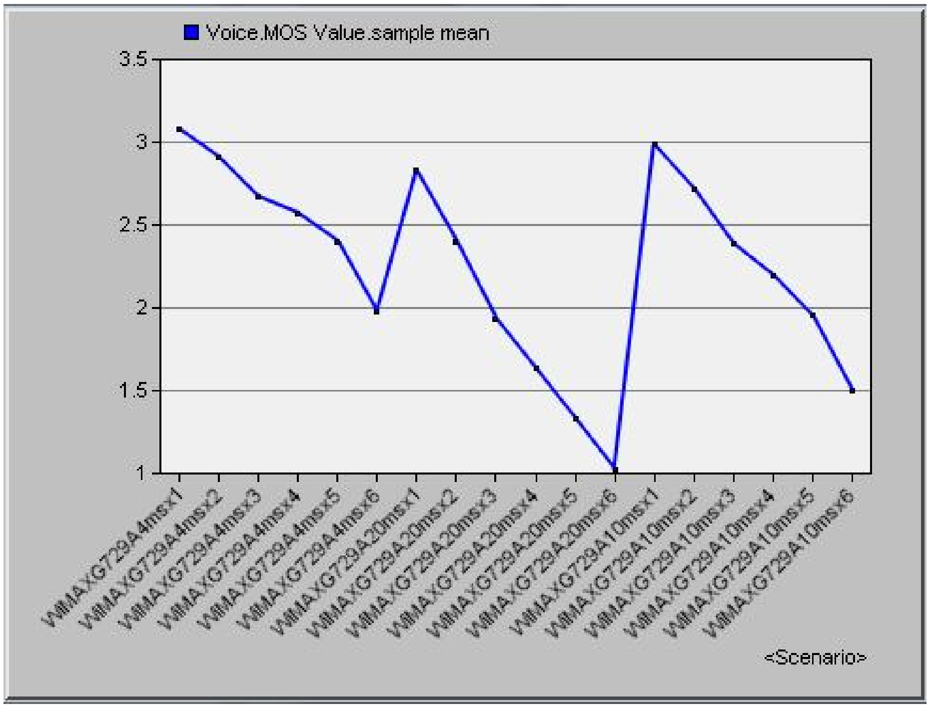


Figure 5-24: MOS Values for G.729A Codec and WiMAX Network

Figure 5-24 shows the sample mean of MOS values of G.729A codec and WiMAX network. As seen from the figure the MOS result that can be obtained in the case of G.729A is close to the MOS values in the case of G.711.

A major observation is that the average MOS decreases with the increase of number of IP packets in UMTS and WiMAX and this can be due to the increase in the network processing time. Where in the MOS has a larger value in the case of WiMAX and this can be due to the Jitter for the UMTS network. Generally the average MOS changing with the time simulation for UMTS simulations, whereas the average MOS remains roughly steady in WiMAX.

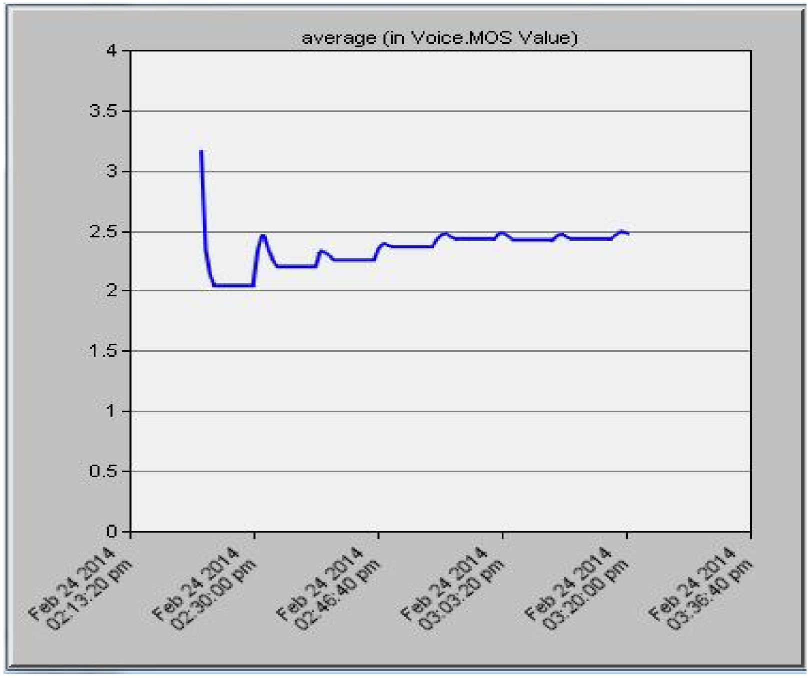


Figure 5-25: Average MOS value for GSMFR with UMTS network

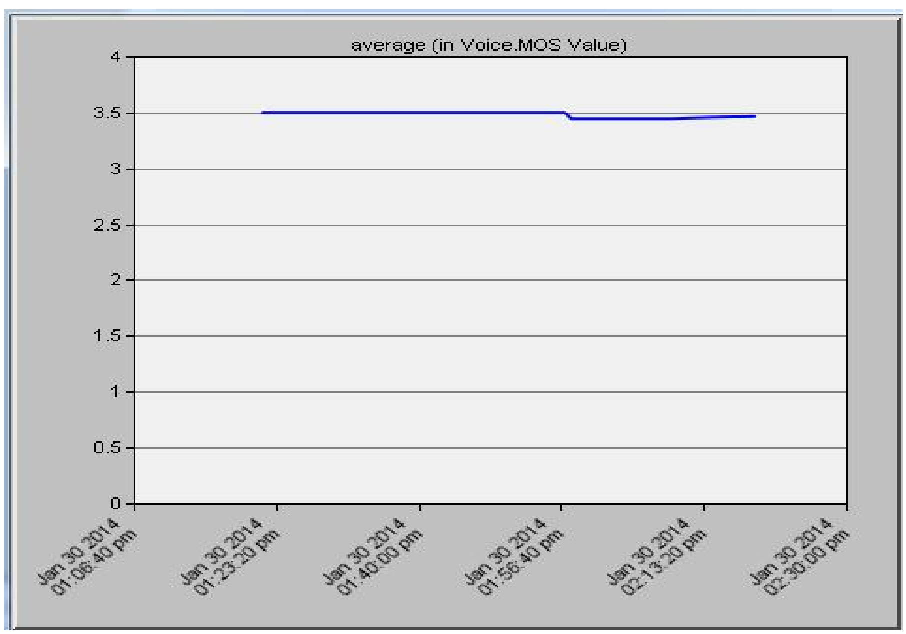


Figure 5-26: Average MOS value for G.711 with WiMAX network(for one scenario)

5.2. Results Analysis

As discussed earlier in this chapter, the G.711 CODEC was found to be unsuitable for VoIP over UMTS network where it can be used with WiMAX network. For the GSM-FR and G.729A CODECs achieved the best simulation results for a 4 msec voice frame size and six voice frames for each VoIP IP packet for UMTS and WiMAX. However the more voice frames in each VoIP IP packet, the less IP packet headers are transmitted which makes the voice transmission less bandwidth-consuming and less time-consuming since there are less IP packets to be processed.

This research come out with a general result that WiMAX network is serving better when using for VoIP calls this can be due to the following:

1. As can be seen from the figures, the average delay in WiMAX is much steady than in UMTS. And that can be clear from the Packet Delay Variation results in the case of GSM-FR where the Packet delay Variation is ranging between 0.20 and 0.03 for the UMTS network where in the case of WiMAX the Packet Delay Variation value equal to 0.00 for almost all simulations.
2. The large Jitter range in UMTS compared to that in WiMAX which casing a lower MOS value for UMTS.
3. The MOS values for the same number of frames and frame size for WiMAX and UMTS explained that the better results of MOS are in the case of WiMAX network and that means that the end users experience better VoIP service when using WiMAX than in the case of using UMTS.
4. The use of codec is limited to GSMFR and G.729A in the case of UMTS because of the large packet loose where the changing of the VoIP codec has no affect on the packet loose for WiMAX network.

It can be found in the related work [38] a study about the packetizaion effect on VoIP traffic and this study can provide this research with an explanation for of the obtained results.

In [38] the researchers found that the packetization which means how many sample frames from the speech coder to include in the packet payload is a very important issue that can affect the bandwidth requirements for the VoIP call. This is an important issue because packetization determines the payload size as well as the network bandwidth requirement. Since VoIP is delay sensitive, a small payload size is needed so that it does not cause too much delay from collecting the sample frames. A typical VoIP packet requires at least 40 bytes of overhead (20 bytes of the IP header, 8 bytes of the UDP header, and 12 bytes of the RTP header). The size of the packet overhead is usually larger than the payload. Thus, a large percentage of bandwidth is used for the transport of overhead bytes. Small packetization results in a low payload-to-overhead ratio. That is, in addition to the effective voice bandwidth, a large percentage of bandwidth is required for the transport of packet overhead. This causes extremely large network bandwidth requirement which can be obtained in the case of WiMAX more than UMTS.

In the same study the researchers formulate a relation between VoIP codecs bit rate and the bandwidth as follow:

The effective voice bandwidth = $\frac{F}{T} kbps$ (5-1)

The overhead bandwidth = $\frac{H}{nT} kbps$ (5-2)

The network bandwidth = $\frac{H+nF}{nT} kbps$ (5-3)

Where

T Sample frame delay (msec)

F Sample frame size (bits)

n Number of sample frames in the payload

H Header size of the voice packet (bits)

The effective voice bandwidth in (5- 1) is the output bitrates of the speech coder. Equation (5-2) is the amount of bandwidth consumed by the packet overhead. The total network bandwidth required by a VoIP session is the sum of (5-1) and (5-2), which gives (5-3). It can be seen from the equations that placing more sample frames into the payload helps to reduce overhead bandwidth as well as network bandwidth.

In the (same study) the researchers found that Small packetization results in a low payload-to-overhead ratio. That is, in addition to the effective voice bandwidth, a large percentage of bandwidth is required for the transport of packet overhead. This causes extremely large network bandwidth requirement. On the other hand, large packetization helps to increase the payload-to-overhead ratio, which helps to reduce the overhead as well as the network bandwidth requirement. Large packetization, however, also causes excessive amount of time being used for waiting for many sample frames, or called packetization delay. Since VoIP is delay-sensitive, this impacts the time remaining to meet the end-to-end delay budget for acceptable voice quality.

And this can be the reason behind the increase of the packet end–to-end delay with the increase of voice frames in VoIP packet in the case of WiMAX network for GSM-FR and G.729A codecs so using G.711 codec with WiMAX represents a solution for the bitrates problem and it will produce the better VoIP quality.

Chapter 6

Conclusion and Recommendations

This research presents an implementation of VoIP application over UMTS and WiMAX networks and an investigation of VoIP QoS related parameters. The study is performed through literature study and simulation implementation.

6.1. Conclusion

The objectives set out for this research were to implement VoIP traffic over UMTS and WiMAX networks and then to compare the statistical results for the VoIP call for each network. The research included implementation of a simulation that could be used to gain results suitable for analysis. The research objectives were successfully achieved.

In this study several important critical parameters have analyzed such as MOS, end-to-end delay, jitter and packet delay variation. Results obtained indicate that WiMAX outscores the UMTS, and is the better technology to support VoIP applications, compared with UMTS.

The simulation results showed that there is a correlation between the end-to-end delay, the CODEC used and the number of voice frames in each VoIP packet. The G.711 CODEC was found to be unsuitable for VoIP over a UMTS network where it can be used with WiMAX network and producing acceptable delay and low packet loose.

Overall, the simulations carried out showed that the selection of CODEC, the voice frame size and how the voice traffic is packaged into VoIP packets will affect the overall VoIP call quality.

6.2. Recommendations

The research findings and conclusions highlight the possibility for future research.

- In this research, the G.711 CODEC was found to perform very differently to GSM-FR and G.729A CODECs for UMTS network. Most of the simulations using the G.711 CODEC result in similar voice traffic with packet-loss rates about 50%. The future research work on the particular topic will include finding the reason for the large packet loss rate for G.711.
- Another research topic could focus on the G.729A and GSM-FR CODEC schemes. In this research, in some simulation scenarios, the end-to-end delays are too large for these CODECs compared with the similar scenarios and other CODECs.
- The network models and protocols provided and used within the Opnet Modeler simulation application were considered as reasonable and suitable for the research. Similar simulations could be done with other simulation applications to compare the results with the results from Opnet Modeler. Even real-world testing could be done to evaluate the simulation results.
- Another Future work can do a study of the auto-configuration mechanism for the guarantee of QoS requirement during network switching for UMTS and WiMAX.
- Although this research tried to give an impression of the main factors in WiMAX and UMTS technologies affecting the use of real time applications many aspects have not been covered in this research. For future research more attention has to be drawn to quality of service requirements in the WiMAX and UMTS systems.
- Another future work can do a study on the affect of using IPv6 instead of IPv4 overhead and the affect of changing the network specifications.

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