PERFORMANCE ANALYSES OF

VoIP OVER IEEE 802.16 BEST-EFFORT CLASS

BY

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Dedicated to my parents

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ABSTRACT

Worldwide Interoperability for Microwave Access (WiMAX) is a new technology that can provide long-distance broadband wireless access in accordance with the standards set forth in IEEE publication 802.16. Voice-over Internet Protocol (VoIP) is a rapidly growing technology for the transport of voice over data networks such as WiMAX. The growth in using such application is due to the ability of integrating voice and data traffic over one network, which will lower the cost and improve network management.

This paper contributes to our understanding of how is it realistically possible to use a Best-Effort (BE) class of fixed WiMAX in support of real-time traffic such as VoIP. The performance of VoIP over unsolicited grant service (UGS) and BE classes on fixed WiMAX was investigated, and the maximum number of VoIP sessions that can be supported with good Mean Opinion Score (MOS) values was measured. The study considered how system performance is affected by increasing the number of VoIP sessions and voice codec schemes and by using the Voice Activity Detection (VAD) method. By means of OPNET simulations, this study provides a clear description of VoIP performance over both UGS and BE classes, voice quality, and a statement of the maximum number of VoIP calls that can be maintained with G.711 and G.729 coding. The results indicate that BE can be used to support VoIP traffic with excellent MOS values.

خلاصه الرسالة

تعد تقنيه الواي ماكس واحده من اهم التقنيات الحديثة لنقل البيانات عبر شبكه الاتصالات اللاسلكية حيث توفر هذه التقنية امكانيه نقل البيانات لمسافات طويله وبسرعات عالية. وسوف تستخدم هذه التقنية في مجال التطبيقات التي تتطلب سرعه فائقة لنقل البيانات مثل اجراء المكالمات عبر خطوط الشبكات حيث إن مثل هذا الاستخدام سيوفر على الشركات المشغلة الوقت والمال في اداره الشبكات الصوتية وشبكات المعلومات من خلال امكانيه اداره شبكه واحده تخدم كل التطبيقات.

يقدم هذا البحث دراسة مفصله حول اداء مثل هذه الشبكات في امكانيه نقل المكالمات ويقوم أيضا بحساب درجه دقه الصوت ومقارنتها بما تقدمه شبكات الهاتف المعتاده. ويتطرق هذا البحث ايضا الى دراسه امكانيه معامله جميع التطبيقات بنفس الاهميه مما يساهم في الغاء التعقيدات المتطلبه حاليا في اجهزه الارسال والاستقبال التي تخدم هذه التقنيه حيث انه لابد لهذه الاجهزه من التعامل مع كل تطبيق على حده وجدوله البيانات الوارده والصادره حسب اهميه التطبيق. من خلال هذه الدراسه يقدم هذا البحث تصورا عن مدى امكانيه تطبيق هذه الفكره ومقارنتها بما اسست عليه هذه التقنيه من خلال عمل مقارنه بينهما من خلال دقه وضوح الصوت وعدد المستخدمين الممكن خدمتهم تحت نفس الظروف. استخدا التقنيه الافضل لمعالجه الصوت الى بيانات واحده من الجوانب التي سيتم دراستها ايضا فيي هذا البحث حيث يقدم هذا مقارنه بين اشهر تقنيتين في معالجه الصوت واعطاء مميزات وسلبيات كل منهما.

نهايه البحث يقدم بعض التوصيات المستخلصه من هذه الدراسه والقيم الافضل اذا رغب بتطبيق هذه الفكره وتوضيح الجوانب المتبقيه والتي يمكن دراستها لاحقا.

CHAPTER 1

INTRODUCTION

The demands on the Internet are expected to become so great that they will create a huge concern for the telecommunication industry. Providers of applications and services will experience enormous demand for high-bandwidth services and applications. Any emerging network technology will need to take this into consideration and provide for rapid growth. For network operators, finding ways to provide subscribers with adequate broadband connectivity will soon be a formidable challenge [1], [2].

1.1 WIMAX AND VOIP

The technology described in IEEE 802.16, also known as WiMAX, is fast becoming a promising means of meeting this challenge. This technology, whose use with Point-to-Point or Point-to-Multipoint (PMP) to setups the Wide Metropolitan Area Networks (WMANs) are now considering, is meant to provide high throughput [3].

The Medium Access Layer (MAC) of this technology is designed to provide the required Quality of Service (QoS) for all applications. This is achieved by introducing multiple layers of services, maintaining connections, and providing unidirectional Service

Flow (SF) between the Base Station (BS) and the Subscribe Station (SS). The WiMAX technology defines and prioritizes five scheduling services for meeting the requirements of different applications. These services are unsolicited grant service (UGS), extended-real-time polling service (ertPS), real-time polling service (rtPS), non-real-time polling service (nrtPS), and best effort (BE) classes. UGS supports applications, such as voice-over internet protocol (VoIP), that generate fixed-size data packets on periodic time frames. ErtPS serves applications that generate dynamic fixed-size packets. A good example of such applications is VoIP with silent suppression application. The third scheduling class, rtPS, is used for applications, such as video, that periodically generate variable-sized packets. The fourth scheduling class, nrtPS, services non-real-time applications such as high-bandwidth File Transfer Protocol (FTP). The last class, the focus of the work described in this thesis, is the BE class. For this class the bandwidth that remains unused after scheduling is completed is assigned to the applications assigned to this service class [4–6].

VoIP is one of the most widely used applications of the Internet. More and more people, all over the world, are using it. It provides reasonable quality at low cost, or sometimes free. Accordingly, many software companies have begun investing in it, producing such applications as Skype, Google Talk, and Windows Live Messenger. As the use of such applications proliferates, and voice and data networks are increasingly integrated over the present network infrastructure, VoIP will undergo huge growth [6–8].

In WiMAX, VoIP traffic mapped to UGS or ertPS, which involves massive and costly changes on the current SS side, because SS has to be intelligent enough to manage various types of traffic and map them to different service classes [4]. Even if this is technically possible, the end-to-end delay is not usually guaranteed in internet networks, because such networks have no QoS for internet traffic. In addition, the lack of a signaling infrastructure prevents the use of all the functions of UGS and ertPs. In an effort to simplify the process, this study examines the feasibility of handling all WiMAX traffic over the simplest service class, BE. This would be more realistic, for the reasons mentioned above [9–11].

This thesis explores, by means of a simulation study, the behavior of a WiMAX network in transmitting VoIP traffic with good QoS using both UGS and BE profiles. The study involves one BS and numerous SSs. Their performance is measured in terms of end-to-end delay, jitter, packet loss, and throughput.

1.2 MOTIVATION FOR THIS WORK

In recent years, WiMAX, which is a broadband wireless access (BWA) technology, has been developed to meet the increasing need for high bandwidth and to provide seamless access to broadband service. Different sets of services are needed. For instance, the services needed for multimedia traffic differ from those needed for normal internet browsing on mobile networks. WiMAX has also been designed to provide last-mile BWA with high data rates, which can be compared with the one provided by cable modem and digital subscriber line (DSL) technology. Moreover, WiMAX provides many levels of QoS. So WiMAX provides the ability to build a unique, fast, and cheap wide-area network [12], [13].

Usually in wireless networks, a wireless medium is shared between connected users. In WiMAX, the medium is allocated differently where bandwidth allocation algorithms are designed to maximize the use of radio resources. Also, the MAC protocol of WiMAX is responsible for coordinating access, bandwidth allocation, and data transmission [1].

The need for better utilization of the internet, with simple equipment and a simple management process, motivates us to check the ability of the WiMAX MAC layer protocol to transmit very sensitive applications such as VoIP traffic by handling them in the simple management class, BE. Provided this is possible, it will reduce the management process and also reduce the complexity of WiMAX components. For reasons mentioned in the introduction, having to do with transmitting VoIP over UGS and ertPs, we also expect all up-and-coming WiMAX networks to map VoIP to a BE profile. A detailed study will help in sizing and implementing this service over WiMAX.

1.3 OBJECTIVES AND CONTRIBUTION

Most previous studies have been focused on handling VoIP traffic in the UGS or rtPS classes, and non–real-time traffic in the BE class . This thesis focuses on evaluating the capability of the WiMAX BE service class to transmit VoIP calls with acceptable quality, similar to the quality delivered by the Public Telephone Switching Network (PSTN).

There are no similar studies in the literature. Specifically, this study evaluates the limits of network capacity as a function of the number of terminals, and determines the configuration parameters that achieve maximum capacity. The work simulates a WiMAX network with different numbers of substations connected in a single-hop network. The use of such a network is referred to as the PMP mode of WiMAX. Other contributions in this work aim to:

1. Compare the ability of the UGS and BE classes in transmitting VoIP calls.
2. Identify the best attributes to maximize capacity for VoIP calls without affecting the QoS.
3. Investigate how to increase the number of VoIP calls that can be handled.
4. Study the effect of applying silent detection systems.
5. Evaluate the capabilities of several simulators to simulate VoIP transmitted over WiMAX.
6. Simulate both UGS and BE.
7. Study the results of the simulations to answer two questions: Can the BE class carry VoIP traffic with acceptable standard VoIP quality, and what is the maximum number of VoIP sessions that can be carried?
8. Recommend the values of the attributes that are best for implementing VoIP over WiMAX.

1.4 ORGANIZATION OF THIS THESIS

The rest of the thesis is organized as follows. Chapter 2 introduces background material for this dissertation: the IEEE 802.16 standard and the WiMAX technology. It then presents an overview of VoIP technology, including network topologies, VoIP protocols, compression algorithms, and QoS. Chapter 3 reviews related literature on VoIP over WiMAX BE class. Chapter 4 describes the simulation environment, including the traffic model and the values of the MAC and Physical (PHY) layer parameters, and defines the performance measures used in assessing VoIP over the BE class. Chapter 5 explains the scenario used, and discusses the results and findings of the simulations. The end of Chapter 5 presents analyses of the collected results, and evaluates the overall performance of WiMAX network. Chapter 6 presents some recommendations regarding the most important parameters for transmitting VoIP over WiMAX. Chapter 7, the conclusion, discusses some of the issues that remain open, and suggests directions for future research.

CHAPTER 2

BACKGROUND

The primary objective of this research was defined in Chapter 1. In this chapter, background material is presented to help us understand the following chapters of this thesis.

2.1 WIMAX

This chapter provides an overview of WiMAX technology and focus more on its QoS, along with details about the BE service class.

2.1.1 WHAT IS WIMAX?

In 1998, the Institute of Electrical and Electronics Engineers (IEEE) 802.16 group was established to develop an air-interface standard for wireless broadband. In the beginning, the standard was developed for operation on a frequency band ranging from 10 to 66 GHz. The initial version of this standard was released in December 2001 and was designed to work with a single-carrier physical layer. It focused on a Time Division Multiplexed (TDM) MAC layer. Some amendments to this standard were made to fix issues that arose; the result was the mobility version. The first certified product for that version was announced in January 2006. Called the MAN protocol, it not only provides a wireless substitute for cable, DSL, or T1 level services for broadband access, but also works as a backhaul for 802.11 hotspots for other wireless technology [5], [6], [14].

2.1.1.1 WiMAX—HOW DOES IT WORK?

The WiMAX infrastructure, like any other cellular network technology, most often consists of a single or multiple BS serving multiple SSs. Such an SS should have WiMAX-compatible chips like those in Personal Digital Assistants (PDAs) and mobile phones. WiMAX can serve SSs in various ranges with different data rates. An SS can achieve a data rate of 70 megabits per second (Mbps) and a range of 50 kilometers (km). For longer distances it can have a range of more than 50 km, though the data rate will be reduced to 1.5 Mbps [4], [7], [15].

The process of communication between the device and the BS is as follows:

1. The SS transmits the connection request and its requirements to the BS through the WiMAX chip.
2. SS transmits the call, using lower power and high encoding.
3. After waiting a while for a response from the BS, the SS takes the power to a higher level.
4. At some point, the BS and the SS agree on the level of power.

5. The SS, depending on the profile agreement, receives data slots through the BS, which is responsible for scheduling the data slots on the uplink direction.
6. Once the reception is completed, the SS transmits the data through the WiMAX chip to the WiMAX BS.
7. The BS transmits the data to the internet.

When there are multiple BSs, data can be transmitted between two of them through a microwave transmission link, and end WiMAX BSs can be connected to the IP backbone network with a wired connection, as Figure 1 shows. Connection between the subscriber and the WiMAX BS is point-to-multipoint, whereas if it is between two or more WiMAX BSs it could be in the form of point-to-point line of sight (LoS) [7].

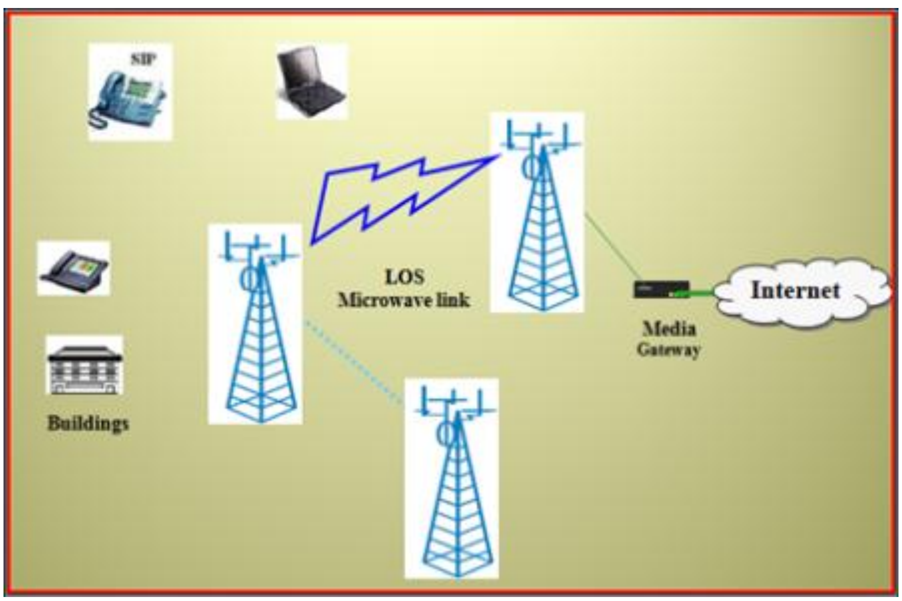

Figure 1:Point-to-multipoint Deployment Scenario with WiMAX Base Station

2.1.2 WIMAX STANDARDS

When WiMAX was established in 2001, there were several WiMAX standards, e.g., 802.16a, 802.16b, 802.16c, 802.16d, and 802.16e. Now, they have been grouped into two general standards, covering fixed and mobility networks:

- Fixed WiMAX: IEEE 802.16-2004
- Mobile WiMAX: IEEE 802.16-2005

In the beginning, WiMAX was specified to work in the range from 10 to 66 GHz. It was then updated to work on a range from 2 to 11 GHz, as specified in standard 802.16-2004.

A final amendment defines the new standard: 802.16.2005, covering operation in spectrum bands of 2.3 GHz, 2.5 GHz, 3.3 GHz, and 3.4 to 3.8 GHz . The names, fixed WiMAX and mobile WiMAX, are not the WiMAX standard names, which are the numbers IEEE 802.16-2004 and IEEE 802.16-2005, respectively. But they are the general terms commonly used in the market. The paragraphs below describe each of those standards and its properties [4], [9], [15].

2.1.2.1 IEEE 802.16-2004

When the first WiMAX was introduced, it was published under the standard known as 802.16a. Its limitation was the support of no–line-of-sight (NLOS). This standard was later updated to 802.16-2004 (also known as 802.16d), supporting fixed NLOS wireless internet service. IEEE, apart from cable, DSL, and T1 technologies, was looking for a standard that had the capability of providing stationary wireless with a higher data rate. With a higher data rate comes fixed WiMAX, a better alternative for cable, DSL, and T1 [7], [9].

The PHY layer of this version uses adaptive modulation based on Orthogonal Frequency Multiplexing (OFDM), which enables it to serve many users in a time division manner in a round robin fashion. This adaptive modulation is used to create the highest data rate for a certain link class. Other features of the 802.16-2004 standards are:

- It has provided NLOS broadband services for fixed users.
- It supports various modulations.
- It supports advance antennas.
- It provides for easy use of point-to-multipoint topology.
- It supports various scheduling profiles that in turn give lower latency for delay-sensitive services.
- It duplexes the frames in both Time Division Duplex (TDD) and Frequency Division Duplex (FDD) [15], [16].

2.1.2.2 IEEE 802.16-2005

A recent amendment to the previous standard established an improved standard, IEEE 802.16-2055, formally known as 802.16e or Mobile WiMAX. This technology is far more complex than the one defined in the 802.16.2004 standard. It enables mobile and fixed broadband networks to be connected through a common wide-area broadband radio access technology and flexible network architecture. It has the following characteristics:

- Union infrastructure for both fixed and mobile access.
- Support of the adaptive antenna system (AAS), which improves coverage rate.
- Deployment of a fast Fourier transform (FTT) algorithm, providing resistance to multipath interference.
- Support for a roaming and handover optimization scheme that supports real-time VoIP applications without compromising service [5], [7], [13], [14].

2.1.3 WIMAX PHYSICAL LAYER

The basic principle of the WiMAX physical layer is that it sets up a connection between the communicating parties and transmits a call as a bit sequence. Also, it handles the type modulation and demodulation, and provides the transmission power on this layer. This layer in WiMAX can transmit by either of two methods: OFDM, and orthogonal frequency division access (OFDMA). Both methods have frequency bands below 11 GHz and can be used with TDD or FDD. The central concern of this thesis is OFDM with TDD. Below is a brief description of this method [4], [14].

2.1.4 TIME AND FREQUENCY DOMAIN OFDM

One of the WiMAX physical layer methods is OFDM. This multiplexing method enables data to be distributed over one of a number of small subcarriers. Each of these small subcarriers uses a different frequency in order to resist inter-channel interference and avoid crosstalk in signal transmission. Hence, the entire channel bandwidth is divided into multiple subchannels, so that incoming information can be distributed on subchannels with different frequencies [3], [7], [14].

This thesis focuses on fixed WiMAX, in which the OFDM physical layer is based on a 256 FFT and the number of subcarriers is fixed at 256. Of those 256 subcarriers, 192 are used for carrying data, eight are used for subchannel estimation, and the remaining are used as guard-band subcarriers, as Figure 2 shows. In general, the spacing between subcarriers is inversely proportional to the number of subcarriers. The figure also indicates that the bandwidth varies, as does the subcarrier spacing [4–7].

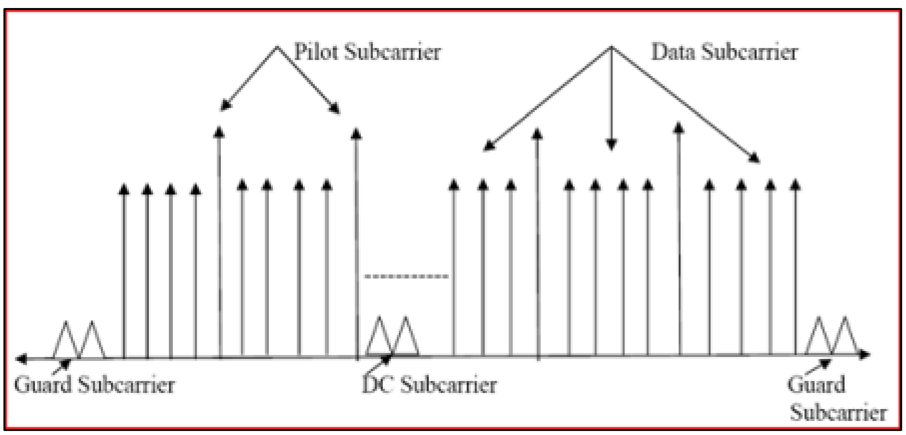

Figure 2: OFDM Symbol Representation in the Frequency Domain [4]

Fixed WiMAX can use either of two duplex methods, TDD or FDD. Each of these methods is used to separate uplink (UL) and downlink (DL) signals. In TDD, both UL and DL traffic are transmitted on a single frequency band, with different timing. More precisely, TDD separates WiMAX data traffic into frames of fixed length. Each frame is divided into two parts, forward and reverse transmission. This enables the transmitter and the receiver to share the same radio frequency, while using part of the bandwidth. TDD is the most widely used method in WiMAX: it is the best candidate for asymmetric traffic. Also, TDD systems are less complex than FDD in terms of diplexers, and therefore less expensive. The work reported in this thesis is concentrated on TDD as a duplexing method [4], [8], [14].

The other, most important feature of fixed WiMAX is that it supports a flexible bandwidth allocation, which enables it to compete with other wireless standards. It makes possible the granular adjustment of channels as needed for users' requirements. The channel bandwidth can vary from 1.25 MHz to 20 MHz [4], [8], [14].

2.1.5 WIMAX MODULATION TECHNOLOGY AND FRAME STRUCTURE

Modulation is a process by which carrier waves are used to carry a digital signal or message. Amplitude shift keying (ASK), frequency shift keying (FSK), and phase shift keying (PSK) are the most commonly used types of modulation. The choice of one of these modulations depends on the technology, the space, and the transmitted power. In WiMAX, a combination of combined ASK and PSK is formed; it is known as quadrature amplitude modulation (QAM) [6], [8].

The WiMAX BS, working from site conditions, distance, channel interference, the user's need, and the QoS desired, dynamically selects a modulation scheme. High-order modulation, such as 64 QAM, provides higher throughput but a lower coverage range.

Low-order modulation, such as 16 QAM, provides lower throughput but a higher range from the same BS. The various modulation and coding schemes used in fixed WiMAX standards are listed in Table 1 [4–7].

Table 1: PHY Mode Modulation Schemes and Signal-to-Noise Ratios (SNR)

Modulation	Coding Rate	Average Received SNR (dB)
BPSK	½	6.4
QPSK	½	9.4
QPSK	¾	11.2
16 QAM	½	16.4
16 QAM	¾	18.2
64 QAM	½	22.7
64 QAM	¾	24.4

As was explained before, WiMAX frame is split into two subframes by the TDD method, as Figure 3 shows. The upper part, known as the downlink (DL) subframe, contains bits coming from the BS and going towards the SS; the uplink (UL) subframe is the place where SS sends uplink data to BS. The DL subframe begins with the preamble, which is used for frequency synchronization. After this comes a Frame Control Header (FCH), which carries configuration information about the coming messages, such as the MAP message length, the modulation, and the coding scheme. The FCH is followed by uplink and downlink MAP messages (DL-MAP and UL-MAP). These messages map the allocated data regions for each SS within the frame. The rest of the frame contains the data burst, broken down for each individual SS in accordance with the DL-MAP [2], [17].

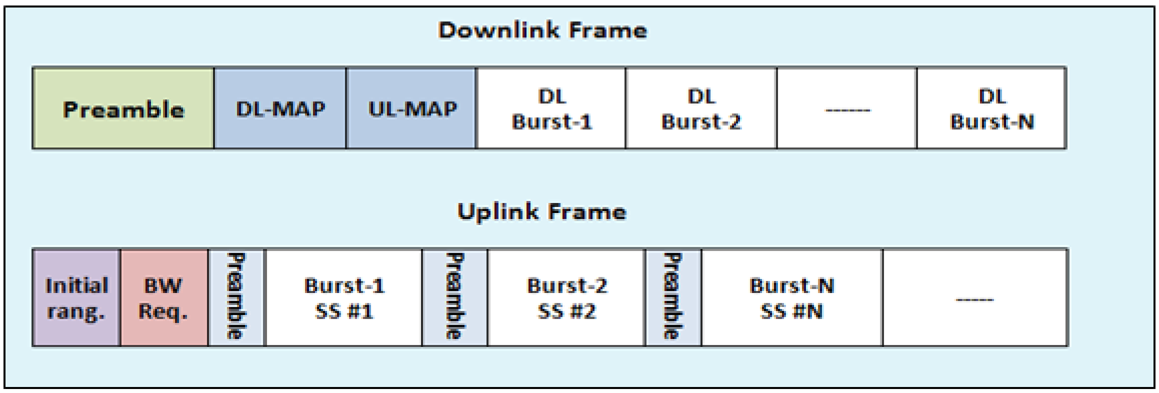

Figure 3: Frame Structure for a TDD System [2]

The second part of each frame is the UL subframe. This begins with a contention interval, which is used for the initial ranging and bandwidth allocation. This is followed by several UL bursts, whose assignments are based on the DL-MAP at the beginning of the DL subframe [2], [7], [14].

2.1.6 WIMAX-MAC LAYER

As was seen in the previous chapter, the physical layer is responsible for transmitting data as a binary communication through the physical medium. The higher layers of the system transmit the data as packets, not bits. So there is a requirement for an interface between the physical layer and higher layers in the open systems interconnection model for the open systems interconnection (OSI) layer.

The MAC layer exists to provide this interface. The WiMAX-MAC layer takes the data packets from the upper layer of the network in the form of frames called MAC service data units (MSDUs). The MAC layer then organizes them into MAC protocol data units (MPDUs) in order to send them to the physical layer. The MAC layer parts of WiMAX are shown in Figure 4. They are divided into three distinct components: 1) the service-specific convergence sublayer (CS), 2) the common part sublayer, and 3) the security sublayer. The CS and its different functions, the MAC common-part sublayer, the construction of MAC PDUs, the bandwidth allocation process, QoS control, and the network entry procedures are described in more detail in the subsequent sections. The function of the security sublayer is outside the scope of this thesis and therefore will not be described [5–8].

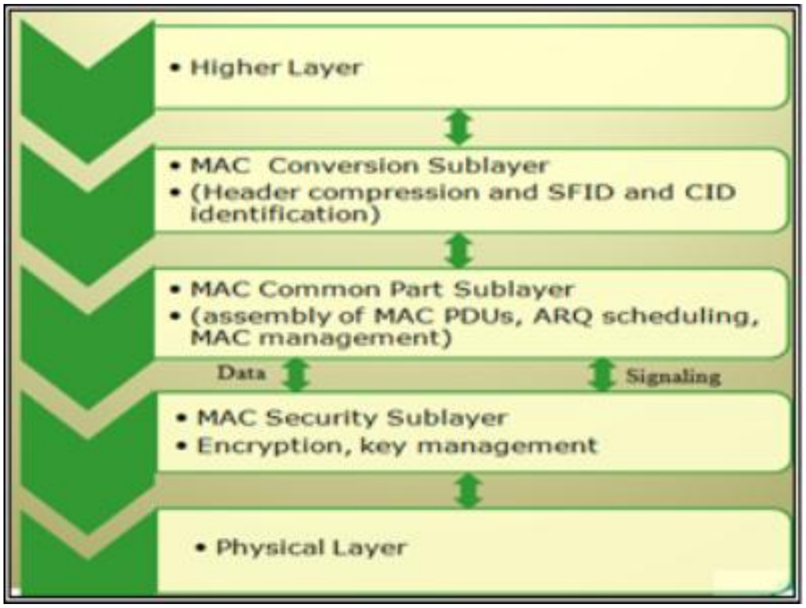

Figure 4: The Three Main Components of the MAC Layer

2.1.6.1 Convergence Sublayer (CS)

One of the most essential features of WiMAX is its ability to provide an interface to another communication protocol, such as Ethernet or IP. For this, WiMAX-MAC layer has a sublayer known as the convergence layer. Well-known protocols such as IPv4, IPv6, Ethernet, and 802.1q are some of the higher-layer protocols supported in WiMAX with the help of this layer. This layer not only provides an interface with higher protocols, but also reduces the overhead for the higher layer by removing the MSDU header. This layer also keeps a map between the IP addresses of the MSDUs and the identity of the PHY and MAC connections. This step is important, because higher-layer addresses, such as IP addresses, are not visible at the MAC and PHY layers. Since the WiMAX-MAC layer is also connection-oriented, it identifies a logical connection between the BS and the SS by a unidirectional connection identifier (CID), which can be considered a temporary, dynamic layer 2 address assigned by the BS to identify a unidirectional connection between the peer MAC/PHY entities. The MAC layer is also used to carry data and control traffic [4], [5], [13].

2.1.6.2 MAC Common Part Sublayer

WiMAX has robust QOS capabilities. To address the QoS parameters with even higher data rates, the MAC common part sublayer offers variable-length MPDUs; that is, it aggregates multiple MPDUs and sends them over the air interface in a single burst, thus evenly reducing overheads for the PHY layer. This sublayer also provides QoS control for other functions, such as the fragmentation of SDUs into the MAC PDUs and the transmission of MAC PDUs. Figure 5 depicts how, with the help of this sublayer, multiple MSDUs can be carried on a single MAC PDU, or a single MSDU can be fragmented to be carried over multiple MAC PDUs [5], [8].

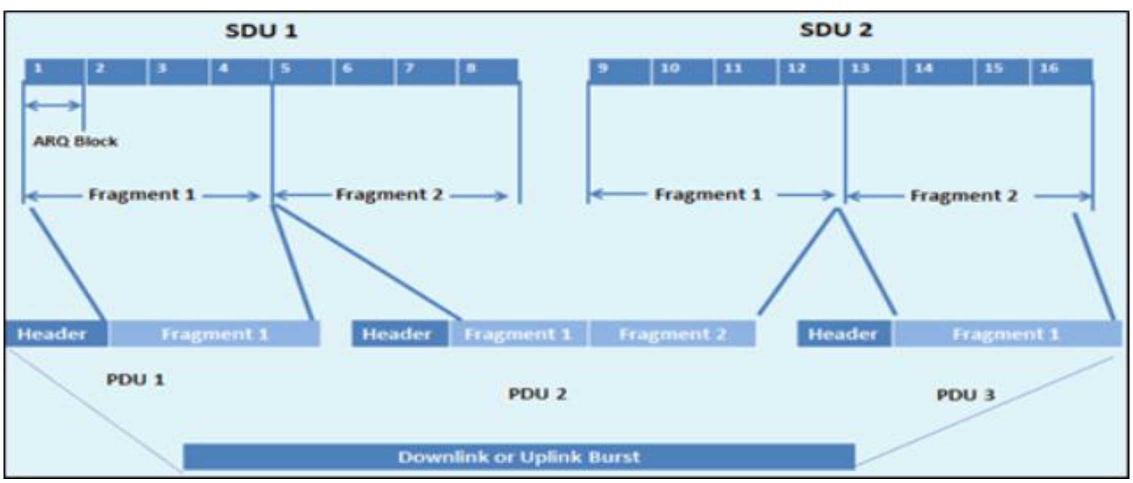

Figure 5: Segmentation and Concentration of SDUs into MAC PDUs [5]

To communicate on the network, an SS needs to start two-way communication with the desired BS by executing the network entry procedure. After acquiring the needed information from broadcast messages, the SSs do as follows:

- Synchronizing the DL channels:

The SS searches for a channel in the given frequency list. When operating over a license band, an SS is usually configured to use a explicit BS within a given set of operational parameters.

- Initial ranging:

When an SS synchronizes with the DL channel, it begins the ranging process by sending a ranging request, using minimum transmission power. It increases the power until it receives a ranging response from the BS, giving the correct power requirement and the correct timing for synchronization.

- Negotiating capabilities:

The SS sends the BS a capability request message, asking the BS to negotiate the properties of the physical layer. BS accepts or rejects the SS, basing its decision on supported modulation levels, coding schemes and rates, and duplexing methods.

- Exchanging authentication messages

Once the capability negotiation is completed, SS is authenticated with BS and provided with a security key to enable the ciphering of data.

- Registration

When authentication has been completed successfully, the SS registers with the network. In the registration, the following information is exchanged: the IP versions supported, the automatic repeat request (ARQ) parameters supported, the cyclic redundancy check (CRC) supported, and flow control.

- IP connectivity

At this stage, the SS begins the process of getting the required IP and establishing IP connectivity via dynamic host configuration protocol (DHCP) (IETF RFC 2131).

Once the network entry process is completed, the SS is ready to establish one or more service flows to transfer data to the BS. It is worth mentioning here that for conditional service flows, the creation of a connection is initiated by the BS. Then the BS sends a dynamic service flow addition, requesting a message to the SS. Upon receiving this message, the SS sends a response to confirm the creation of the connection. The SS creates a connection for non-conditional service flows by sending the BS a message requesting a dynamic service flow addition. The BS responds with a confirmation of service [3], [5], [10], [17].

Thus the MAC layer is responsible for determining the method by which the channel will be accessed. By providing various mechanisms for controlling access to the medium, WiMAX is successful where its predecessors, such as WiFi, were not. In WiFi, a medium or a channel is accessed in a random way. The station closest to the access point (AP) manages to affects the performance of further stations, or those that cannot even see the AP.

As was explained above, and unlike wireless LAN, WiMAX supports different methods for accessing a channel. Wireless LAN uses a contention-based MAC layer, whereas WiMAX has a request-grant access method. In WiMAX, it is the responsibility of the BS to manage and control bandwidth assignments.

Once the SS receives the allocated bandwidth, usually as an aggregated bandwidth, it distributes the accumulated bandwidth among the multiple service connections it uses. From the BS side, on the basis of the required QoS and the available bandwidth, the downlink is allocated without involving the SS. On the uplink side, bandwidth is allocated as appropriate for the QoS that the SS has requested and agreed on with the BS. Also, the BS periodically sends unicast or multicast messages among multiple SSs by a process called "polling." The time slot allocated to an SS remains reserved regardless of whether SS has something to send through the assigned channel [4], [7], [13], [17].

2.1.7 SERVICE CLASSES AND QOS

WiMAX's QoS is achieved by defining transmission ordering and scheduling on the air interface. This can be achieved by using a mechanism to handle data transmission on a connection. Each connection is assigned mapped to a single data service, and each data service is associated with a set of QoS parameters describing its behavior and requirements. According to standard IEEE 802.16-2004, four classes of scheduling have been defined on the UL. The following is a brief description of those classes; more details about the BE class will be given, because it is the focal point of this thesis [6], [14], [18].

- UGS

The UGS is supports real-time applications which generate fixed-size data packets at periodic intervals, such as T1/E1 and VoIP, without silence suppression.

- rtPS

Unlike UGS, rtPS is supports applications which generate variable-sized data packets at periodic intervals, such as a moving picture experts group (MPEG) video.

- nrtPS

The nrtPS is supports applications which generate variable-sized data packets and are delay-tolerant, such as FTP.

- BE

BE service is designed to provision data streams for services which doesn't need minimum service level and can be handled on an availability basis. BE traffic has the lowest priority; that is, only the bandwidth (BW) that remains after all higher-priority flows have been serviced will be allocated to it. WiMAX uses a contention-based method with exponential back-off to manage multiple access for BE traffic. BW requests are sent to the BS using contention. This adds to the overhead in the form of contention slots and latency due to collision and back-off. To get a minimum level of successful contentions, the BS is required to allocate the needed number of contention slots. In the simulation part of this thesis we will see what affects the performance [2], [7], [15], [18].

2.1.8 ANALYSIS OF BEST EFFORT SERVICE CLASS IN WIMAX

This section describes in detail how the BE service class operates. The operation of the BE class is includes the following processes:

1. Transmitting a bandwidth request (BW-REQ)
2. Contention resolution in case of collision of the transmitted BW-REQ with other requests from other SS.

These two processes are described in detail below.

2.1.8.1 Requesting Bandwidth

On the SS side, when a single packet or multiple packets are to be transmitted, they are put into the appropriate queue in an order that depends on their IP precedence. The SS then waits until it receives the broadcast part of the frame, which contains, among other information, the UL-MAP. This part of the frame contains an element called REQ Region-Full, which is signified by an uplink interface use code of 4 (UIUC = 4). In searching for this element, all the SSs scan the frame that consists of the contention slot, as depicted in Figure 6. The width of this element (in time), which is variable, is communicated to all SSs through an uplink channel descriptor (UCD) message by the BS. In a random base, the SS picks a slot out of those given in the frame and transmits a bandwidth request protocol data unit (BW-REQ PDU). This unit, which contains only 6 bytes including the CID, uniquely identifies the SS, the UL flow, and the aggregated bandwidth (BW) required, in bytes. Because this is a BW request only, this message includes no payload. Any collisions happening at this period of asking for a BW, and any method for reducing the collisions, will improve link utilization and performance in general.

Another way to apply for BW from BSs is to use a BW piggyback request, which uses 6 bytes of data PDU. Because we are searching for a way to be fair to all SSs, this mechanism is not analyzed in this work [5].

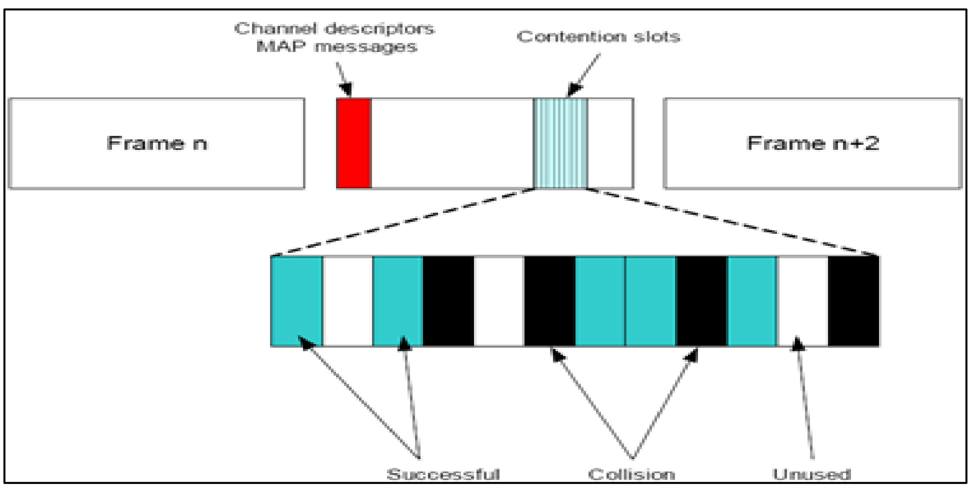

Figure 6: Contention Slots in a WiMAX Frame [5]

The question remaining is, how do the SSs know whether a request has succeeded? The following section answers this question [2], [5].

2.1.8.2 Resolving Contention

Looking at it from the SS side, if contention is successful, it is implicitly made clear later on by a BW message, containing the start time and duration of the allocation. It is now up to the SS to transmit its PDU within that period.

If the SS does not receive a reply within a certain time, called T16, it considers the transmission lost, because of either collision or rejection by BS. The SS then increases its back-off window by a factor of two, provided that the result is less than the maximum back-off. The SS randomly selects a number within its new back-off window and repeats this process so long as no slot is guaranteed. This process continues until the SS reaches the maximum number of retries that is, “Request Retries” for bandwidth requests, or “Contention Ranging Retries” for initial ranging. If time passes without a successful attempt, the PDU in the SS side is discarded and a new PDU starts the process again. The maximum number of retries is independent of the initial and maximum back-off windows that are specified by the SS and set by the BS. If the SS, after requesting bandwidth, receives a unicast request or a data grant burst type at any time while waiting for this CID, it stops the contention resolution process and uses an explicit transmission opportunity.

For simplicity, ease of distribution, fairness, and efficiency, the BS may choose to set up the request back-off start and request back-off end timing and communicate it to all SSs. The BS may also make the request back-off start and the request back-off end identical and frequently update these values so that all SSs are using the same, hopefully optimal, back-off window [5], [6], [12].

In summary, IEEE 802.16-2004 specifies the Wireless MAN air interface for wireless MANs. It defines the MAC layer and the physical layer specifications of a fixed point-to-multipoint broadband wireless access system. It is actually a collection of standards for point-to-point, point-to-multipoint, and mesh architectures, and provides complete PHY and MAC layer specifications for all. The part of the standard that is considered in this thesis is the Wireless MAN OFDM specification, which covers the fixed WiMAX.

Because the mechanism for providing QoS in the design of the WiMAX standard, QoS is very well supported by WiMAX. There are four native QoS classes to support real-time VoIP and video traffic, interactive traffic, bulk data transfer, and burst low priority traffic. This thesis explores the ability of the lowest-priority class to serve VoIP traffic.

2.2 VOICE-OVER IP (VOIP)

VoIP is the technology used for transmitting voice conversations over IP networks. It converts and compresses speech from analog form into digital data packets. This transmission process, unlike the one in the PSTN, does not establish a circuit between the two parties, but it does utilize an IP network in managing voice packets over the data network. The main requirements are that transmission must be achieved without any loss of voice quality, and must comply with the specifications set forth by the International

Telecommunications Union (ITU). As long as it meets these requirements, any kind of IP network is suitable, such as internet, Ethernet, or wireless network. WiMAX is one of several that has the capability to transmit VoIP packets with a high QoS as is required for such applications [6].

2.2.1 TYPES OF VOIP

VoIP can be transmitted and received by various devices, similar or different at the two ends. These devices can be computer, IP phone, or PSTN phone. Each of these devices has its own unique requirements. This section discusses the most common combinations: computer to computer, IP phone to IP phone, and computer to phone.

2.2.1.1 Computer to Computer

This is one of the easiest ways of communicating without paying for the call. Software applications such as MSN or Skype are a popular way of communicating. Distance is not an issue. All MSN and Skype users require is a headset and a DSL connection or WiMAX, which guarantees the minimum QoS for VoIP.

2.2.1.2 IP Phones

These are new phone devices for VoIP communication. They work on internet connections instead of PSTN. Unlike ordinary phones, these devices use RJ-45 connectors instead of RJ-11. You can also install a soft phone in the computer. A session initiation protocol (SIP) to SIP call can be made if both parties have the correct software installed, or SIP hardware. The call will be free if either one of them is used.

2.2.1.3 Computer to Phone

Once communication from a computer to a landline phone begins, the person who receives the call begin paying when the call is put through. If VoIP software is installed on your computer, the procedure for such a call is the same as for a computer-to-computer call. These calls are cheaper than landline-to-landline calls [6–9].

2.2.2 VOIP SYSTEM

As Figure 7 shows, the VoIP network consists of multiple components:

- Gatekeeper

This equipment, also known as a call manager, is optional for a VoIP network. Its function can be summarized by saying that a routing manager and a central manager coordinate all calls in an H.323 IP telephony environment. This device is practical also in managing VoIP call connections, including end-terminals, gateways, and multipoint control units (MCUs). Therefore, such equipment can improve security and the QoS [19].

- VoIP Gateway

This equipment is mandatory for VoIP communication. It handles external calls and converts VoIP calls to and from conventional PSTN lines. It also works as a connection point between a conventional private branch exchange system (PBX) and an IP network.

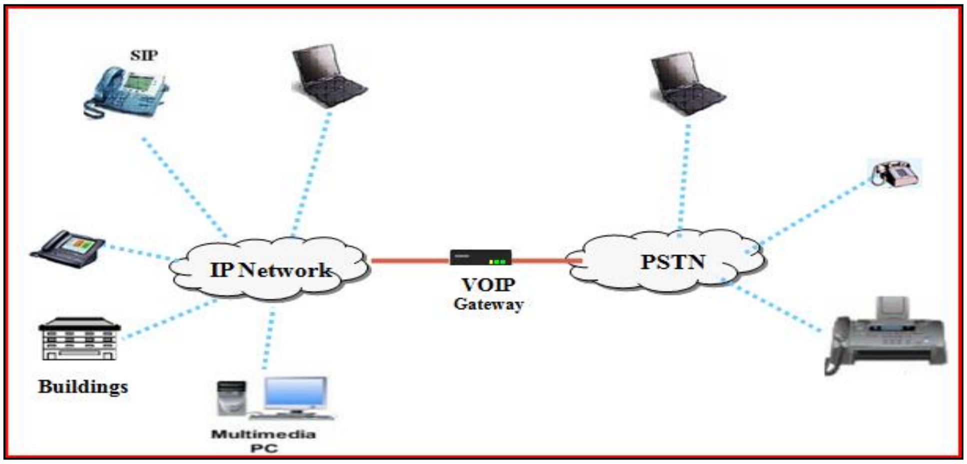

Figure 7: Typical VoIP Network Topology

- VoIP Client

This is the end user's equipment, which is an important piece of equipment in the topology. "It can be an IP Phone, a multimedia PC, or a VoIP-enabled workstation running VoIP software" [6].

2.2.3 VOIP ENCODING AND COMPRESSION ALGORITHMS

VoIP operates differently from conventional PSTN in many ways. One difference is the signaling, wherein VoIP signals are converted to digital form before transmitting and are completed by an analog-to-digital converter. The same thing, in reverse, is applied at the receiver end, where the signal needs to be converted from digital to analog. This can be done by any of several many transitional devices, as Figure 8 shows.

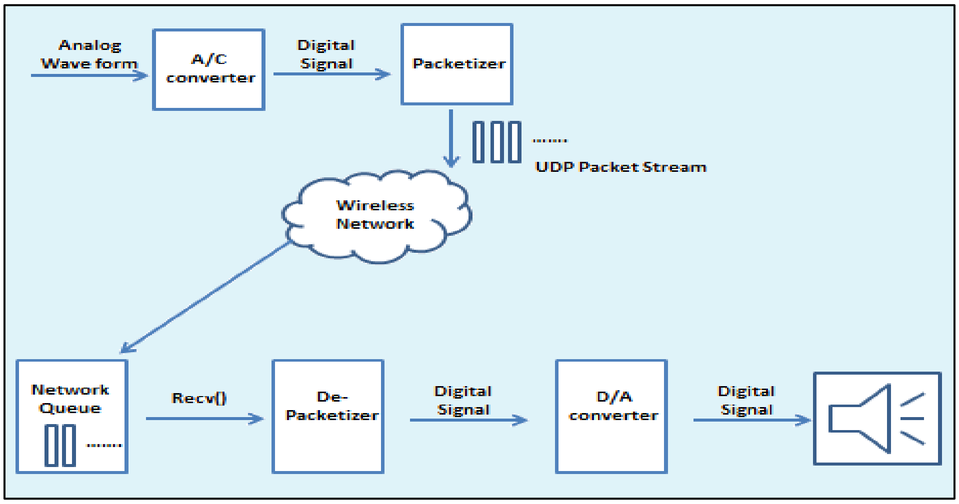

Figure 8: VoIP Process [7]

To save the network bandwidth and utilize the network perfectly, systems usually use VoIP codecs. Various audio codecs for voice applications are available . A few that have been proved to be simple and efficient are G.711, G.723, and G.729 [6], [7].

G.711 (64 kb/s) is the simplest one. It uses a pulse code modulation (PCM) and has a compression process. The G.723 and the G.729 do higher compression and have slower data rates, 8 bytes for the G.729 and 5.3 and 6.4 kb for the G.723.1. For multimedia applications, the G.723 encoder system was established and approved by the ITU.

Compression may be a good way to reserve bandwidth, but it leads to complexity and requires somewhat longer to encode and decode, which will result in lowering the VoIP quality, as in G.723 and G.729 [6–9], [13].

When bandwidth is not a major issue, as in WiMAX, and QoS is a major concern, the pinpoint should be changed. As in this case, and to study of the QoS, G.711 and G.729

are considered as the default choice because they give the best voice quality. The properties of the main voice codecs are summarized in Table 2.

Table 2: Voice Codecs and Their Properties [6]

Codec	Bit Rate	Payload	Packets Per Second (pps)	Quality	Ethernet Bandwidth	Sample period
G.711	64kbps	80 bytes	25 pps	Excellent	95.2 kbps	10 ms
G.729	8 kbps	20 bytes	50 pps	Good	39.2 kbps	10 ms
G.723.1	6.3 kbps	24 bytes	34 pps	Good	27.2 kbps	30 ms

2.2.4 VOICE ACTIVITY DETECTION (VAD) OR SILENCE COMPRESSION

Voice conversations consist of spurts of talk and spurts of silence, because we cannot talk and listen at the same time. Silent periods add up to about 60–65% of the total conversation. A method for suppressing packet transmissions during these periods is needed. VAD is such a method. Instead of sending VoIP packets of silence, the available bandwidth is allocated to other flows. During silence suppression, VAD periodically provides comfort noise generation (CNG) to prevent users from mistaking a time of silence for a sign that the call has been disconnected. This can be achieved by generating white noise so that the calls seem to the users to be connected and fewer packets are transmitted [7].

2.2.5 VOIP QOS

QoS is a key feature in any voice transmission, whether IP-based or regular. Currently, many IP services run without guaranteed QoS from network operators. This cannot be the case if the number of network users grows. For example, in many organizations IP-based voice and video services do not usually have great QoS support. This is because the local area networks (LANs) usually do not provide enough bandwidth for such applications. It is hard to guarantee QoS for real-time multimedia services across internet networks. Many aspects affect voice quality such as choice of codec, delay, packet loss, and jitter. The following subsections discuss each of these factors [6], [7], [9].

- Delay

To provide high-quality VoIP service, in accordance with ITU, a VoIP packet should not be delayed more than 150 ms.. Moreover, environments need to be echo-free, to ensure users' satisfaction. The major components of the delay in voice communication are: 1) delay triggered by the used codec, 2) delay form the queuing algorithm on the transmitting equipment, and 3) other factors, such as network states, the VoIP device being used, weather, etc. To get the QoS required for VoIP traffic, delays in voice traffic must be minimized. One way to do so is to carefully choose the correct codec and queuing algorithms. The ITU states that the maximum acceptable end-to-end delay is 150 ms. Di [6] states that a delay of up to 200 ms is allowed. For planning, a one-way end-to-end delay between 150 ms and 400 ms is considered acceptable. However, this study considers 200 ms the maximum acceptable one-way end-to-end delay [6], [7].

- Jitter

Because of the nature of IP networks, consecutive packets may travel over different paths. Also, these networks are subject to network congestion and packet loss. As a result, the delay between consecutive packets may vary; the variation may cause the phenomenon known as jitter. To keep jitter at a tolerable value, less than 50 ms, a jitter buffer is used [6].

- Packet loss

This is the second most important factor for VoIP's QoS. Its importance is equal to that of the delay factor. When the network is overloaded, some packets are lost. For VoIP applications, any delay in packet transmission will degrade the QoS, which is very time-sensitive. Because of this, any packet loss can significantly affect VoIP's QoS. Although 1% is considered a tolerable level of packet loss, an early study shows that voice quality is reasonable when that level is within 1 to 3% [2], [6].

Quality of service has to be assigned the highest priority in a VoIP system. To achieve the needed quality, jitter and packet loss have to be controlled, so that VoIP can be transmitted even through low-speed connections. Implementing prioritized mechanisms can also contribute to QoS. We conclude that VoIP quality is the measure of the success of VoIP service in any network.

2.2.6 VOIP QUALITY MEASUREMENT

In an environment of various applications, a system can be evaluated by evaluating the performance of each application, using specific metrics for each one. This makes possible an objective approach to measuring the user-perceived quality (UPQ) achieved for a specific application. Today's telecommunication networks use huge sets of voice services that require many transmission systems. This creates a need to evaluate the QoS provided for each system. One way to evaluate QoS is to compare the quality of received voice to that of human communication [4], [9].

MOS, supplied by ITU, is the first subjective metric standard for evaluating the voice quality produced by a system. Further attempts to standardize objective metrics led to the E-model, now considered one of the main methods for measuring VoIP UPQ. Details on these two methods are presented in the following paragraphs [12], [20].

- Mean Opinion Score (MOS)

This standard, established by ITU in 1996, has been defined as a subjective standard for use in determining the average opinion of voice quality. Results obtained by applying this standard are applicable to any kind of degradation: loss, circuit noise, transmission errors, environmental noise, echo, distortion due to encoding, etc. Table 3 describes the MOS levels, which can range from 5 (excellent) down to 1 (bad). VoIP is considered good if the MOS value of the transmitted voice traffic is 4 or higher, comparable to the quality achieved with PSTN. By this standard, if the MOS value degrades to 3.6 or lower, many users will not be satisfied with the call quality. MOS presents its own difficulties. It depends on users' evaluations and opinions, and can be described as slow, time-consuming, and expensive. Also its usefulness is limited, because it can be used only on a small scale [18].

Table 3: Mean Opinion Score Scale[18]

MOS	Quality Rating
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

- E-Model

After the MOS was amended, the E-model was established in 2000. Since then it has gone through several updates, the latest in 2005. This standard is a computational method to quantify VoIP quality. It measures VoIP quality and takes into consideration all impairment elements that affect voice quality, for instance delay, packet loss, jitter, and codec behavior. The result of this computation, called the R factor, is scaled from zero to 100. Then MOS can be estimated by mapping it to the obtained R factor value. The general formula for the E-model is as follows:

$$R = R_0 - I_s - I_d - I_{e-eff} + A \quad (1)$$

Where:

R_0 = basic signal-to-noise ratio

I_s = Simultaneous impairment factors

I_d = delay impairment factor

I_{e-eff} = packet-loss-dependent effective impairment factor

A = advantage factor (system specific)

Based on recommended values, the rating R can be defined by:

$$R = 94.2 - I_d - I_e \tag{2}$$

Where I_d has the following form:

$$Id = 0.024d + 0.11(d - 177.3)H(d - 177.3) \tag{3}$$

Where d is the one-way delay in milliseconds, and $H(x)$ is the heavy side or step function, where

$$H(x) = \begin{cases} X > 0, & 0 \\ otherwise, & 1 \end{cases} \tag{4}$$

I_e is codec dependent

$$I_e = a + b\ ln\left(1 + \frac{cP}{100}\right) \tag{5}$$

Where P is the packet loss rate in percentages, and a, b, and c are codec-dependent values. The values of a, b, and c for different codecs are given in Table 4 [20].

Table 4: Parameters for Various Codec

Parameter	***a***	***b***	***c***
Bit rate (kb/s)	64	8	6.3
Frame size (ms)	20	10	30
G.711	0	10	15
G.729(10 ms)	30	25.21	36.59
G.723.1	15	15	6

2.2.7 MAPPING BETWEEN MOS AND E-MODEL

The E-model can estimate MOS value by using the R-value obtained. The following equation describes the relation between R and MOS values [17], [21], [22]:

$$MOS = \begin{cases} R < 0, \quad 0 \\ 0 < R < 100, \quad 1 + (0.035 * R) + (R(R - 60) * (100 - R) * 7 * 10 - 6) \\ R \geq 100, 4.5 \end{cases} \tag{6}$$

In Table 5 the R factor values from the E-model are shown in the first column and the mapped MOS values in the second column. The third and fourth columns show the speech quality category and the expected level of satisfaction, respectively. Figure 9 is a graph of the dependence of MOS on R [18], [22].

The OPNET simulator, used for the simulation study for this thesis, uses this mapping method to calculate the MOS values automatically, and reports this value for all VoIP traffic.

Table 5: Relationship between Rating Factor R, MOS Value, and User Satisfaction

R-value: lower limit	MOS	Speech Quality Category	User Satisfaction
90	4.34	Best	Very satisfied
80	4.03	High	Satisfied
70	3.6	Medium	Some users dissatisfied
60	3.1	Low	Many users dissatisfied
50	2.58	Poor	Nearly all users dissatisfied

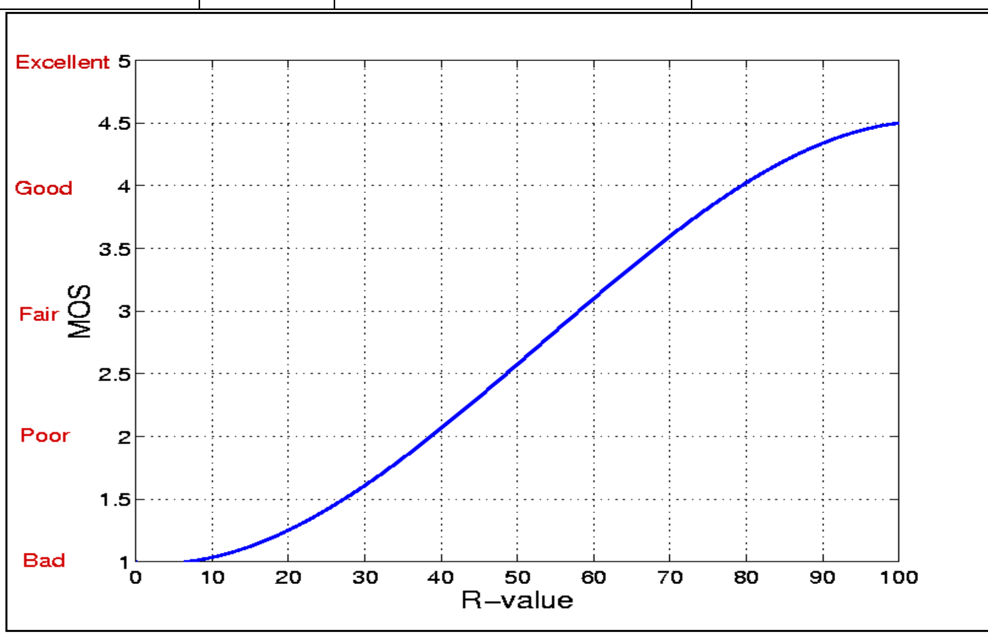

Figure 9: MOS vs. rating factor R [23]

CHAPTER 3

RELATED WORK

3.1 VOIP OVER WIMAX

A significant amount of work on or related to WiMAX and VoIP has already been done, but it mostly focuses on the physical layer and the scheduling framework in the MAC protocol.

The study closest to my work is the one done by Matos *et al*. [1]. Their testbed study shows the capability of WiMAX BE service class in supporting 50 bidirectional VoIP calls and 3 IP Television (IPTV) sessions. The MOS value of VoIP sessions is not evaluated in that study, because it shows delay only and packet loss values in general. In addition, we cannot compare that work with ours, because in that study the VoIP calls are from one SS only. In our case, we were trying to simulate reality by initiating VoIP calls from various SSs at different locations, using different coding method as mentioned before.

Pentikousis *et al.* [11] study a set of tests that considers video streaming and VoIP in a WiMAX PMP testbed. But that study considers only the throughput part of the network;

it does not evaluate other parts, such as delay and packet loss. Their study is the first publicly available assessment of VoIP and video streaming over a WiMAX , where only two SSs are used in the same cell. It also reports the performance of FTP and VoIP over a WiMAX testbed in Italy. Their results indicate acceptable MOS for VoIP in a cell within 2 km, but do not show the upper limit to the number of VoIP sessions the network can support. Therefore the results of this study are not comparable with what we are doing here.

Sengupta *et al.* [21] use a WiMAX testbed and the E-model to do a VoIP traffic evaluation. They evaluate codecs G.729.2 and G.723.1, mapping their data traffic to rtPS class in WiMAX. The results point to interesting conclusions. For fewer calls, G.729.2 provides fairly good quality, but after ten calls G.723.1 performs much better. Further, VoIP traffic performs better on the uplink than on the downlink. This is justified by the use of the piggyback bandwidth allocation mechanism. Pellegrini *et al.* in [20], using an experimental testbed, evaluate WiMAX support for VoIP. They show that for a computed index of voice quality, the BS capacity for high-quality voice calls ranged between 10, with a G.723.1 codec, and 17, with a G.729 codec.

In [18], the results of analyses of the performance of the scheduling algorithms recommended in IEEE 802.16e systems UGS and rtPS are presented. The analysis of resource utilization efficiency and VoIP capacity shows that the UGS and rtPS algorithms experience problems. First, uplink resources for the UGS algorithm are wasted. Second, the bandwidth request process in the rtPS algorithm that supports the VoIP traffic causes additional access delay and MAC overhead. In addition, for analysis of VoIP capacity, the study utilizes OPNET simulation and shows that the ertPS algorithm can support 21% and 35% more voice users than the UGS algorithm, and 35% more than the rtPS algorithm.

Hong [22] identify the problems associated with delivery of VoIP services in 802.16. They identify the absence of a MAC scheduling algorithm for supporting VoIP codecs with voice activity detection, the absence of an interface between the IEEE 802.16 service flow and network flow, and the absence of an efficient admission control algorithm.

The QoS of VoIP applications over the WiBro network was evaluated by Han *et al.* in an experimental study [18]. They measured and analyzed the characteristics of delay and throughput under fixed and mobile scenarios. Also, using the E-model, they evaluated the QoS of VoIP applications. Their results indicated that the achievable maximum throughputs are 5.3 Mbps in downlink and 2 Mbps in uplink, and that the quality achieved with VoIP is better than or at least as good as the quality achieved with PSTN.

For VoIP, we can relate this work with IEEE 802.11 measurement studies. A simulation study by Mohd *et al.* [17] shows that G.711 encoding is a better choice for VoIP than IEEE 802.11, because it provides the best QoS and a comparable number of sessions with other coding.

Moreover, Pentikousis *et al.* [24] carried out a study of VoIP call's QoS, over fixed WiMAX, with respect to the delay and loss of packets. They observe that loss is more sensitive than delay; hence, they compromise the delay performance within acceptable limits in order to achieve a lower packet loss rate. Through a combination of methods like forward error correction, automatic repeat request, MPDU aggregation, and mini-slot allocation, they reach a balance between the desired delay and loss. This study shows us that the retransmissions, aggregation, and variable-length MPDUs are effective and increase the R-score and the MOS by about 40%.

3.2 BEST EFFORT

In recent years, many papers have presented performance analyses of the WiMAX BE class. Those papers range from studying scheduling algorithms to proposing new algorithms. Most of them focus on the performance of bandwidth request mechanisms. Pentikousis *et al.* [24] study the influence of the contention slots number in delivering the bandwidth requests and average access delay, but they do not investigate the delay incurred by a data packet after a request is successfully transmitted. Kim *et al.* study the effect of multiple bandwidth schemes on WiMAX BE performance [10]. In the first schema, each SS submits a bandwidth request for every frame, to reduce delay. In the second schema, the number of bandwidth requests submitted is limited, in order to reduce collisions with other SSs. In the third schema, BS controls the bandwidth requests, it allocates slots for bandwidth request to SS based on its sending rate. Their simulation work shows that the third schema is a good solution for increasing network utilization.

An analytical model to evaluate the performance of BE service in a saturated IEEE 802.16 network is developed in Mohd *et al.* [17]. This study shows how the choice of the initial contention window affects the network throughput.

More recent studies, with current algorithms, have been described by Cicconetti *et al.* [14]. Their study confirms the existence of a deadlock in the bandwidth request algorithm of WiMAX. They propose a simple mechanism, called bandwidth request reiteration (BRR), for use by the SSs to prevent deadlock. Simulation results show that BRR achieves better performance than the timeout-based approach, as measured by the average delay of packets.

The study by Doha *et al.* [15] shows the need for dynamic control of the size of the reservation period, to improve the performance of reservation in MAC protocol. The study also shows how the size of the reservation period affects the performance of the MAC protocol of the BE traffic. It has been verified through computation and simulation that there is more improvement when the size of the reservation period is set to dynamic.

As for the BE service class, studies vary from using it to transmit un-sensitive data traffic such as web-browsing to carrying real-time traffic. Şekercioğlu *et al.* [2] study the performance of a WiMAX network with multiple scheduling. They mapped data traffic to the BE class, and VoIP and video traffic to the rtPS class. The results show that as the network saturates, there is a sharp rise in the delay of BE traffic but not of rtPS traffic.

In a similar study, Pellegrini [20] simulated VoIP performance over rtPS, and used BE to simulate TCP traffic as background traffic with no guarantee of bandwidth or delay.

Although the study is not focused on the performance of BE, it does show the use of BE in general.

Bestetti *et al*. [25] study the effect of the BE access parameters on network performance. The study simulates video traffic and FTP over the BE class. They conclude that there is a need for proper scheduling to support traffic differentiation and the priority of CBR traffic with respect to other traffic, such as FTP.

Cicconetti *et al*. [14] show the results of simulation for a mix of BE and multimedia flows. Their results show that for TCP BE traffic the average delay increases more sharply on the uplink than on the downlink mostly because of the bandwidth-request mechanism. On the other hand, delay and delay variation for rtPS multimedia traffic are constant until the SS population saturates.

The upper stream and the capability of BE in handling real-time traffic were studied through the OPNET simulation by Klein *et al*. [12]. They show that BE, using FDD, can support 22 VoIP calls and two FTP connections. They focused on end-to-end delay only; they did not assess VoIP QoS. They conclude that transmitting VoIP calls on BE connections is possible.

Hrudey *et al.* [26] simulated the capability of the BE class to handle video. Their simulation platform was OPNET, and the simulation was on a limited scale, with only four stations. It simulated a two-hour-long transmission of MPEG video in both directions. The uplink and downlink minimum sustainable data rates are 3 Mbps in uplink and 640 Kbps in downlink. The study demonstrates that packet loss in WiMAX is more comparable to that of the DSL.

In a real test environment, Han *et al.* [18] evaluated the QoS of VoIP applications over the WiBro network, which supports the BE only. The study shows that the maximum achievable throughputs are 5.3M bps in downlink and 2 Mbps in uplink. In general, the WiBro network supported VoIP calls even with mobility. This study supports our research.

Transmitting IPTV streaming and VoIP over BE on a fixed WiMAX is introduced by Pentikousis *et al.* [27]. This study found that WiMAX could efficiently cope with VoIP and IPTV traffic.

Klein *et al.* [12] show that the WiMAX had the capability to support multimedia applications; in particular, VoIP and video streaming are evaluated on a testbed. Although the results are focused mostly on the rtPs class, BE too was studied. For the over-provisioned case, the results show that BE can support multimedia applications, either video or voice, with good QoS. Those results are comparable to those for rtPS. For under-provisioned conditions, however, the performance was not acceptable.

Sengupta *et al.* [21] evaluated the performance of VoIP and audio/video streaming over a fixed WiMAX testbed in Oulu, Finland. It comprised one BS and two SSs. The results confirm that the testbed sustained 50 emulated bidirectional Speex-encoded VoIP calls and 5 simultaneous emulated IPTV streams.

Finally, a simulation study by Cicconetti *et al*. [14] showed the performance of data transmission in the BE class of WiMAX. The results show that the maximum number of SSs that can be served with an average delay of less than 150 ms is 35 SS.

To the best of my knowledge, the simulation we have here is one of the first to address both the capacity and the capability of the WiMAX BE class in supporting real-time traffic such as VoIP. This study is also unique in that it evaluates the quality of VoIP calls that can be provided.

CHAPTER 4

THE SIMULATION ENVIRONMENT

This study examines both the capabilities of UGS and the potential of the WiMAX BE service class in serving VoIP in the context of the IEEE 802.16-2004 MAC layer. More specifically, in the case of BE the effect of reservation slots, frame length, and VoIP traffic nature is studied. The experiments are executed under various network conditions, such as light or heavy load, by changing the number of SSs in the network.

To achieve more accurate results and estimates, a 95% confidence level based on 10 independent runs is calculated. As was mentioned before, the MAC layer is the focus of this research. The simulation models are more intense on this layer's parameters, and pick up the most appropriate combinations of those parameters, which in turn maximize the system capacity. The following sections summarize the programs used and the system parameters, and discuss the collected results.

4.1 SIMULATION TOOL

In simulation work, the choice of the right simulation tool for accuracy, speed, and simplicity is the key factor. Starting from that fact, the simulation program adopted to perform this part was carefully chosen after a search of the market. NS, Qualnet, OPNET are the best in this field. Others require more amendments for this task. The simulations were performed by two simulation programs: Qualnet and OPNET. The reasons are given in the following paragraphs.

4.1.1 SIMULATION EXPERIENCE WITH QUALNET

QualNet by Scalable Network Technologies was chosen from a few popular commercial and open-source simulators because it is easy to use and modify and to debug by use of Microsoft Visual C++. For the simulation work the latest version, qualnet-4.5, was used. This tool is attractive because it has a specific module for WiMAX [28].

For the initial simulations, Qualnet was used. A Constant Bit Rate (CBR) traffic simulated VoIP traffic from one SS to one BS. The results obtained contradicted the theory and the expectation of behavior. In one trial, for example, end-to-end delay improved as the number of terminals was increased. Also the results for packet loss were not realistic; a simulation of two SSs and one BS for 10 minutes showed a packet loss of 50% when the bandwidth of the uplink bandwidth was 12 Mbps

The developers of Qualnet confirmed that all of these issues were due to bugs in the system. Fixing such issues in a trial version takes time. Therefore an alternative simulation program was sought

4.1.2 OPNET

OPNET is an object-oriented simulation tool for planning and modulating, and for the performance analysis of the simulation of communication networks in general. This simulation program offers a great number of models for network elements, along with the ability to simulate real-life network configurations. These features enable the user to simulate a real-life network with all of its complications, simply and accurately. To make life easier, it has other features, such as a complete library of network protocols, a user-friendly Graphical User Interface (GUI), and easier data collection and results analysis. [29].

OPNET Modeler version 14.5 is a fully functional version with the WiMAX module, the focus of this study. It has the capability to simulate all of the functions of WiMAX networks, with all factors.

With regard to VoIP, this version of OPNET Modeler can simulate voice traffic. The obtained results, either statistical or graphical, include the values of jitter, end-to-end delay, packet loss, and MOS. This meets the needs of this work, since these values are the most important and will determine the success of the proposed study.

The performance of the OPNET simulator differs with different platforms. For this study the Windows version was used. The following summarizes the specifications of the simulation testbed.

4.2 TESTBED

- OPNET Modeler 14.5
- Hardware platform: HP
- Operating System: Windows Vista Service Pack 1
- Intel Core 2 Duo CPU 6420 @ 2.4 GHz
- 4.00GB of RAM

4.3 SYSTEM MODEL

The main criterion for building the system model was to find the best WiMAX attributes and encoding mechanism to support VoIP calls with an MOS value of 3.7 or greater. This value is considered the best voice quality, as was explained in the preceding section. Model work starts by choosing the parameters that maximize the capacity and performance of the WiMAX network while providing the intended MOS values.

4.4 PARAMETERS OF THE PHYSICAL LAYER

The physical parameters of the networks, an essential component of the equation, are outside the scope of this work. They were the same whether we used UGS, rtPs, or BE.

What came to mind when we referred to the physical parameters is the frequency band. This value is set to 5 GHZ, an unlicensed band. The bandwidth of the network that is assumed to be reasonable when compared to reality is set to 12 MHz for both uplink and downlink. In the WiMAX system, data is structured in frames. Frame length influences the delay; the longer the frame, the more it will delay the delivery of VoIP packets. This value is set to 10 ms in order to achieve a balance between the number of packets carried, and the delay and overhead. There are 512 independent subcarriers with OFDM and TDD in which the frame is structured with 50% divisions to simulate the synchronous transmission of VoIP calls.

Other physical parameters on the WiMAX model of OPNET required precise configuration, because some of them could become sources of confusion. OPNET has a parameter called “Efficiency Mode”; if this parameter is set to the default, which is “Efficiency Enabled,” the packet loss will be zero on all results regardless of the distances between the BS and SS. Once this value is changed to “Physical Layer,” the behavior becomes more reasonable. Figure 10 shows the rest of the configuration parameters of the WiMAX physical environment.

Transmission power is a variable that can be set from the BS and the SS. For real deployment, this parameter is set to typical values: 10 watts (W) for BS, and 0.5 W for SS. The antenna plays an important role at the BS, because it is the source of the transmitted signals. Thus, the OPNET model gives flexibility in altering some of the variables of this component. The antenna gain is set to 15 dBi; values are taken from the implementation examples [30].

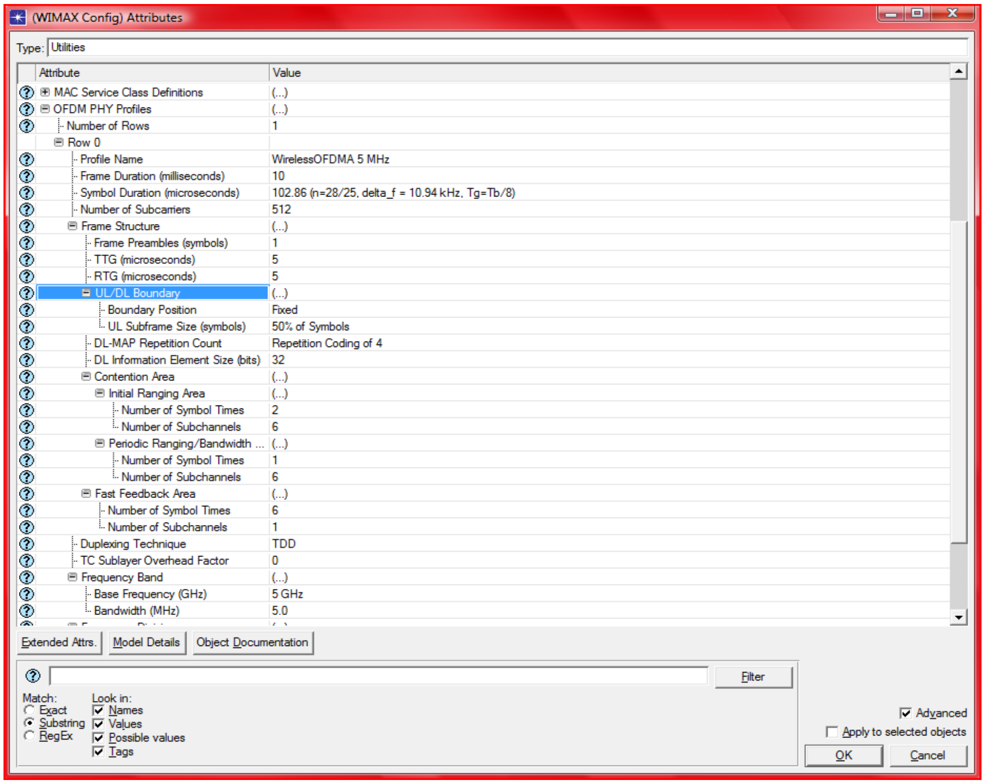

Figure 10: Figure 10: WiMAX Physical Setting

4.4.1 ACCESS PROTOCOL PARAMETERS

MAC service class definition is a parameter which specifies the required QoS. This QoS is the most important target in this research. The OPNET Model defines three classes of QoS, Gold, Silver, and Bronze. A packet is assigned priority the basis of its mapped class. For UGS simulations, the Gold class is used. Other parameters are the default values, because this service class is designed for UGS traffic.

For BE simulations, the Bronze class is selected, to simulate the BE class. In this class, priority will not have a role in the performance of the system. Because of the value limitation on OPNET in this service class, the WiMAX parameter, the maximum upload value per connection, has been set at 0.5 Mbps, the minimum value at 384 Kbps, and the scheduling type, by default, at BE.

Contention is a main consideration for BE traffic, the subject of this study . Careful choice of any parameter related to contention will make a big difference in the performance of the entire network. OPNET offers multiple parameters that can be set to control the contention method. The most influential values are described in the following paragraphs.

4.4.1.1 Contention Size (Cz)

The first parameter among these values is contention size C_z, which controls the number of transmission opportunities per frame. Transmission opportunities are used by the SSs to transmit their bandwidth requests when they have something to send. With OPNET, the value of contention size can be any integer number. Logically, the bigger the value, the smaller the chance that bandwidth requests from various SSs will collide. But transmission requests reduce the resources available for data transmission, because they consume part of the UL time. With these points in mind, the simulation was begun with the default value for the contention set at size 2. This study examines the optimal value that led to multiple simulations with different values for this parameter.

4.4.1.2 Back-off Window Size (W)

This parameter is shared with all SSs during the connection phase. The minimum (W_{min}) and maximum (W_{max}) values can be set to any integer number. The bigger the back-off window size, the longer the SSs have to wait for the next transmission opportunity. The default value for this parameter is set at 0 for W_{min} and 2 for W_{max}. The W_{max} value is tuned with other parameters to support the maximum number of SSs.

4.4.1.3 Request Retries Limit (r)

As was mentioned earlier, OPNET can simulate real-life situations by providing all parameters. This value controls the maximum number of retries during which an SS can compete for bandwidth for a packet that is to be transmitted. Each SS keeps a record of this number and monitors it each time a bandwidth request fails. When the maximum number of retries is reached, the SS removes that packet from the queue and starts with a new one. The bigger this maximum value is, the greater the number of retries for a packet before a bandwidth request fails. Thus a larger maximum value can introduce more delays for the rest of the packets in a queue. So, the optimal value of this parameter affects the performance of the entire network.

4.4.1.4 Timer (T16)

This value points to the time, in ms, that a SS needs to wait before considering a bandwidth request a failed try. When this much time passes and SS has not received a slot for transmitting data, the SS considers the latest BW request a collided or failed request and restarts the contention. Optimal values of this parameter can reduce the access delay. When the value is high, the SS takes longer to react to a collision request. When the value is low, the SS retransmits useless requests that have already been received by the BS but that have been delayed because of congestion at the BS. In our case, the minimum value for this parameter has been set to 10 ms, the time set for one frame. During the simulation, this value was increased progressively with other parameters to determine its optimal value.

4.4.2 THE OPTIMAL VALUES

A combination of the parameters mentioned above will have a definite effect on the BE class, because that class depends on contention algorithms to access the network. To determine optimal values for those parameters, multiple simulations were performed with various possible values. The aim of this study is to determine the best access parameter values to maximize the network capacity while providing good MOS values. In each simulation, if a change in one parameter in a combination of parameters led to an increase in the number of SSs supported, the resulting combination was considered optimal. Table 6 shows the maximum number of SSs supported for each combination of access parameters. Figure 15 shows graphical representations of those results, where, for reliability, each point was confirmed in ten repeated simulations. The optimal value of Cz is the smallest one, because this parameter has a direct effect on transmission delay. So, as the table shows, Case 7 is the best choice.

Table 6: Selection of Access Parameters for the WiMAX Experiments

Case No.	C_z	W_{max}	T_{16} (ms)	R	Maximum Number of SSs for MOS >3.7
1	2	2	10	4	5
2	4	4	10 or12	4	15
3	4	6	10–15	8	15
4	8	4	10–20	8	25
5	8	6	10–20	8	30
6	16	2	10–20	8	30
7	16	4	10–20	8	50
8	16	6	10–20	4–8	50
9	24	4	10–20	4–8	50
10	32	4	10–20	4–8	50

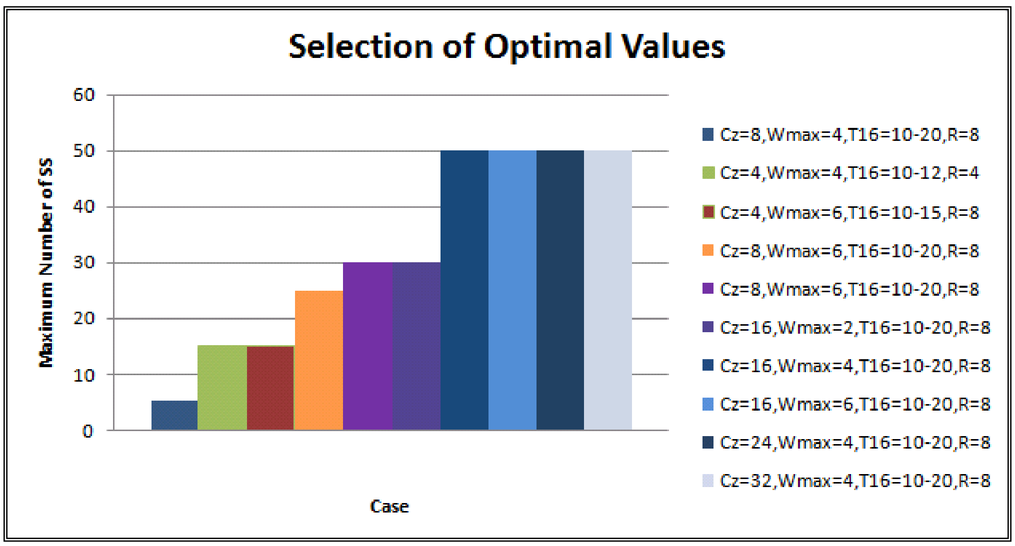

Figure 11: Maximum Number of SSs as a Function of Multiple Variables

At the BS, the antenna gain is set to 15 dBi and the maximum number of stations supported is set to 200, the maximum number of VoIP calls supported. Figure 12 shows the rest of the BS configurations. The modulation and coding schema selected for the system simulation are summarized in Table 7 and Figure 12.

Table 7: Modulation and Coding Schema

Schema Number	Modulation	Coding Rate	Slot size (bit)	Received SNR (dB)
1	QPSK	½	48	8.7
2	QPSK	½	72	10.6
3	16QAM	¾	96	15.8
4	16QAM	½	144	17.5
5	64QAM	¾	144	21.9
6	64QAM	½	192	23.9
7	64QAM	¾	216	25.1

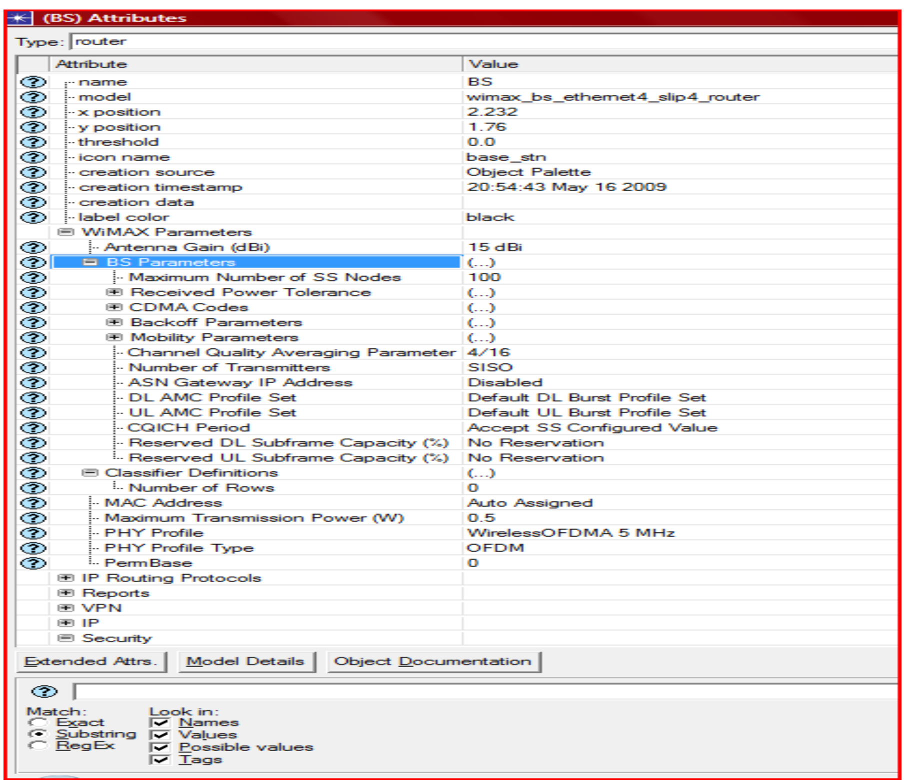

Figure 12: BS Configuration

4.5 LOCATION AND DISTRIBUTION OF SSS

In the system simulation, the SSs were distributed randomly within 5 km from the BS. For each simulation run, the SS was placed in a new position with respect to the BS. The modulation and coding rate, therefore, differ according to the SNR in the signal received from the SS, in accordance with the standard.

4.6 TRAFFIC MODEL

The aim of this work is to support VoIP calls with MOS values greater than or equal to 3.7. According to Table 8, this is possible but only with G.711 and G.729 encoding, because G.723 does not give the required MOS value. Any other encoding method gives lower MOS values at best. Taking this in consideration, the simulation part of this work compares encodings made with both G.711 and G.729.

Table 8: Maximum MOS Value vs. Codec Option

Codec	MOS (Mean Opinion Score)
G.723	3.6
G.729	3.9
G.711	4.1

As Figure 13 shows, the caller SS is placed to support the configured profile, with either UGS or BE. The callee is configured to be a server that is set to receive voice traffic.

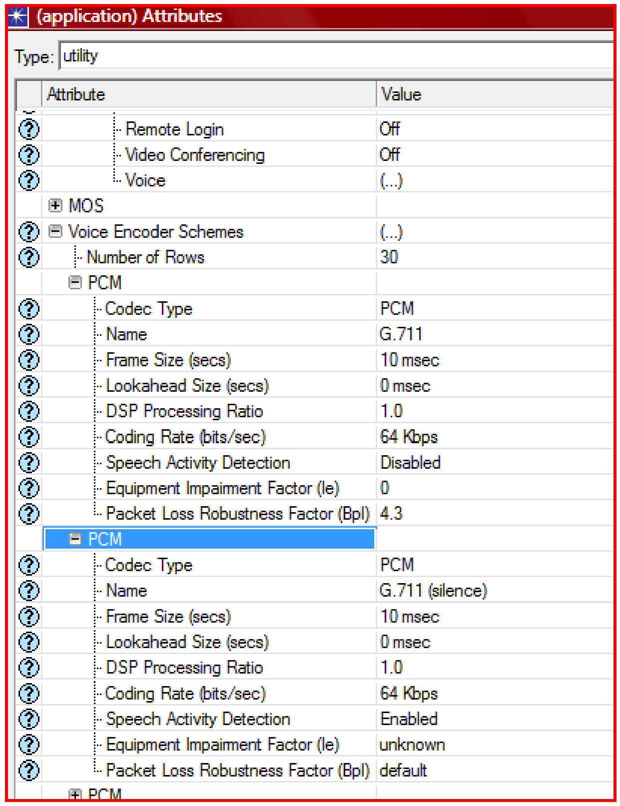

Figure 13: VoIP Encoding Settings

The simulation work involves two experiments, each with multiple scenarios. The first experiment focuses on normal VoIP traffic without any compression. The second one implies the use of VAD method to reduce bandwidth use and support more VoIP calls.

The aim of this work is to study the performance of the WiMAX network, with the above specifications, in supporting VoIP traffic. The nature of this traffic is one frame every 10 ms in both G.711 and G.729. Table 9 summarizes the scenarios simulated for each experiment.

Table 9: Scenarios Tested

Scenario	1	2	3	4
Class	UGS	UGS	BE	BE
Encoding method	G.711	G.729	G.711	G.729

In each scenario, the number of nodes was increased, from one node in the first simulation to the maximum at which the system can support a good VoIP call. The MOS value for each simulation is observed, because it is the focal point of this simulation.

The next chapter presents the results of these experiments, collected from an average of 10 simulations.

CHAPTER 5

RESULTS AND ANALYSIS

5.1 NETWORK DELAY

As was stated in the preceding chapter, VoIP will not perform well if delays between the communicating parties exceed a QoS threshold of 150 ms. This delay includes a one-way delay of packet delivery, network access, and a codec–related delay, including framing delay and look-ahead delay. If this delay is minor, it does not significantly degrade the MOS value of the VoIP, but if it exceeds 150 ms the degradation does become significant.

The first scenario studies the typical situation, in which UGS is used as the class for transmitting VOIP calls. Figure 14 depicts the average delay of this scenario, where each point corresponds to the average delay versus the number of SSs included in a particular simulation. From the displayed results it is clear that for G.711 the UGS provides almost fixed delays, between 46 ms and 76 ms. It starts with 46 ms when there is one SS only, and goes to 74 ms when the number of SSs is increased to 30. For more than 30 SSs, the BS cannot provide good MOS quality.

As can be seen in Figure 14, the delay with G.729 is greater than with G.711. It starts from 70 ms for only one SS, and reaches 117 ms when there are 60 SSs in the network. Most of the difference is a result of the encoding delay introduced by the G.729 method as known by theory.

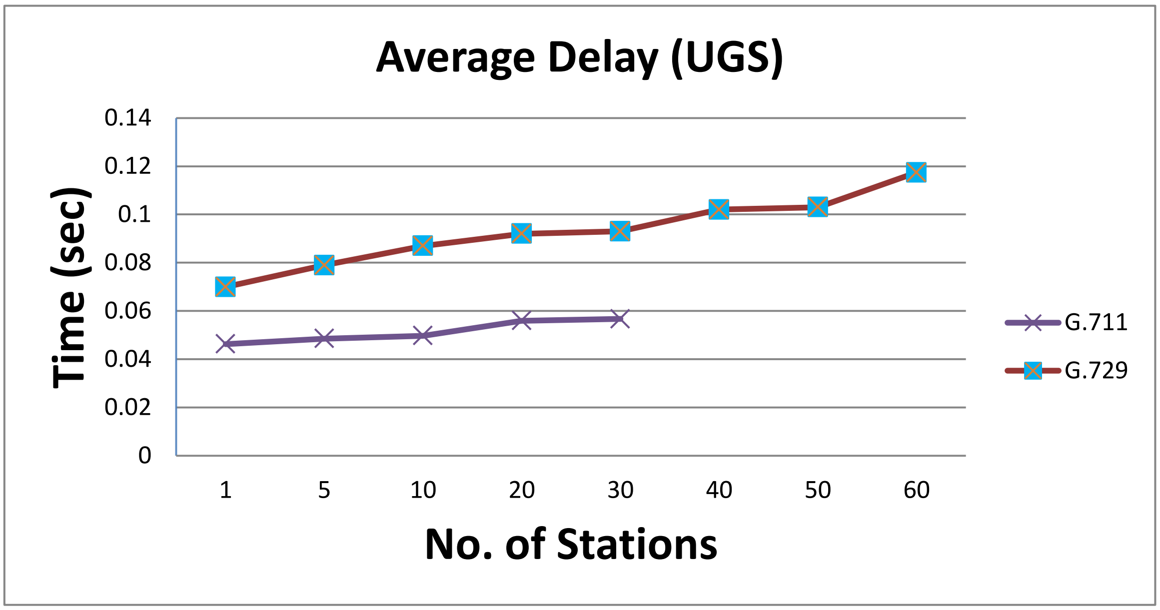

Figure 14: Delay Values for Experiment 1, Scenarios 1 and 2

The results for the third scenario, shown in Figure 15, show the average delay of the BE class with G.711. The results indicate that the delay is short when the number of SSs is lower than 50. The minimum delay that can be achieved is 40 ms, for only one SS. This value increases linearly as the number of SSs is increased. Once the number of SSs reaches 45, the delay increases sharply, but it is still below the recommended maximum value of 150 ms.

For 60 SSs, the average delay increases considerably, because the heavy load on the network increases the access delay and the number of connection failures. We expected degradation in the MOS value when the number of VoIP sessions reaches 60, because the delay value exceeds the recommended value of 150 ms. Thus this experiment shows that the maximum number of SSs that can be supported is around 55.

In a similar situation, for G.729, the delay values observed is represented in Figure 15. The minimum value, 88 ms, is reached when there is only one SS. The delay values increases linearly as the number of SS grew. It reaches its peak for 40 SSs, where it is over 150 ms. It increases to 200 for 50 SSs, and reaches 270 ms for 70 SSs. From this figure we can notice that G.729 results in a longer delay than G.711 with the same class, BE. This is expected to be due to the effects from the encoding and the process timing as per the theory.

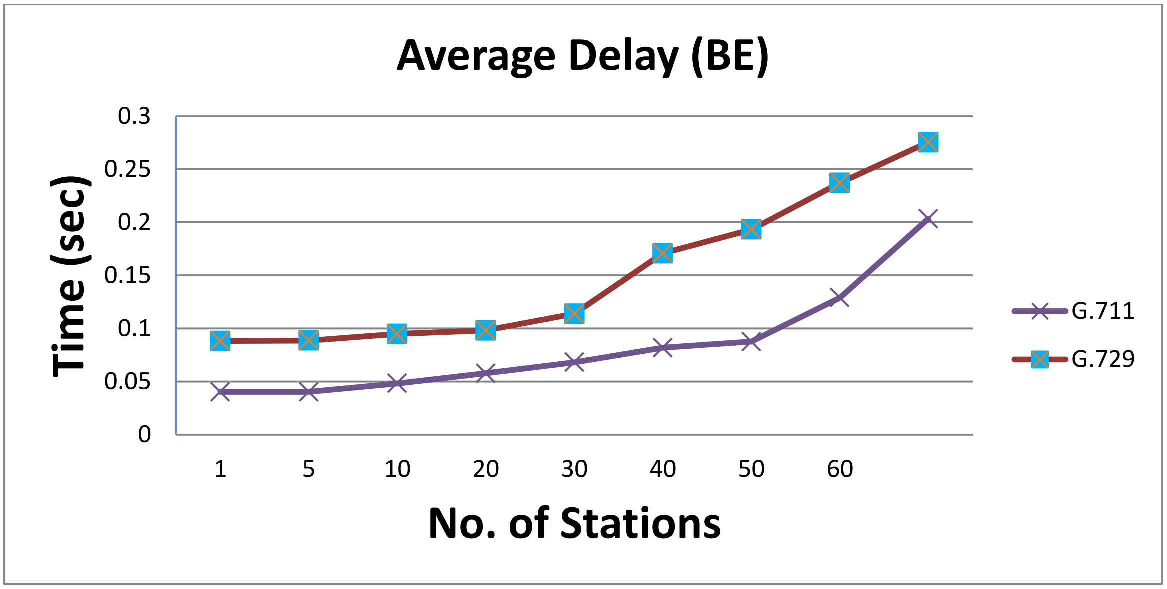

Figure 15: Delay Values for Experiment 2, Scenarios 3 and 4

The results obtained with these two scenarios show that both classes, UGS and BE, can deliver VoIP calls with a delay of less than 150 ms. Also, either G.711 and G.729 can be used as a coding method to keep the delay under 150 ms. The goal for this system is to support the maximum number of VoIP calls while keeping the delay below 150 ms. This can be accomplished by using G.711 with the BE class, because the network can support up to 55 SSs at the same time with a delay of 150 ms or less. Other scenarios show a good level of delay values, but with fewer SSs, depending on the situation. Other methods can be used to increase the number of SSs. We will explore some of those methods in the next experiment.

In the second experiment, in exploring for a way to increase system utilization, the VAD method was implemented. Figures 16 and 17 show the average delay observed from a set of simulations for this experiment, using UGS and BE classes. In these figures, each point corresponds to the delay versus the number of SSs included in a particular simulation, with a 95% confidence interval. It is clear that the delay is lower than those shown in the previous figures, where the VAD was not implemented. The difference occurs because without VAD the packet generation rate is higher. Excessive packet generation leads to a large overhead in packet headers from the RTP/UDP/IP/MAC protocol stack.

Figure 16 shows the delay for G.711 with the UGS class. For 40 ms and only one SS in the network, the values are very similar. For 70 SSs, the value increases to 60 ms. The results of the simulation show that the system can’t accept more SSs because it can’t deliver the required QoS.

Figure 16 also shows the delay for G.729. The values are higher than with G.711. With one SS, they start at 67 ms. This value shows a slight linear increase as the number of SSs increases. The more SSs there are, the longer it takes to arrange the traffic from them. When 90 SSs are doing VoIP calls, the delay reaches 117 ms. BS cannot accept more SSs while maintaining the required QoS. So this number of SSs can be considered the maximum that can be supported at the same time in this scenario.

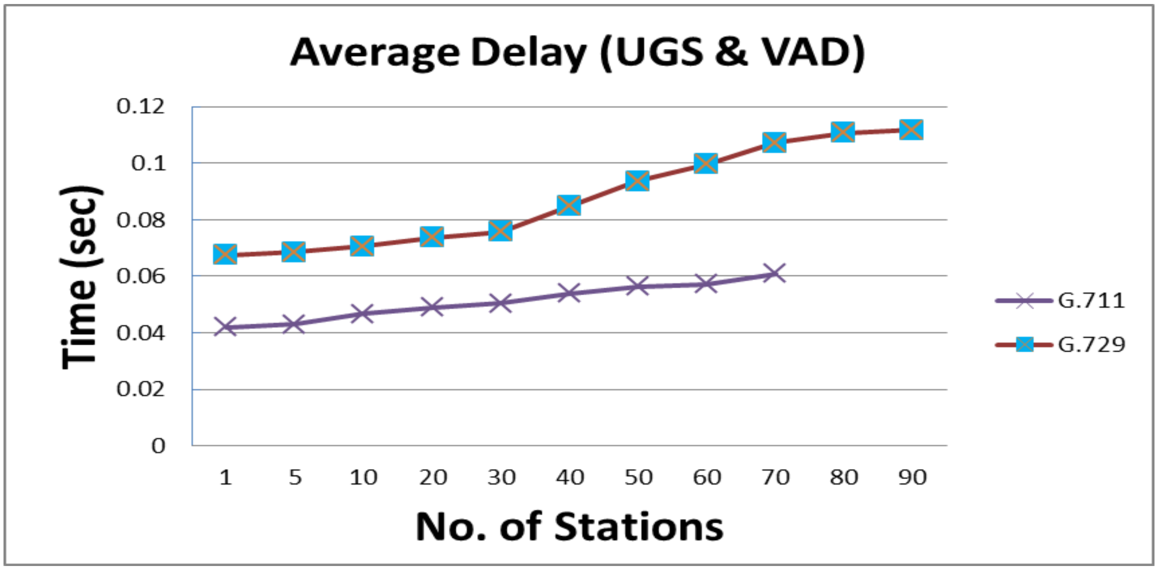

Figure 16: Delay Values for Experiment 2, Scenarios 1 and 2

The third and fourth scenarios study the ability to use BE to increase the number of SSs while keeping the delay value at 150 ms or less. The line for G.711 in Figure 17 shows, increases in the number of SSs, up to 50, cause the delay for G.711 to undergo only a minor change. For more than 50 SSs, the delay value increases slightly, because an increase in the number of packets leads to an increase in contention collision. But even with that increase, the delay value is still below the recommended maximum value. This continues until the number of SSs reaches 105. Once that number is increased to more than 105 SSs, the delay value increases dramatically. The main cause of this increase is the increase in the connection time; when many SSs compete to transmit at the same time, the load on the network is heavy.

Figure 17 displays the delay values observed when G.729 was used. The delay value started at 80 ms and increased slowly as long as the number of SSs was below 50. For more than 50 SSs, the value increased, as more packets were generated from multiple SSs and competition to access the medium increased the delay. For 60 VoIP calls, delay values went above 180 ms; they continued increasing until, for 110 SSs, they went up to 270 ms. This scenario shows that G.729 cannot support more than 55 SSs while maintaining an average delay value of 150 ms or lower.

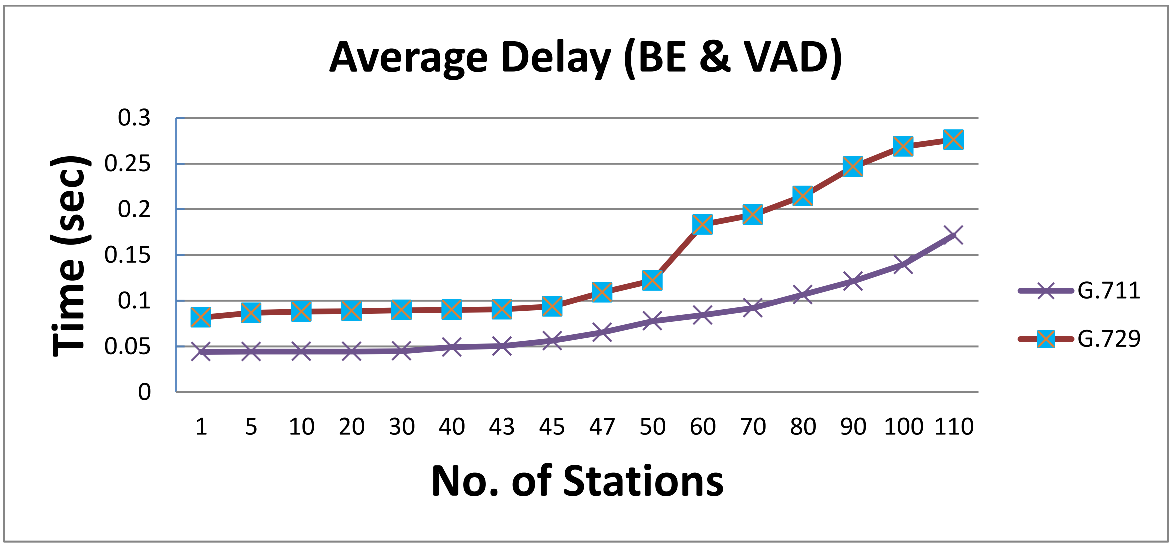

Figure 17: Delay Values for Experiment 2, Scenarios 3 and 4

The results from these two simulations confirm that WiMAX, with the UGS and BE classes, can support VoIP class within the recommended range of delay. We see a big difference in the maximum number of SSs that can be supported while the same level of delay value is maintained. This value is almost double between the two experiments, UGS and BE. This proves that the VAD method can be used to support a larger number of sessions while maintaining the same level of VoIP quality. Moreover, in this situation, G.711 seems to be a better choice than G.729 as encoding method, since in both experiments it was found to support more SSs with less delay.

5.2 PACKET LOSS

Packet loss is the ratio between the number of lost packets and the number of packets transmitted. It results when packets are sent but not received at their intended destination. Packet loss is an important parameter, for it affects the performance of the network. Unlike applications like email or FTP, which simply request a retransmission when data is lost, VoIP just discards voice samples that are lost or arrive too late. Packet loss degrades the quality of voice transmissions. By industry standards, the recommended tolerable packet loss is about 3%. The percentage of packet loss for various numbers of SSs at different operating ranges is graphed in Figures 18 and 19.

Figure 18 shows the results of a simulation in which silent detection is not enabled, and UGS is used. For G.711, it is almost constant between 1% and 1.5% for the accepted VoIP sessions. As has already been explained, UGS guarantees the required bit rate from the beginning of the registration, and tries to maintain this level. That explains why packet loss in this scenario is less than with those for BE.

The figure shows the effect of using G.729. Much as in the preceding scenario, the value is low when there is only one SS, and increases slowly, reaching its maximum , 2.6%, when 60 VoIP calls are accepted. This value contributes to the MOS value of VoIP calls; as it becomes higher, fewer VoIP calls can be supported.

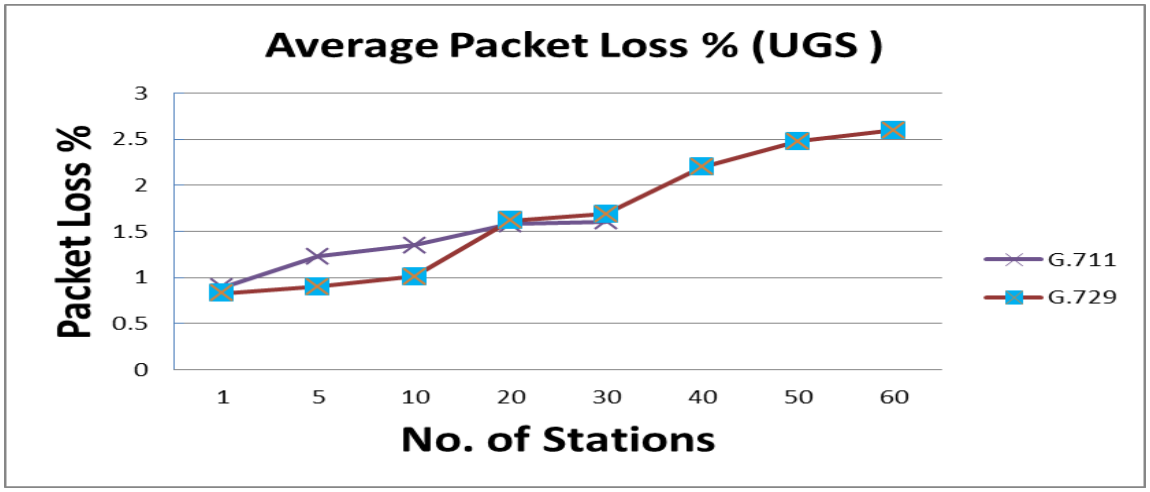

Figure 18: Packet Loss Values for Experiment 1, Scenarios 1 and 2 (UGS)

The third and fourth scenarios of this experiment produced simulation results for the use of the BE class. In Figure 19, we see that as the number of SSs is increased, VoIP traffic suffers dramatically from packet loss, for both G.711 and G.729. This can be explained as the result of the time required in getting connected for the first time and the possibility of receiving packets with errors. The percentage of packet loss for G.711 starts at 0.2% for just one SS. The value rises smoothly until the number of SSs exceeded 30 SS, when the average packet loss increased sharply. It reaches 3% when the number of SSs at various distances from the BS reaches 50. It increases to 5.23% when the number of SSs reaches 60 SS. The higher the number of packets generated, the higher the conflict with other SSs. That conflict causes many packets to be dropped.

For VoIP packets encoded with G.729 in the BE class, the results are similar. Packet loss starts at 0.02% for only one SS. As the number of SSs is increased, the loss increases more slowly than with G.711. For 35 SSs, it reaches over 3%; for more than 40 SSs, it goes to 5% and higher. Here again, the causes are network congestion and the generation of many packets.

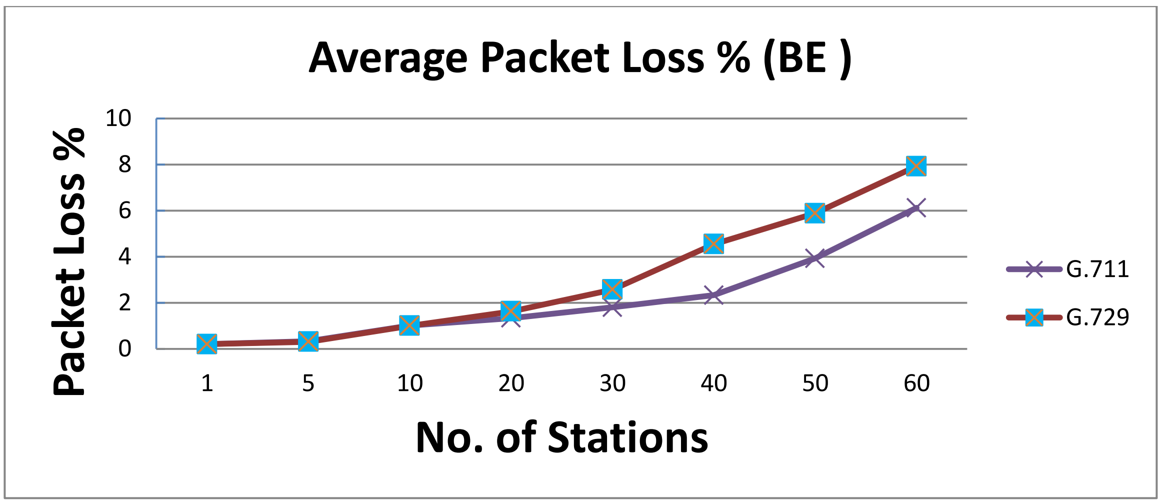

Figure 19: Packet Loss Values for Experiment 1, Scenarios 3 and 4

The results of these simulations support the first findings of the maximum number of SSs that can be supported. In all of the scenarios we observed almost exactly the same numbers, within the recommended values.

Figures 20 and 21 show the capabilities of the system to support more SSs while keeping the packet loss within the same range. The differences are due to the higher packet generation rate without VAD, and to the fact that with BE, the generation of more packets leads to more conflict among packets, and thus to packet loss.

In Figure 21, for the first scenario, using UGS and G.711 and assuming only one SS, average packet loss began at 0.7. For 50 SSs, this average reaches 1.6%. This value is still within the acceptable range, below the maximum of 3%. For G.729, by contrast, the average starts at 0.6% for one SS and increases linearly up to 2.4% for 90 SSs. Again this is within the acceptable range, and more SSs can be supported than with G.711. We can notice almost a fixed packet loss that is due to the nature of the UGS QoS class, where there is a confirmed slot for each SS to transmit its data, and the packet loss is minimal and depends on other signaling conditions, such as SNR.

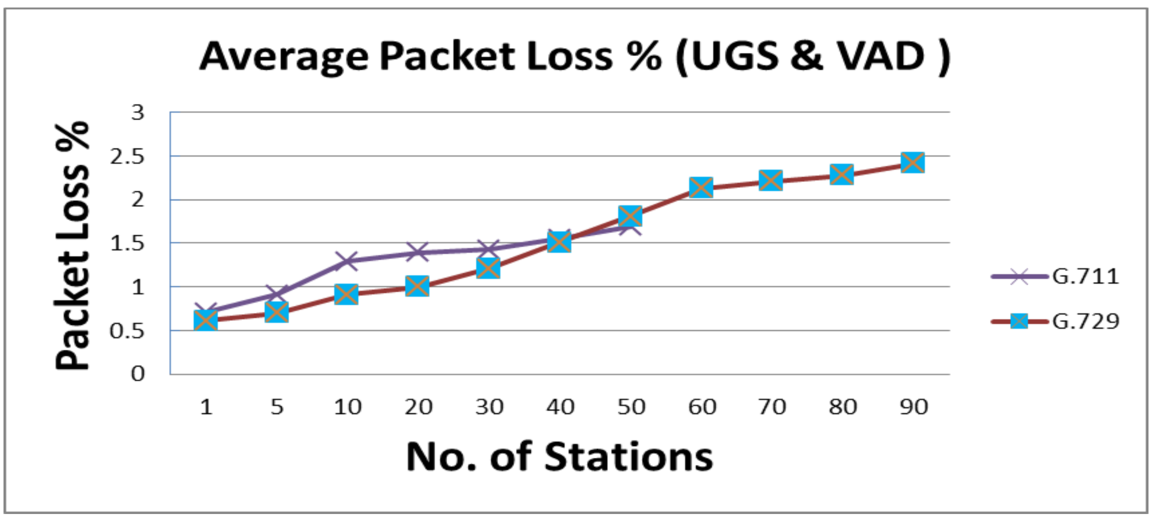

Figure 20: Packet Loss Values for Experiment 2, Scenarios 1 and 2

Even better system utilization is shown in Figure 21, for BE with both G.711 and G.729. For one SS, the packet loss is 0.02%. For more SSs, there is a higher percentage of packet loss; for more than 85 SSs, it exceeds the recommended value, 3%. For 100 SSs, leading to many collisions, retries, and queuing from the sender side, the value reaches 5.2%.

With G.729, the packet loss starts at 0.2% for one SS, and exceeds the recommended value at 47 SSs. For more SSs, the whole system suffered from more packet loss. Moreover, this value increases sharply when more packets are generated by more VoIP calls. This is normal because an increase in the number of packets increases the probability of collisions. Figure 21 shows a packet loss of 7.79% at 110 SSs. This is even worse than was seen in the scenario with G.711 because more packets were generated.

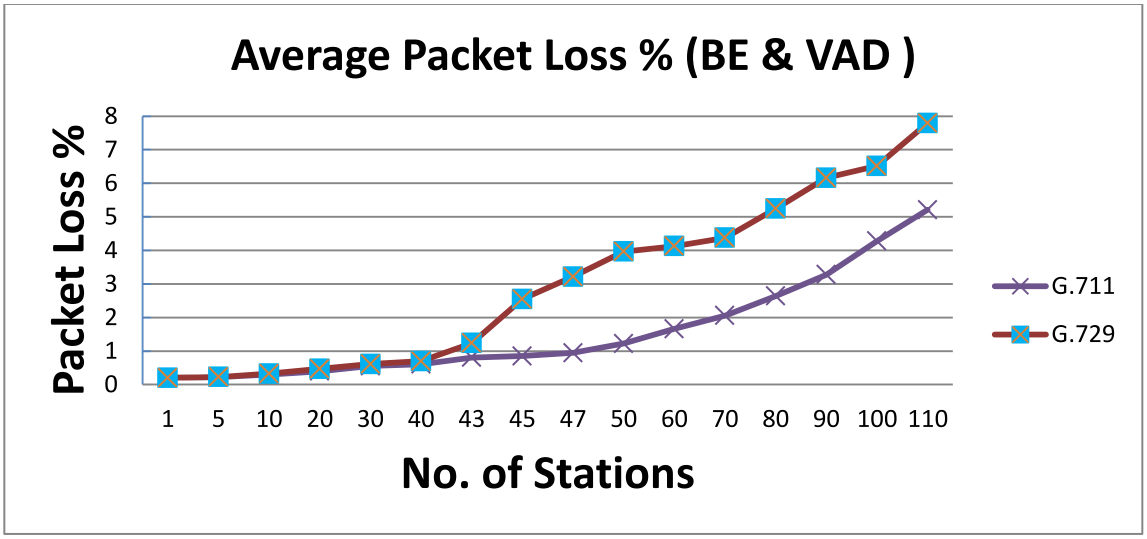

Figure 21: Packet Loss Values for Experiment 2, Scenarios 3 and 4

The results of these two experiments show that the WiMAX BE service class has the potential to support a low packet loss with different loads. The results of both experiments show that WiMAX with UGS can maintain the packet loss within acceptable values to some extent. Higher utilization of the system and maintaining an acceptable level of packet loss can be achieved by using the BE class. The values of packet loss are a little higher than with UGS, but are still within the accepted value of 3%,. By comparing results, we see that the VAD is the best choice to support larger numbers of SSs by increasing network utilization with the same VoIP QoS. Among the mentioned results, the results are best when G.711 encoding with BE class is used. In this scenario, the system supports up to 85 SSs with packet loss of 3%. This will enable the network to serve more VoIP.

5.3 JITTER

As was shown in the preceding chapter, jitter does not directly affect mean opinion scores and R-factors. However it does affect those parameters, and thus VoIP, indirectly. This is because when the jitter is too high more packets are rejected in relation to network jitter buffers. The main reason for having buffers is to eliminate as much jitter effect as possible. “Jitter can be defined as inter-packet delay between the packet just received and the previous packet” [20]. It is calculated by the following formula:

$$D_{ti} = t_i ti - t_{ii-1} \tag{7}$$

Where i = 1, 2, ..., N, and t is the time at which the i^{th} packet arrives.

Figure 22 shows the jitter values in Experiment No. 1. When there is only one SS in the system with UGS and G.711, we see only a small jitter value, 0.15 ms. This value increased slightly as the number of SSs increased. For 30 SSs, it reached 0.35 ms. In case of G.729, it faces almost same jitter value, and increased linearly with number of SSs. For 60 SSs, the value of jitter reached and saturated at nearly 0.4 ms, which is acceptable value for jitter. Applying jitter buffers on the recipient side reduced jitter considerably.

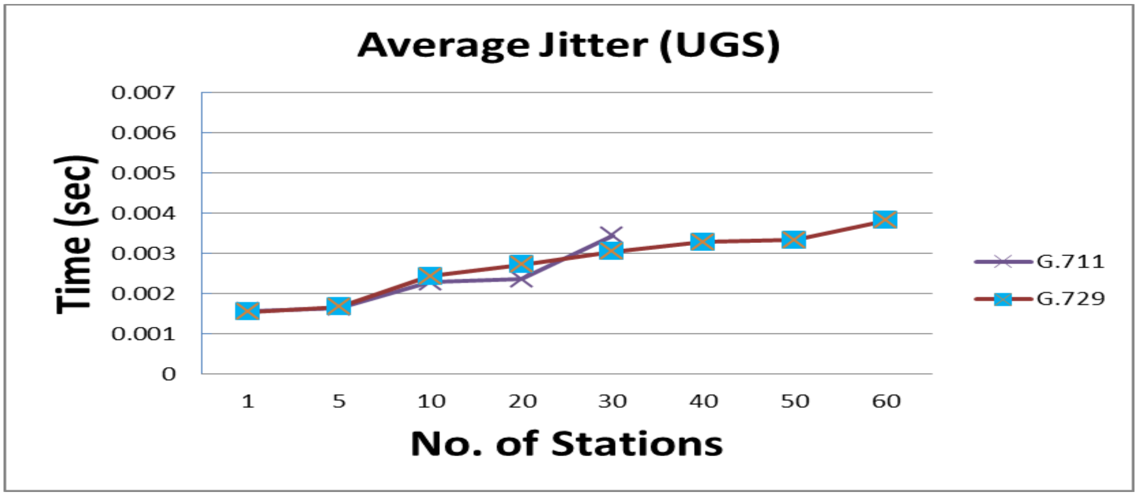

Figure 22: Jitter Values for Experiment 1, Scenarios 1 and 2

Jitter values for BE are shown in Figure 23. At the beginning of the simulation, the values for G.711 and G.729 were the same, because the network is not congested yet. With G.729, the values increase slightly with the number of SSs. For a large number of generated packets and more than 40 SSs, jitter values are higher, above 0.5 ms. But jitter with this value affects MOS far less than do other factors, such as access delay. Because its effect is far less significant than that for delay, jitter value will not have a major effect in this experiment.

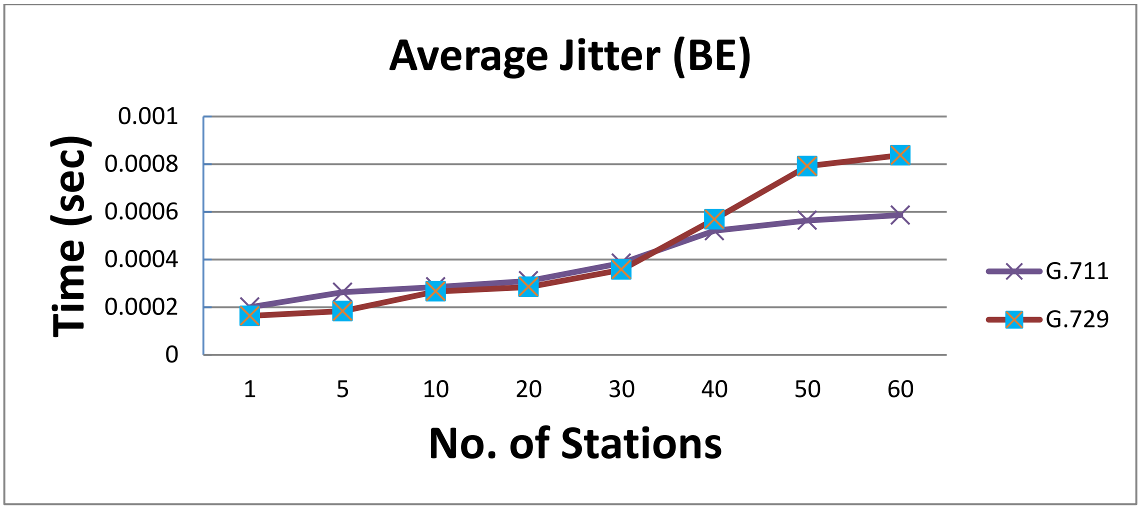

Figure 23: Jitter Values for Experiment 1, Scenarios 3 and 4

Figures 24 and 25 show the jitter values obtained in experiment 2. The results were almost the same as those obtained in experiment 1, except that more SSs are supported. In this experiment, the jitter value is negligible by comparison with the average delay value and does not have a major effect on the QoS of VoIP sessions. In summary, the value of jitter is higher for the BE class, but that class can support more VoIP calls with good MOS values.

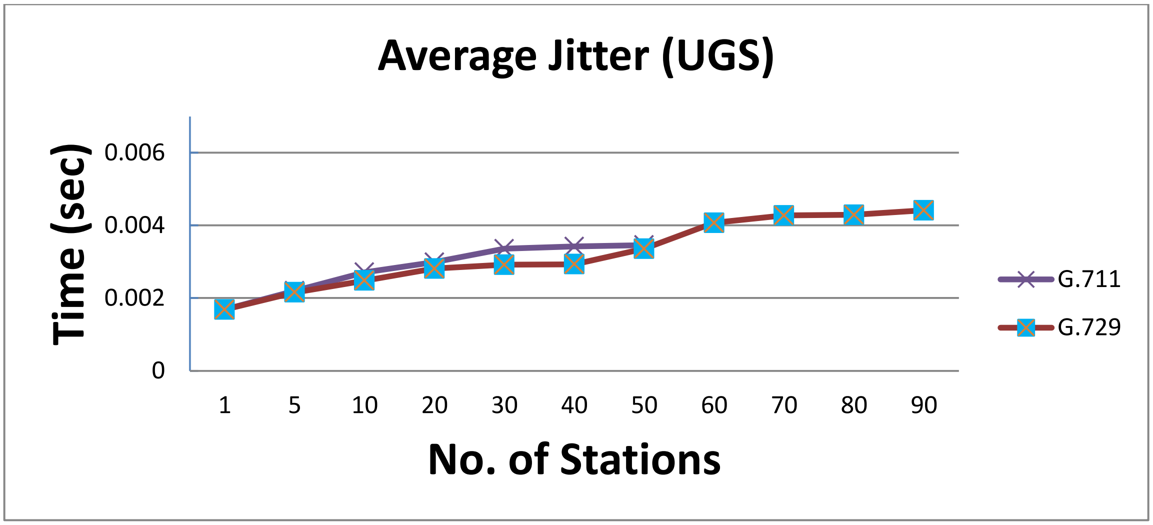

Figure 24: Jitter Values for Experiment 2, Scenarios 1 and 2

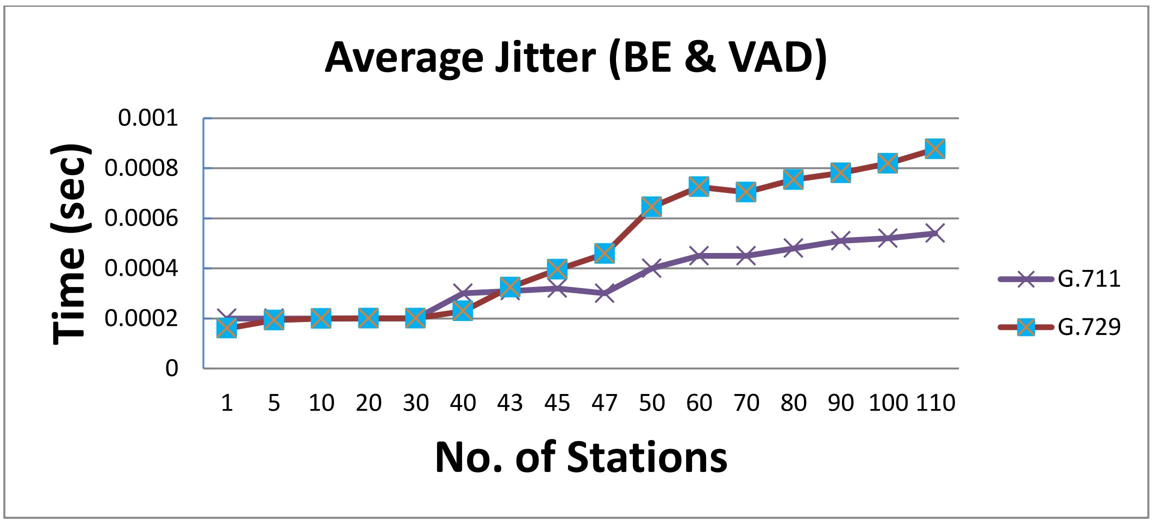

Figure 25: Jitter Values for Experiment 2, Scenarios 3 and 4

5.4 BANDWIDTH USE AND UTILIZATION

The next simulations were concerned with the effect when VoIPs were loaded from multiple SSs, in particular at the effect of successful and collided requests on bandwidth and link utilization. In this analysis, we considered the uplink portion of the link only as it was constrained by the VoIP performance. The queue length for each SS was closely related to the sending rate. Any difficulties in accessing the link would directly affect the bandwidth sending rate and the queue link on the SS, and vice versa. If bandwidth requests from two or more SSs collide, the receiving frame will not find slots for both. Therefore, the utilization of the bandwidth will be degraded. On the other hand, if all SSs transmit their bandwidth requests successfully and the BS grants a slot to each of them, the bandwidth utilization will rise. Moreover, the use of dynamic modulation affects the net throughput of the UL, because the maximum can be attained only if all SSs use 64QAM. On the other hand, throughput will be lower if binary frequency shift keying (BFSK) is used

The total raw uplink bandwidth is 6 Mbps, as was said earlier. Not all of this bandwidth is used for data transmission; some of it is consumed by transmitting control data, including MAC headers, ranging requests, and, with BE, bandwidth requests. Because we selected the optimal values before and through the multiple simulations, we observed that the net amount of bandwidth available for data is around 5 Mbps.

Figures 26 and 27 compare the results of UGS and BE for transmitting VoIP calls encoded with G.711 and with G.729. The first graph shows the bandwidth utilization of UGS for G.711 encoding and for G.729. For G.711 and one SS, it starts at 130 Kbps. For more SSs, BS assigns more bandwidth as the number of SSs increases. For 30 SSs, the maximum bandwidth utilization reaches 3 Mbps. Some bandwidth remains unused, but

BS cannot maintain the required QoS, because bandwidth is not the only constraint on QoS. Utilization is lower with G.729, because packets are transmitted in 8 Kpbs. This is clear from the figure, where G.729 is less than G.711 by almost 50%. The highest value for this utilization was 2.1 Mbps, for 60 SSs.

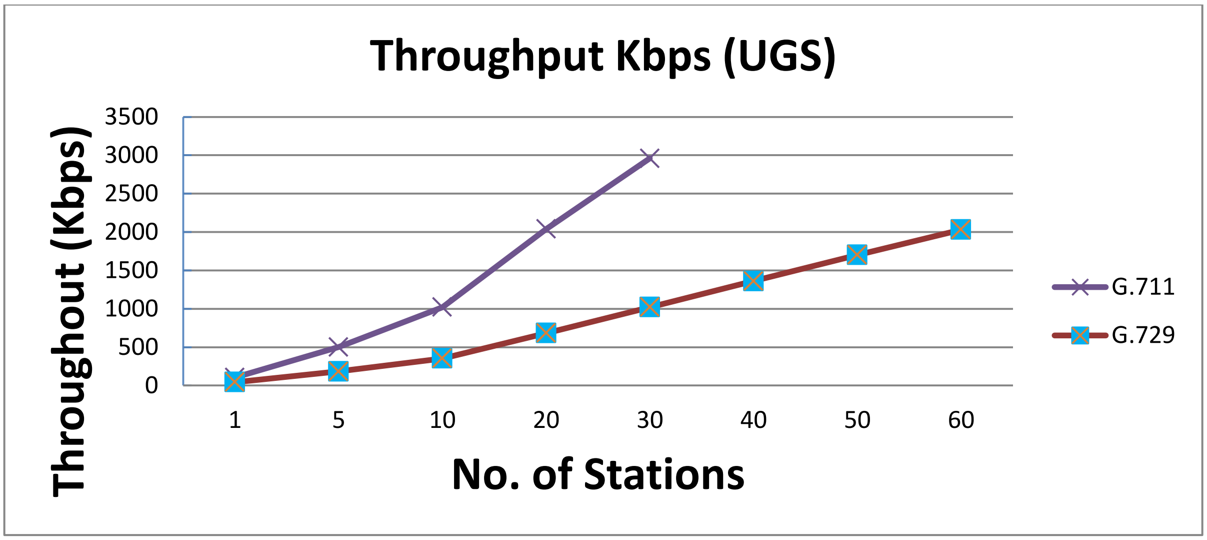

Figure 26: Throughput for Experiment 1, Scenarios 1 and 2

The third scenario of this experiment was a simulation for BE. With low collision rate, all the SSs received the bandwidth they required. As a result, this utilization of the link increases linearly with the number of SSs. For G.711 the average total throughput increases with the number of SSs. It is clear that the peak value of the aggregate, 4.8 Mbps, is reached when the number of SSs reaches 50. For more than 50 SSs, more bandwidth requests collide, and since the bandwidth baggy back, where bandwidth request is sent along with data packets, was not enabled, the aggregated throughput and the utilization of the uplink declined. Also, the fact that the SSs were distributed among multiple locations and transmitted via different modulations contributed to the lowering of this value.

The results for G.729 encoding are the same, because the link utilization increases linearly with the number of SSs. The network utilization is less than with G.711, and this supports the theory mentioned in the introduction, by which G.729 needs less than 50% of what G.711 needs. This value starts at 40 Kbps for one SS and increases to 2.3 Mbps for 60 SSs. This increase is proportional to the number of SSs.

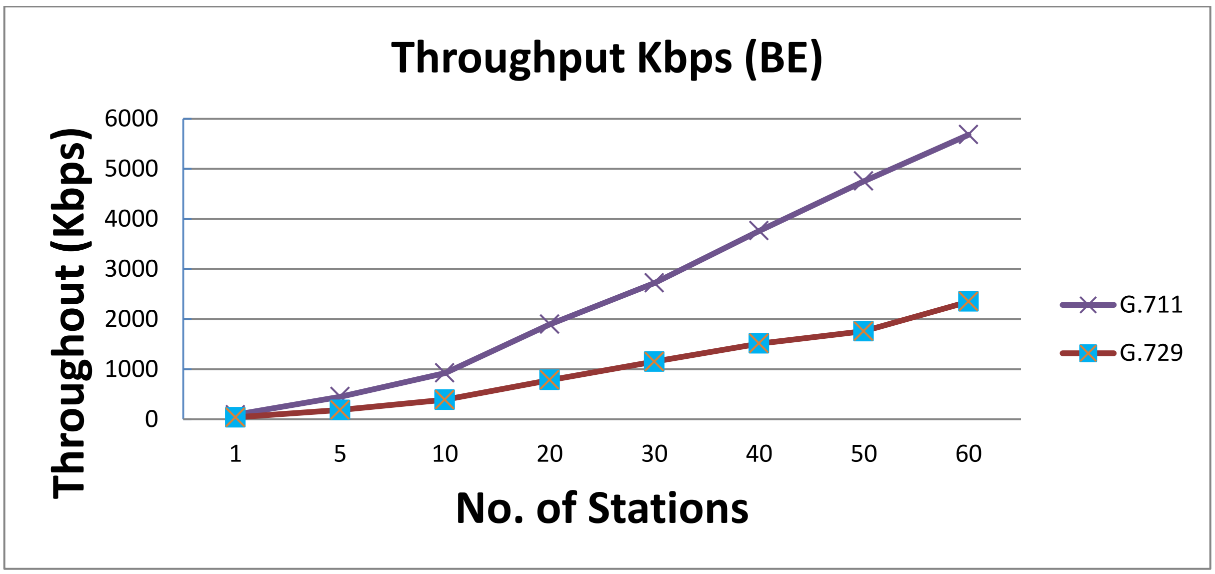

Figure 27: Throughput for Experiment 1, Scenarios 3 and 4

In all the scenarios of experiment 2, the utilization of the available bandwidth is clearly better than in experiment 1. Figure 28 shows that when VAD is used, bandwidth use is higher than in the comparable situation in experiment 1. The results for G.711 range from 58 kbps to 3 Mbps, depending on the number of SSs. In this scenario and the following one, the bandwidth requirement was determined from the BS during registration, and remained the same till the last communication.

The results shown in Figure 28 show that with only one SS, less bandwidth was used with G.729 than with G.711. As the number of SSs increases, bandwidth use increases linearly. For 90 SSs, it reaches the same level as it reached for 60 SSs with G.711.

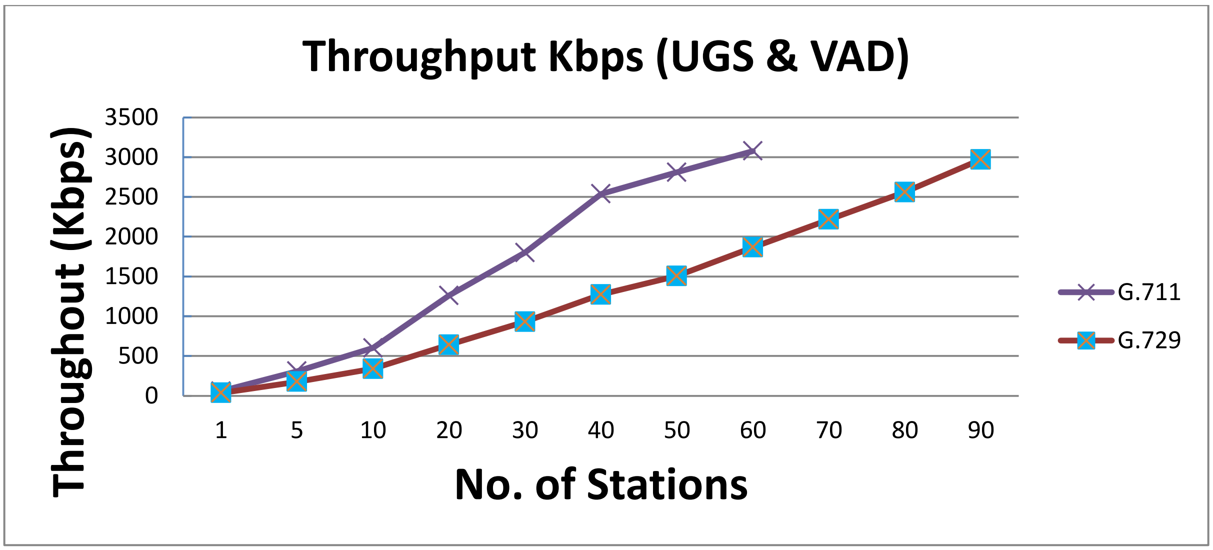

Figure 28: Throughput for Experiment 2, Scenarios 1 and 2

In the same manner but on a different scale, the next scenarios show the same thing. Figure 29 shows the link utilization for VAD with BE class. Because of the use of VAD, the required bandwidth for each VoIP session is reduced by half, for both G.711 and G.729. The graph in the panel below shows how bandwidth consumption increases for larger numbers of SSs. The results are different from those obtained in the first experiment. For both coding methods, peak value is attained when for 100 SSs. At this point, the total throughput obtained is 4.5 Mbps for G.711 and 1.5 Mbps for G.729.

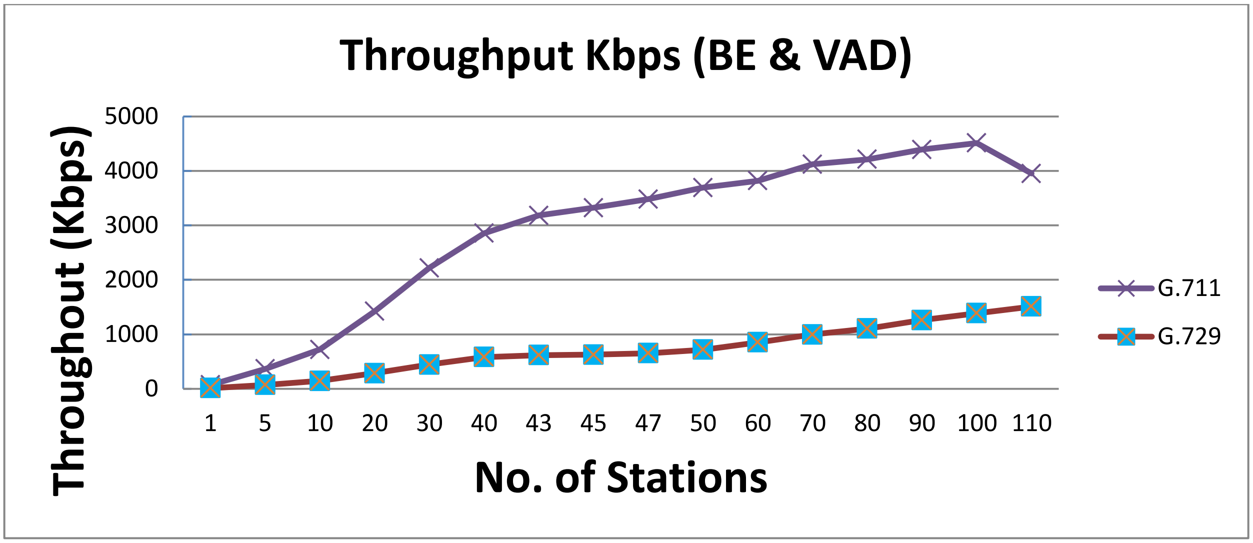

Figure 29: Throughput for Experiment 2, Scenarios 3 and 4

5.5 VOICE QUALITY

From the values of average delay, packet loss, and jitter, the OPNET can calculate the expected quality of VoIP transmissions. The average MOS values from different simulations for experiments 1 and 2 are shown in Figures 30 to 33.

Figure 30 shows the MOS value of UGS for both G.711 and G.729. For G.711, the highest value was almost constant because of the nature of UGS. For only one SS, the value is 4.15; for 30 SSs, it drops slightly, to 4.02. All of these values, above 4, prove that for this scenario the delivery of VoIP calls will be excellent. For G.729 the results are similar where that the value remains almost constant with a greater number of SS. For one SS, it is 3.8. As the number of SSs increases, the quality of VoIP is slightly degraded, and the MOS value decreases to 3.7. This can still be considered excellent VoIP quality. This value is the target QoS value we are looking for in this work, because when the MOS value is below 3.7, many users will be dissatisfied with the voice quality. As would be expected from the delay and packet loss values in the preceding sections, MOS value reaches a good level value, 3.7, when there are 60 SSs in the network. This is due to the proportional relationship between the MOS value, delay, and packet loss values.

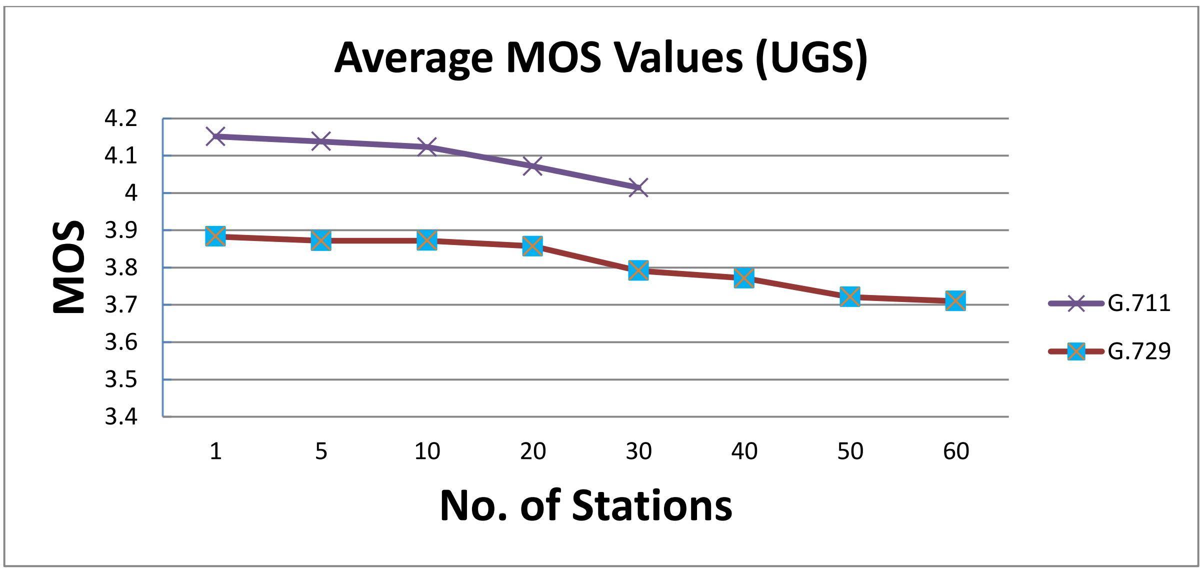

Figure 30: MOS Values for Experiment 1, Scenarios 1 and 2

The same situation is shown for BE traffic in Figure 31. Here, for both G.711 and G.729, the MOS is degraded as the number of SSs is increased. The results for G.711 show that the MOS value is 4 for one SS, and is degraded as the number of SSs is increased. Once the network becomes congested, the MOS value is decreased. For 50 SSs, it reaches the borderline, 3.7. For 55 or more SSs, it drops below that value. For G.729 the MOS value behaves much the same, but it drops lower with a large number of SSs. It starts at 3.85 with one station and is degraded when the number of SSs is increased. It reaches 3.7 with 30 VoIP calls, and then begins to decrease. This is due to the delay and the packet loss that occurred during a transmission. It is a little less than 40% of what G.711 can support with a good MOS value, because both delay and packet loss were lower with G.711.

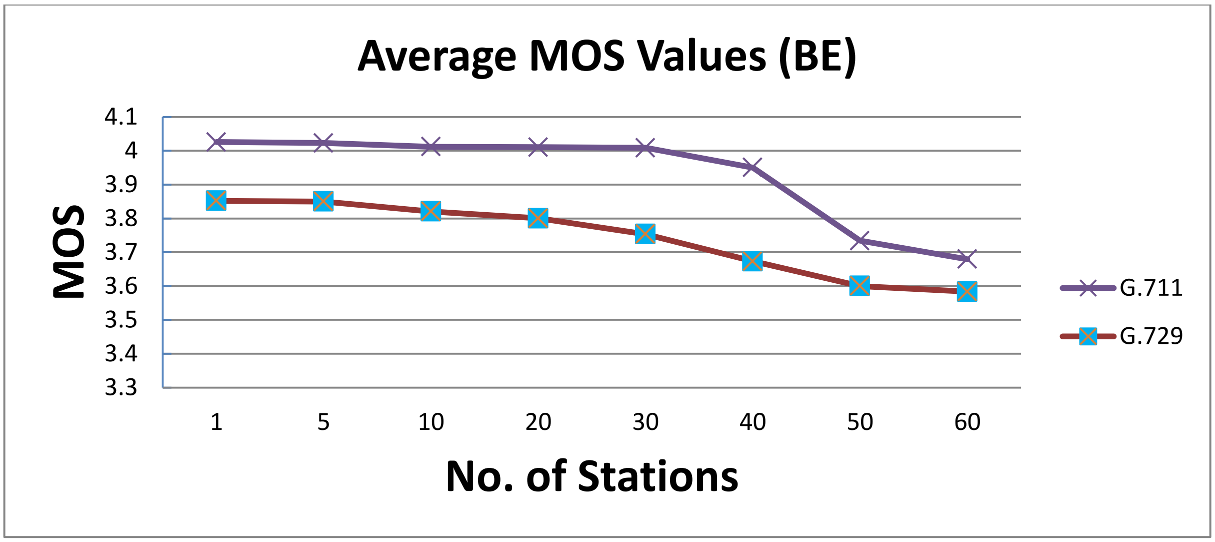

Figure 31: MOS Values for Experiment 1, Scenarios 3 and 4

The OPNET results obtained in the second experiment show that better results are obtained with the VAD method. Figures 32 and 33 show the results of applying this method with different scenarios.

In Figure 32, about G.711 with UGS, each of the first four points shows an MOS value over 4.1. The value is 4.15 for only one SS in the network, and drops to 4.1 for 30 SSs. This is explained by the increases in both delay and packet losses shown before.

Even when the number of SSs is increased to 50, the MOS value continues to be excellent, staying at 4. When the number reaches 60 SSs, the MOS value drops to 3. The same thing happens for G.729. The MOS value starts at 3.88 and remains above 3.8 until the number of SSs reaches 40. With more VoIP calls, the MOS value drops slightly. With 80 or more SSs on the network, it drops to 3.7. This drop is caused by the increase in the delay and packet loss, as was shown above.

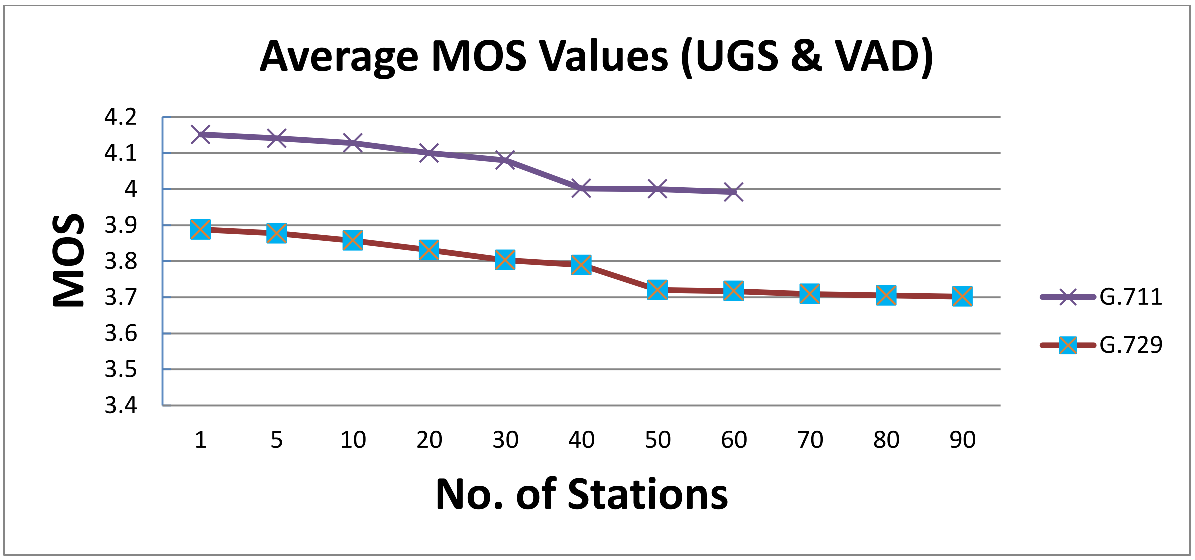

Figure 32: MOS Values for Experiment 2, Scenarios 1 and 2

Lower values of MOS, but still with excellent support, are shown in Figure 33, where the VAD method was used with the BE class. With G.711, as the figure shows, excellent VoIP quality can be provided for 100 SSs. The MOS value starts at 4.02 and remains there for the first five points. As the network becomes congested, this value drops linearly, reaching a value of 3.7 for 100 SSs. At an MOS value of 3.6 or lower, most users would not be satisfied with the voice quality. From this result we can see that BE with G.711 can serve the maximum number of users with good voice quality.

G.729 encoding could be used to serve users with low bandwidth requirements. For the first 50 users, the method can provide good voice quality while consuming only a small part of the bandwidth. For more than 50 users, however, the MOS decreases to 3.6, because this method of encoding introduces additional delay. Further network congestion degrades the MOS, to 3.5 with a network of 100 SSs.

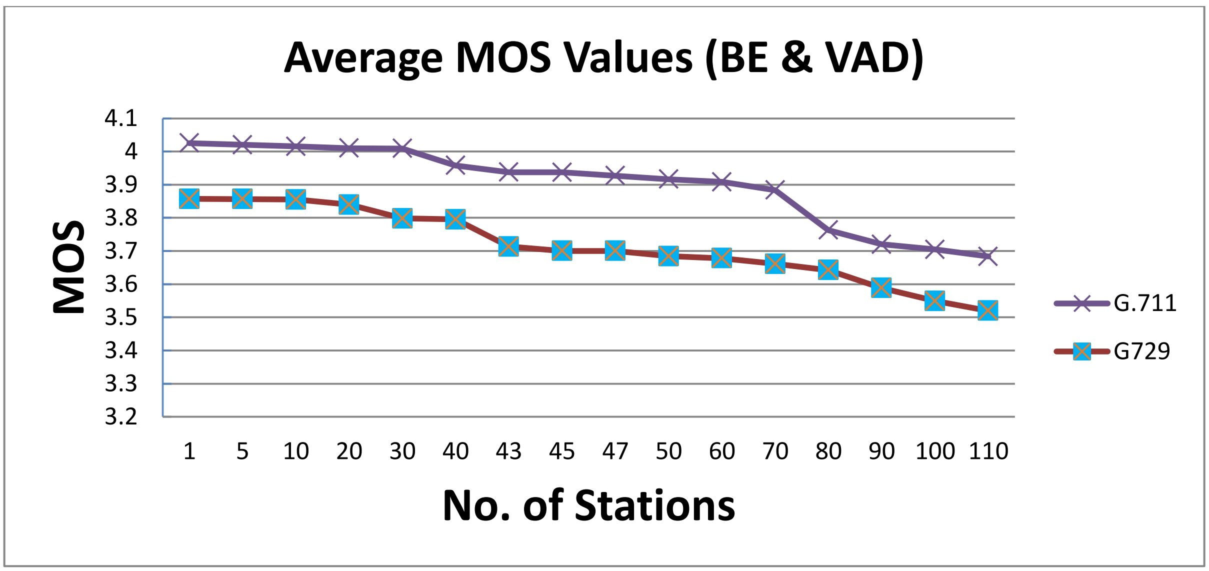

Figure 33: MOS Values for Experiment 2, Scenarios 3 and 4

From the results of these experiments, we can conclude that VoIP is supported by an excellent QoS over WiMAX UGS and BE classes. They show further that UGS can be used to deliver excellent VoIP calls consistently, provided that the protocol includes a method for guaranteeing service. If the network imposes a limitation on the bandwidth, G.729 can be used, because it reduces the bandwidth requirement by half. In this situation, UGS still can support more users but with lower voice quality.

The results of experiments 1 and 2 also show the ability of the BE class to support good voice quality for a large number of SSs. In the first experiment, they show that at least 50 SSs can be supported when the VoIP encoding is G.711 and the SSs were distributed at random distances over 5 Kms from the BS. With G.729, this number is lower, but with lower bandwidth consumption.

For better utilization of the WiMAX network, G.711 with VAD system can be used. This doubles the number of SSs that can be supported, and thus doubles the number of VoIP calls. The results also show that the same system can support at least 100 bidirectional calls, with the same quality. This number, for the BE class with G.711 encoding, was the highest number obtained in any of the scenarios.

The assessment of the best case depends on the network operator's requirements. For the best VoIP call regardless of the number supported, the better choice would be to use UGS and G.711. But for the maximum number of supported users, with good VoIP quality, they should use BE class with G.711 encoding. This is the choice in this work, where the aim is to find the best solution for supporting the maximum number of users with good VoIP quality.

CHAPTER 6

RECOMMENDATIONS FOR DEPLOYMENT

This chapter offers recommendations, based on the results reported here, for deploying WiMAX. These recommendations are for a WiMAX infrastructure meant for providing VoIP connectivity in specific areas.

One way to improve the economic deployment of any technology is to increase the capacity of the network—that is, maximize the number of simultaneous connections supported by the network, while maintaining QoS. For this purpose, as the results of the simulations show, any provider of VoIP services should be able to implement G7.11 with the VAD mechanism. Another way to improve deployment is to reduce the cost, for either the operator or the user. Many other aspects of deployment should also be considered. Some of these aspects are applicable to the deployment of the VoIP over WiMAX, and others are specific to VoIP over the BE class. All of them affect system performance and deployment cost. The following is a summary of those factors and the suggested values.

6.1 ENVIRONMENTAL CONDITIONS

Because capacity analyses and requirement gathering are the two most important steps before deployment, many aspects of the target areas need to be investigated. Each region is unique in such characteristics as area, population growth rate, and customer distribution. These are critical factors in order to specify requirements.

6.2 BASE STATION DEPLOYMENT

The WiMAX BS is one of the most crucial components in the WiMAX network solution. A typical rural area or small town has only a few hundred subscribers, or at most a few thousand. Depending on the user's location and geographical conditions, the number of BSs required has to be quantified. The operators can either make use of the existing mobile sites or build a more expensive Greenfield installation. If two or more BSs are required, they should be placed on a cellular structure in a typical WiMAX network deployment, similar to GSM networks [30], [31].

There must also be a BS backhaul connection to the core network. This can be a leased lines or a point-to-point wireless link to an aggregation node or fiber node. Whichever is used, the backhaul capacity must be sized as appropriate for the capacity of the base station [30].

As for BS hardware, our simulation study shows that a simple WiMAX BS with transmission power of 35 dBm will be able to cover an area with a radius of more than 5 km. For a targeted area within this range, a WiMAX BS is sufficient.

6.3 END-USER DEVICES

For end users with fixed WiMAX networks, the customer premises equipment (CPE), indoor or outdoor, is appropriate. The end user is connected directly to the WiMAX BS. A less costly choice is to use the current analog phone device. Telephone devices in a group are connected to a Foreign Exchange Subscriber (FXS). The FXS then converts each analog telephone call into a VoIP call [19].

6.4 UL LINK BUDGET

In most cases, the UL Link budget is an element limiting both range and coverage area of a WiMAX BS. The simulation in our study used a 1:1 ratio. For VoIP this is the best fit, because users expect to receive the same amount of data on the UL and DL sides.

6.5 DUPLEXING

Two types of duplexing, TDD and FDD, have been discussed in earlier chapters. This study shows that TDD is the best choice, because it ensures that the channels will be divided equally between the UL and the DL. Also it eases the requirements from the user side, because it does not need a smart device that can send and receive at the same time on different frequencies, such as FDD [25].

6.6 TRANSMISSION POWER

Transmission power of 10 W for base stations and 0.5 W for fixed users is sufficient to cover an area with a radius of 5 km.

6.7 CONTENTION PARAMETERS

Initial ranging and bandwidth requests are the two primary sections of the call admission control (CAC) procedure. The IEEE 802.16 mechanism for resolving contention is controlled mainly by two sets of parameters: the number of contention slots, and the initial and maximum window values of the back–off. We found in the simulation that a larger number of contention slots, greater than 16, does not improve performance, and a lower number increases the probability of collision. As for back–off window size, starting from 0 to 4 gives the best fit. The retry process continues until the maximum number of retries has been reached. The best choice for the maximum number of retries is 6; after that, the SS will drop that packet. The parameters mentioned above, which are set from the BS side, are the best choices for transmitting VoIP with the BE class.

CHAPTER 7

CONCLUSIONS AND RECOMMENDATIONS FOR FUTURE WORK

This thesis has explored the feasibility of operating VoIP on WiMAX BE service class as an alternative for default UGS and rtPS MAC services. In two experiments, using OPNET, two different service classes, UGS and BE, have been simulated. Also two methods of encoding VoIP were simulated. Through two different scenarios, In each experiment, different numbers of SSs made calls from various distances on the networks to collect statistics for packet loss, delay, and jitter. The results show that BE class has the required capabilities for serving such application with G.711 encoding.

In the first experiment, all of the simulations showed that the WiMAX BE class can support more than 50 G.711 VoIP bidirectional calls. The best utilization of the network was found to be provided by G.711 with the VAD method. The second experiment showed the capability of the same network in the same circumstances to support 100 bidirectional calls. From these findings, we can conclude that the WiMAX BE class can support quality-sensitive applications within the bounds of the network range. This can replace the use of UGS and rtPS classes, which present their own problems in bandwidth utilization and additional delay, respectively.

Some network configuration parameters, though, need to be set to maximize capacity. During the simulation works, we found that the contention period needed to be increased to 16 slots in order to reduce the network access and bandwidth requests collisions. Other parameter adjustments were mentioned in Chapter 6.

The project and the results have raised several new questions that can be answered in future studies. If we are to deploy citywide WiMAX networks with the support of multiple applications, we need to know how do we build a larger network with multiple base stations to provide the best possible coverage and signal strength? How many users can a single BS support before performance is degraded? What is the effect if a user moves around town while making a call?

The biggest challenge that faced the project was to choose the best simulator. During this work, we tried multiple simulators. With each one there was a steep learning curve to reach the point of reasonable competence. The OPNET WiMAX module worked the best, but there was no documentation for its use with the WiMAX model. This made it difficult to prepare the WiMAX scenario. Only after many trials were we able to determine all of the required settings for a successful simulation.

Overall, the project has given a good insight into the technical details of WiMAX and a good knowledge of the details of the OPNET simulator.

GLOSSARY

AAS	Adaptive antenna system
ADSL	Asymmetrical digital subscriber line
AP	Access point
ARQ	Automatic repeat request
ASK	Amplitude shift keying
BE	Best effort
BFSK	Binary frequency shift keying
BRR	Bandwidth request reiteration
BS	Base station
BW	Back off window size
BWA	Broadband wireless access
BW-REQ	Bandwidth request
CAC	Call admission control
CBR	Consistent bit rate
CID	Unidirectional connection identifier
CNG	Comfort noise generation
CPE	Customer premises equipment
CRC	Cyclic redundancy check
CS	Convergence sublayer

C_z	Contention Size
DCD	Downlink channel descriptor
DHCP	Dynamic host configuration protocol
DL–MAP	Downlink map
DSL	Digital subscriber line
ertPS	Extended real time polling service
FCH	Frame Control Header
FDD	Frequency division duplex
FES	Foreign exchange subscriber
FFT	Fast Fourier Transform
FSK	Frequency Shift Keying
FTP	File Transfer Protocol
FXS	Foreign Exchange Subscriber
GSM	Global system for mobile connections
GUI	Graphical user interface
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IP	Internet protocol
IPTV	IP television
ITU	International Telecommunication Union
LAN	Local area network

LoS	Line of sight
MAC	Medium access control
MAN	Metropolitan area network
MCU	Multipoint control unit
MOS	Mean Opinion Score
MPDU	MAC Protocol Data Units
MPEG	Moving Picture Experts Group
MSDU	MAC service data units
NLOS	Non–line of sight
nrtPS	Non–real-time polling service
OFDM	Orthogonal frequency division multiplexing
OFDMA	Orthogonal frequency division multiple access
OSI	Open system interconnection model
PBX	Private brand exchange
PC	Personal Computer
PCM	Pulse code modulation
PDA	Personal digital assistant
PDU	Protocol data unit
PHY	Physical layer
PMP	Point-to-multipoint
PPS	Packet Per Second

PSK	Phase shift keying
PSTN	Public switched telephone network
QAM	Quadrature amplitude modulation
QoS	Quality of service
rtPS	Real-time polling service
SDU	Service data unit
SF	Service flow
SIP	Session initiation protocol
SNR	Signal-to-noise ratio
SS	Subscriber substation
TCP	Transmission control protocol
TDD	Time division duplex
TDM	Time division multiplex
UCD	Up link channel descriptor
UDP	User datagram protocol
UGS	Unsolicited grant service
UIUC	Uplink interface use code
UL	Up link
UL MAP	Uplink map
UPQ	User-perceived quality
VAD	Voice activity detection

VoIP	Voice-over internet protocol
WiMAX	Worldwide interoperability for microwave access
WLAN	Wireless local area network
WMAN	Wireless metropolitan area network

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