**<u>QUALITY OF SERVICE IN MOBILE NETWORKS(WIMAX)</u>**

**DISSERTATION SUBMITTED IN PARTIAL FULFILMENT OF THE**

**REQUIREMENT FOR THE DEGREE OF**

**MASTER BY ADVANCED STUDY IN *(COMPUTING)***

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and to all my friends too numerous to mention, thank you, only God can reward appropriately, I can't.

# ABSTRACT

Project Title: QoS In Mobile Networks (Wimax)

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Quality of service (QoS) can be said to be the collective effect of service as perceived by the user. Also, it can be said to be the quality of service guarantee for the successful delivery of multi-media network traffic. Recently the landscape of mobile communication has continued to change rapidly.

Quality of service (QoS) has become an important factor in the provision of a variety of services using network resources. Services such as voice over IP(VoIP),and varied multimedia services.

This dissertation will be looking at the quality of service in mobile networks using wiMAX as our sample network, by the simulation of mobile traffic using a java simulation software, precisely at the effect of different types of traffics on mobile network, picking wiMAX (as it is a first –mile service) and its popularity and expected acceptability by telecommunications companies worldwide in the not too distant future, as our sample network.

Attempts were made in simulating various application services such as VoIP, Video, and multimedia (web/streaming) using their respective suggested bandwidth as recommended by the international telecommunication union and the quality of service (Qos) parameters such as throughput, average delay/delay jitter and packet loss/ blocking probability , on the simulated network were recorded and analysed . The result show that video traffic has the most effect negatively on the network, the average delay figures for video traffic were the highest, equally the blocking probability figures for the video traffic did not show any decline with rise in number of stations as the multimedia traffic did, but

rather it continued to rise to the highest figure for blocking probability of all the traffic types simulated. As delay and dropped packets leads to all sorts of ancillary costs associated with running a network which in this case will go up.

I hereby conclude that video traffic has the most negative effect on the network, followed by multimedia traffic and the traffic with the least effect on the network is VoIP traffic.

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# CHAPTER ONE

## 1.1: INTRODUCTION

The wireless communication industry is in a constant state of flux, currently the industry is in transition between 2G and 3G with the intention to converge everything over IP (4G and beyond). With the increase in services which can be offered, service assurance is now an integral part of every mobile network.

Wimax (Worldwide Interoperability for Microwave Access) which is its commercial name, was built with quality of service in mind[9,11.p161], hence the choice of wiMAX as the sample network, coupled with the fact that it could be merged easily with the main mobile wireless systems, such as GSM/GPRS/EDGE, UMTS, CDMA2000 providing top notch first mile or backhaul carrier service [11] and can be used as an ad hoc mobile network in disaster areas and locations such as Africa where very little infrastructure is in place (hence my interest) and the demand for broadband connectivity is set to greatly soar in the years to come. wiMAX is actually a part of the standard referred to as the IEEE 802.16 family of wireless broadband

standards with the newest in the series being the IEEE 802.16-2009,the most popular is the IEEE 802.16e-2005 referred to as mobile wiMAX.

This standard defines the MAC (Media Access control layer) (layer 2) and the PHY (Physical layer) (layer 1) of the Open Systems Interconnection (OSI seven layer network model) .These two layers will be my focus mainly. Network architecture layer (3-7) above layer 1 and 2 is not defined by this standard (802.16e-2005). Operators will continue to have the need to use equipment from different vendors and connect to other operators using totally different vendors.

The wiMAX forum was formed primarily to promote the standard (IEEE802.16), interoperability of broadband wireless products, based on the harmonised IEEE 802.16 standard. This forum (wiMAX forum) is to ensure the standard is introduced in the marketplace and ensure a bridging between vendors and operators so as to create an end to end wireless broadband network.

At the moment the service flow for each type of application service differs one to the other, for that particular application service to be successfully delivered end to end to clients across networks.

QoS in mobile networks using wiMAX as the network of choice, looked at the requirement for each application service to be successfully delivered and how it affects the networks ability to do same using java simulation

Typical application service includes VoIP, Video, T1/E1 Transport, FTP, HTTP EST.

## 1.2: HISTORY

The first proposal to use cellular systems in the field of mobile communications was conceived in 1947 by D.H. Ring at Bell labs. [15]

*First generation*: (1G) mobile networks were first introduced commercially in the early 1980's, these systems were based on frequency division multiple access (FDMA) and analogue signal processing, even then it had handover and in some of the technologies international roaming. Three of these systems were particularly successful

1) Nordic Mobile Telephone (NMT), which was developed in Scandinavia and readily adopted in a few European countries.
2) Advanced Mobile Phone System (AMPS) developed in the USA, used mainly in the far East, outside the USA.

3) Total Access Communication System(TACS) a UK derivative of AMPS mainly used in regions overlapping where NMT and AMPS was already in use, These 1G networks had lots of disadvantages, like high cost to the network subscriber, security issues,huge

.

*Second generation 2G*: Was mainly digital, using error correction and compression technologies and small cell size allowing them to use radio spectrum more efficiently, of course with higher capacities. GSM is the most successful, originally designed for voice, uses both FDMA and TDMA, better security (encryption of data and signalling message) the use of a *subscriber identity module* (SIM) was introduced, *general packet radio service* (GPRS) (2.5G) was equally introduced to increase the capacity of GSM networks to carry data efficiently, by building packet switched networks side by side with the circuit switched network already in place. *Enhanced Data Rates for Global evolution* (EDGE) was introduced in the American market as a result of frequency allocation issues, this simply modifies the modulation scheme of GSM to increase bit rate per user.

The other 2G system is the *cdmaOne (*IS-95), widely used in the USA and the far-east, first to use

CDMA technology and it’s an offshoot of AMPS. It has enhancements like the IS-95B, for the support of packet data at rates up to 64kbps, also classified as 2.5G like GPRS

*Third generation 3G:* were mainly characterised by changes to the air interface that support higher bit rates.

The international Telecommunication Union (ITU) in 1997 published a set of requirements for 3G communication technology under the name of IMT-2000.

The four systems that met these requirements are:

*1) Universal Mobile Telecommunication system* (UMTS):

Compactable with GSM edge to edge, but cannot operate at the same frequency uses a higher signal bandwidth 5MHz equally known as *wideband* CDMA (W-CDMA). It’s been up-graded since its introduction, high speed packet access (HSPA) known as 3.5G was introduced to increase the bit rates on the air interface in-case of non-real time packet data.

*2) Cdma2000* is a direct upgrade of *cdmaOne*, the two systems *cdmaOne* and *cdma2000 can co-exist side by side, using the same carrier frequency*. Two variants exist *cdma2000 1x RTT (1x Radio Transmission Technology):* uses *64* more codes to run along the original of *cdmaOne* to achieve double the capacity. *Cdma20001x EV-DO (1x Evolution Data Optimised):* uses separate 1.25MHz carriers for voice and data.

3) Time division synchronous code division multiple access (TD-SCDMA), it’s a variant of UMTS, equally known as TDD low chip rate option, it’s been developed in China.

In the year 2007, the ITU approved another system as compatible with IMT-2000:Worldwide Interoperability for Microwave Access (wiMAX).It uses a different technology from earlier 3G systems called *orthogonal frequency division multiple accesses* (OFDMA), at its

beginning (IEEE 802-16-) it looked nothing like a cellular technology, but with advancement to IEEE 802.16e (2005) or *mobilewi*MAX.support of crucial aspects of mobility such as moving devices, roaming and handover are in place. [10]

## 1.3: QUALITY OF SERVICE

In the year 1994 the International Telecommunication Union (ITU) recommendation E.800 [2-page 2], attempted to define QoS in telecommunication. This definition is very broad, listing 6 components:

support, Operability, Accessibility, Retainability, Integrity and Security.

In 2008, the ITU published a document discussing QoS in the field of telecommunications called ITU-T Rec series E: 800: (overall network operation,telephone service, service operation and human factors) this says the term Quality of Service refers to the probability of the telecommunication network meeting a given traffic contract, end to end, which also states that QoS is affected by both network performance and non-network performance factors. Therefore, QoS is a complex matter to determine accurately as this recommendation rightly shows often times the QoS requirement and the perceived QoS often are two different which don't always match on the customer side, as well as the provider who often times have a significant difference in the QoS offered and the Qos the provider is actually, able to provide.

In the field of packet-switched networks and computer networking it is used informally to refer to the probability of a packet succeeding in passing between two points in the network. Although the name suggests that it is a qualitative measure of how reliable and consistent a network is, there are three (3) basic parameters that could be used to measure it quantitatively. These include throughput, packet delay, packet loss.

The use of different kinds of applications in a network, results in heterogeneous traffic load. The traffic from different applications may require certain type of quality of service, for example:

- streaming multimedia may require guaranteed throughput
- IP telephony or Voice over IP (VOIP) may require strict limits on average delay/jitter
- Video teleconferencing (VTC) requires low jitter
- Dedicated link emulation requires both guaranteed throughput and imposes limits on maximum delay and jitter

• A safety-critical application, such as remote surgery may require a guaranteed level of availability (this is also called hard QoS). Packets travelling across a wireless network, in this case say wiMAX, might experience some if not all of the following problems.

• Delay: some times referred to as latency is the unpredictably longer time it takes for packets to reach the Destination due to unavailability of network resources.

Delay-*Jitter* - Packets from source will reach the destination with different delays. This variation in delay is known as jitter and can seriously affect the quality of streaming audio and/or video.

*Jitter*: also referred to as out-of-order delivery is When a collection of related packets are routed through the Internet, different packets may take different routes, each resulting in a different delay. The result is that the packets arrive in a different order to the one with which they were sent. This problem necessitates special additional protocols responsible for rearranging out-of-order packets once they reach their destination.

• Packet loss or Error rate: is the failure of the network to deliver all the packets transmitted to a recipient. As many packet protocols contain provisions at the receiving end if there is a severe packet loss, for the sender to resend. [16-p163]

## 1.4: QoS MECHANISM

QoS support has become essential to all networks, as they need to deliver real-time services like video, audio, and voice over IP. Essentially two ways exist in making provision for guarantees.

The first is to simply overprovision bandwidth resources, enough to meet the expected peak demand with substantial safety margin.This approach generously over provisions the network and is expensive and its success is dependent on the number of users being small at all times. All packets get a quality of service sufficient to support applications sensitive to QoS. This approach is relatively simple, but expensive in practice.The second approach is the use of ancillary IP protocols (such as RSVP,IntServ,DiffServe etc.) for Qos guarantees by requesting for a certain amount of bandwidth from the network . To provide QoS support in IP layer there are two popular methods:

1. Integrated Services (IntServ) [18,19]: is a model used for providing traffic forwarding service in networks. for IntServ it's a router function which creates the QoS for each traffic flow, this router function is called "traffic control", this is executed using three components .

The classifier, the packet scheduler and the admission control.Applications requiring guaranteed service or controlled-load service must set up the paths and reserve resources before transmitting their data.

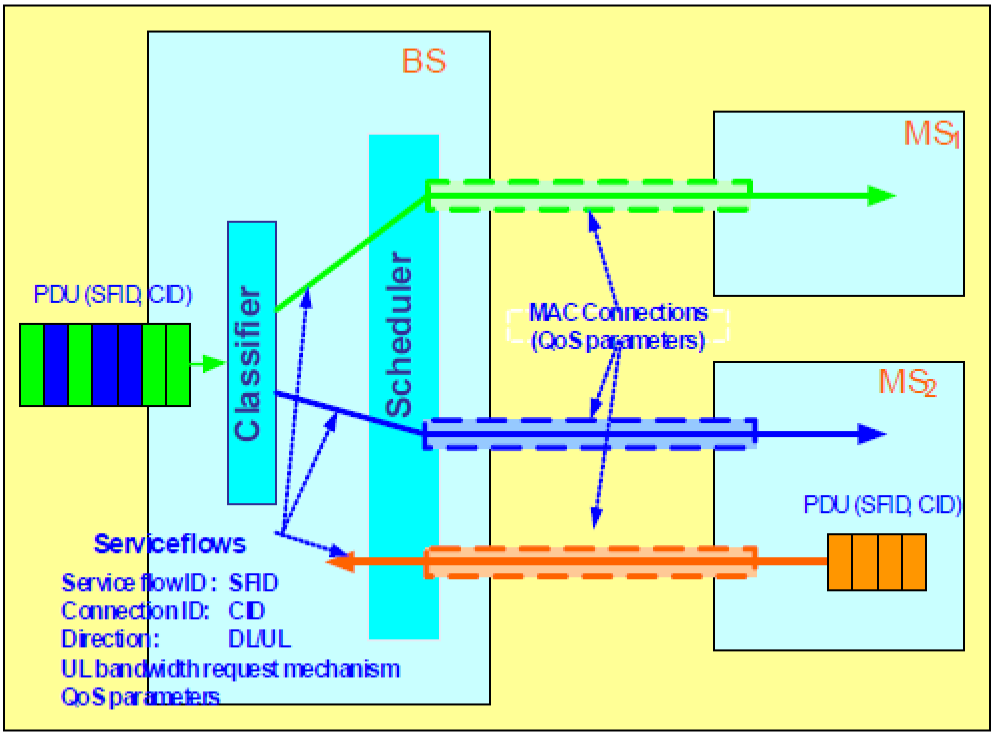


**Fig 1: Mobile wiMAX Qos Support [23]**

*The classifier*: for traffic control purposes, all incoming packets are mapped by the classifier into a class, and all packets get same treatment, mapping into a class is carried using the contents of the headers and /or some classification number added to each packet. e.g. all video flows are mapped as such and treated.

*The packet Scheduler*: Usually implemented where packets are queued. It has the responsibility of forwarding various packets streams using a set of queuing algorithm and maybe timers. After classification of packets in a specific queue, the packet scheduler will then schedule the packet to meet its QoS requirements

*Admission Control:* Implements a decision algorithm for a router or host in granting a QoS request to a new flow without contradicting an existing guarantee. Usually invoked at every node and must be consistent with the service model of any flow type. The problem with InterServ has to do with its scalability. It does not maintain a consistency in performance as the network size increases.

2. Differentiated Services (DiffServ) [16, 20]: In use more these days. It is a computer networking architecture that is simple and scalable. Basically It relies on the classification of packets rather than

the reservation of bandwidth in network management, in its effort to provide the required QoS for any one service flow type. Service flow types were defined by IEEE802.16 set of standards. The MAC layer is responsible for allocating bandwidth to subscribers. The MAC layer allocates bandwidth according to the subscriber request and the relevant QoS associated with providing the service flow able to deliver end-to-end the application service which the subscriber has requested for. These service flow types have been listed below, with the relevant services associated with each service flow type.

**Table 1 wiMAX /802.16 QoS Model [13, 23]**

| **Qos Category** | **Applications** | **QoS Specification** |
|---|---|---|
| **UGS**<br>Unsolicited Grant Service | VoIP | ☒ Maximum Sustained Rate<br>☒ Maximum Latency Tolerance<br>☒ Jitter Tolerance |
| **rtPS**<br>Real-Time Polling Service | Streaming Audio or Video | ☒ Minimum Reserved Rate<br>☒ Maximum Sustained Rate<br>☒ Maximum Latency Tolerance<br>☒ Traffic Priority |
| **ErtPS**<br>Extended Real-Time Polling Service | Voice with Activity Detection (VoIP) | ☒ Minimum Reserved Rate<br>☒ Maximum Sustained Rate<br>☒ Maximum Latency Tolerance<br>☒ Jitter Tolerance<br>☒ Traffic Priority |
| **nrtPS**<br>Non-Real-Time Polling Service | File Transfer Protocol (FTP) | ☒ Minimum Reserved Rate<br>☒ Maximum Sustained Rate<br>☒ Traffic Priority |
| **BE**<br>Best-Effort Service | Data Transfer, Web Browsing, etc. | ☒ Maximum Sustained Rate<br>☒ Traffic Priority |

**Description of Applications**

These service flows can be created, changed, or deleted by the issuing dynamic Service Addition (DSA), Dynamic Service Change (DSC), and Dynamic service Deletion (DSD) messages. Each of these actions can be initiated by the subscriber Station (SS) or the Base Station (BS) and are carried out through a two or three-way-handshake. For example, a new service flow initiated by the SS is built as follows. When SS detects the emergence of a new service flow, it will calculate the available resources to determine whether a DSA request will be sent or not. Upon reception of the DSA request, the BS verifies whether this request can be supported, and sends a DSA response. Based on the DSA response, the SS sends a DSA acknowledgement and enables the new service flow if the request is approved. The standard provides some rules to classify DiffServ IP packets into different priority queues based on IP QoS indication bits in IP header. So, in general, the QoS Architecture of IEEE 802.16 can support both IntServ and DiffServ.

## 1.5: PROJECT BACKGROUND

This project's main objective is to investigate quality of service (QoS) routing in mobile networks .The project is intended to not only investigate quality of service in networks, but also to identify the techniques for quantifying the level of service quality.

To achieve the above objective, I decided to use a Java simulation package originally designed to depict a wi-fi network as the uplink pattern is similar to that of wiMAX in its simple form where the subscriber stations communicate with the base station. Most wiMAX simulations had being done in ns-2 or opnet, which was my initial choice for simulation software, but these were not readily available for use. QoS implementation in the wiMAX Network architecture will be presented. This includes various service flows as defined by the IEEE 802.16 Standard .MAC layer details showing service flow types will be presented, as this is a major factor in the implementation of QoS in wiMAX. After due consideration and consultation Java became the simulation software of choice, which was stated earlier in these paragraph, as it was easier to adjust and I could exercise more control over the investigation being carried out. Data relating to expected real life usage was investigated. The target is to investigate service flows for varying application services e.g. VoIP etc and how these application services affect the network QoS parameters such as throughput, delay/jitter, packet loss of the network.

## 1.6: PROJECT WORK

A modified Java module earlier used by C. Xia[26] was used to simulate and analyze the effect of various applications service traffic on network QoS parameters such as throughput, delay/ jitter and packet loss/packets blocked. These above mentioned QoS parameters will be considered.

Use cases simulated VoIP, video, and multimedia (e.g. web) traffic. The VoIP traffic is set up using VoIP codec (G.711), the video traffic is set up using H.262 codec and the web traffic to depict multimedia streaming is equally set up. Also the outcome of a given number of nodes in the network with VoIP traffic is analyzed. The experiment is repeated for both video and web traffic packets. The effect of the traffic types (VoIP, video, web etc.) on the quality of service parameters such as throughput, delay/jitter and packet loss is studied .The results of the simulation were analyzed and conclusions are presented.

## 1.7: PROJECT STRUCTURE

Chapter one of this thesis gives a brief introduction to IEEE 802.16/WiMAX networks. The quality of service mechanism in wiMAX networks is introduced and the QoS model as set by wiMAX were presented. The background of this project was further stated and more clearly defined. The project work actually to be carried out and how I intend to do this is equally clearly stated. Chapter two presents details of the wiMAX network architecture. QoS known basically to be implemented at the MAC layer the wiMAX networks. The details of the MAC layer in the wiMAX networks are looked at followed by related recent research in this area. Chapter three describes the simulation set up, the simulation environment used. The parameters that indicate QoS are described in details and the equations used to calculate the parameters are presented. This is followed by the description of various simulations performed. Chapter four presents the results obtained. The parameters that indicate QoS is looked at for varying application services using readily available service flow mechanism. The last chapter presents the conclusions drawn from the work performed and also gives insight into the future work. The literatures referred in preparing this thesis is listed in the reference section of every chapter

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# CHAPTER TWO

This chapter presents in detail the wiMAX network architecture. The implementation of QoS mechanism in the MAC layer follows. The various types of service flow as defined by wiMAX are described . Finally, the literature analysis on QoS is presented.

## 2.1: wiMAX NETWORK ARCHITECTURE

The wiMAX End-to-End Network Architecture is based on an all IP platform. An all packet network with no circuit switched network [12-page40-41].It's an architecture that logically represents The mobile wiMAX network architecture. This architecture highlights functional entities and reference points over which interoperability is achieved. This architecture was developed with the purpose of providing unified support of functionality needed in a range of network deployment models and usage scenarios.*Figure 2* shows components of a wiMAX network topology. The user terminals are connected over the air interface to the base station (BS). The base station is part of the Access Service Network (ASN) and connects to the Connectivity service Network (CSN) through the ASN Gateway. In generic telecommunication terminology, ASN is equivalent to RAN (Radio Access Network) and CSN is equivalent to Core.

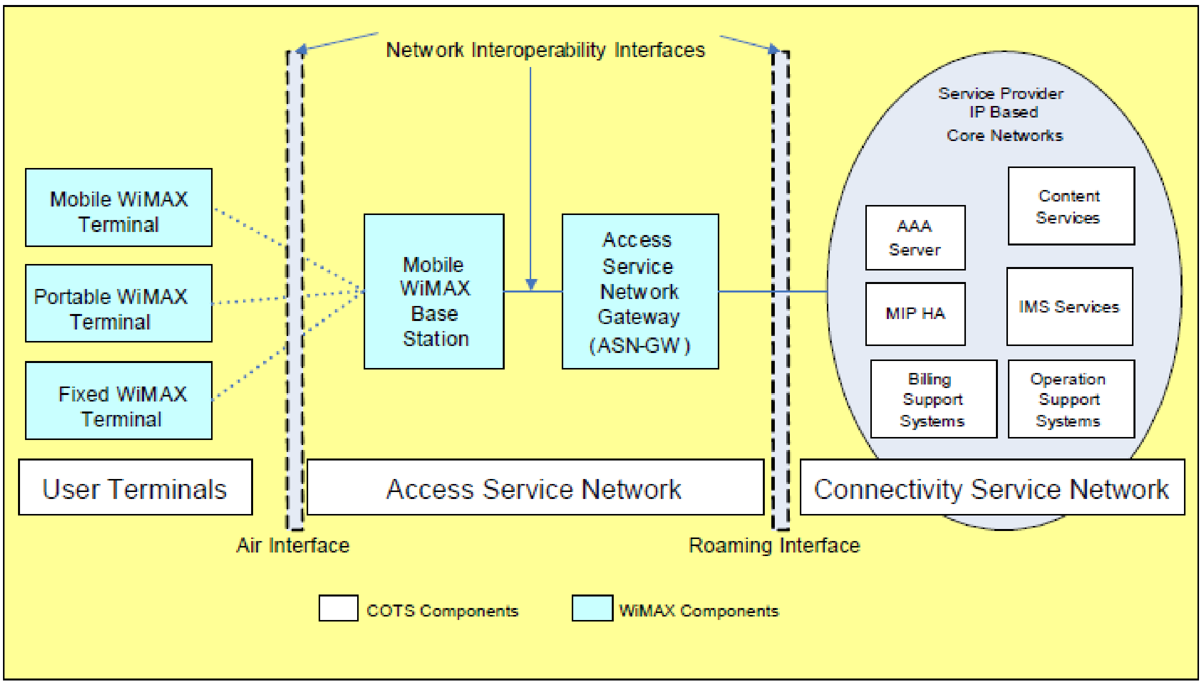


**Figure 2: wiMAX Network IP-Based Architecture [12]**

The access service network (ASN) performs the following main functions:

A) wiMAX Layer 2 and Layer 3 connectivity with the subscriber stations, including IP address allocation

B) Network discovery and selection of an appropriate network service provider that the subscriber station accesses

c) Radio Resource Management

The connectivity service network (CSN) performs the following main functions:

a) Internet access

b) Authentication, Authorization and Accounting

c) Policy and admission control based on user subscription profiles

## 2.2: The MAC layer In Some detail

The MAC layer provides a medium independent access to the PHY layer.

The MAC is connection- orientated.

The MAC layer is divided into three sub-layers.

- CS(Convergence Sub-layer)
- MAC CPS (Common-Part Sub-layer)
- Security Sub-layer.

*The CS Sub-layer: is* responsible for receiving PDUs (Protocol Data Units) from the higher layers, classifying these PDUs to appropriate connections, processing these PDUs and delivering them to the appropriate MAC-SAP (Service Access Point).

*The MAC- CPS (Common Part Sub-layer):* provides the fundamental MAC functionalities including connection establishment and management, generation of MAC management messages, SS initialization and registration, ranging, bandwidth management, service flow management and scheduling services.

*The Security-sub layer:* basically provides security of data by authentication, encryption and decryption between the subscriber station (SS) and base station (BS) [13 pg 2, 14 page 2].The MAC layer of wiMAX supports the

- Point to multipoint (PMP)

-and the mesh topologies.

wiMAX uses *Orthogonal Frequency Division Multiple Access* (OFDMA). The standard supports two duplexing techniques namely *FDD*, where downlink (DL) and uplink (UL) sub frames take place at the same time but on different frequencies, and *TDD*, where DL and UL sub frames take place at different times and usually share the same frequency.

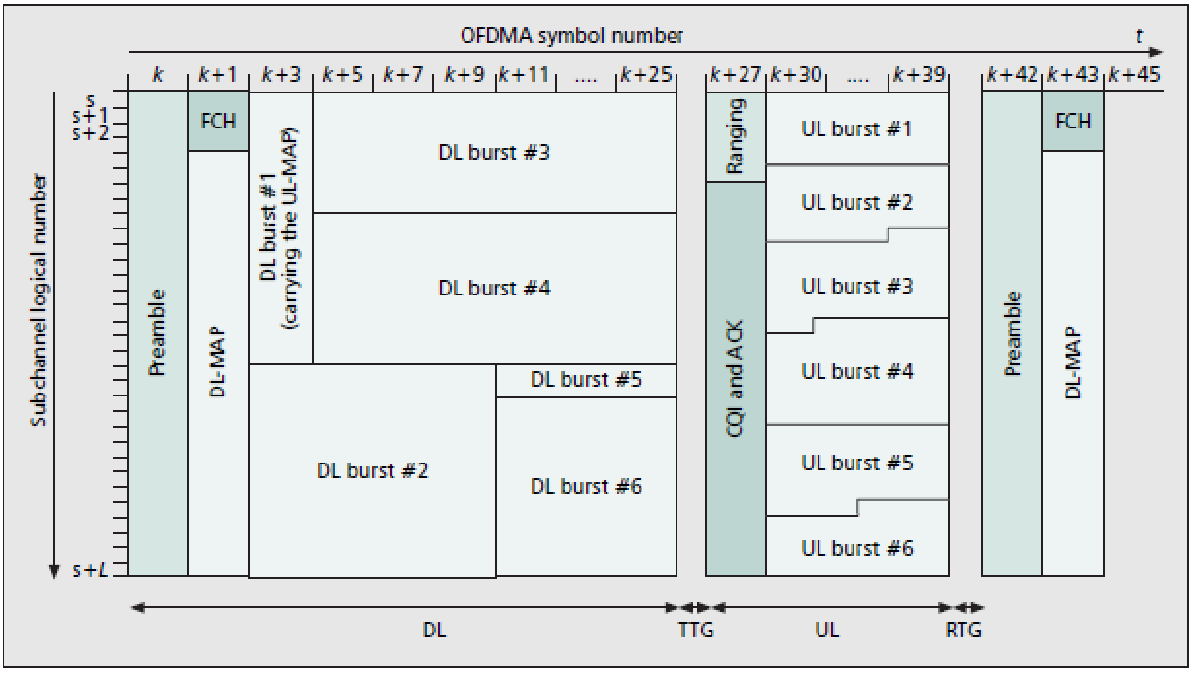


**Fig 3a: Mobile wiMAX OFDMA/TDD frame structure. [12a]**

Scalable OFDMA(S-OFDMA) was introduced with the IEEE 802.16e-2005 amendments in order to support scalable bandwidth channels.

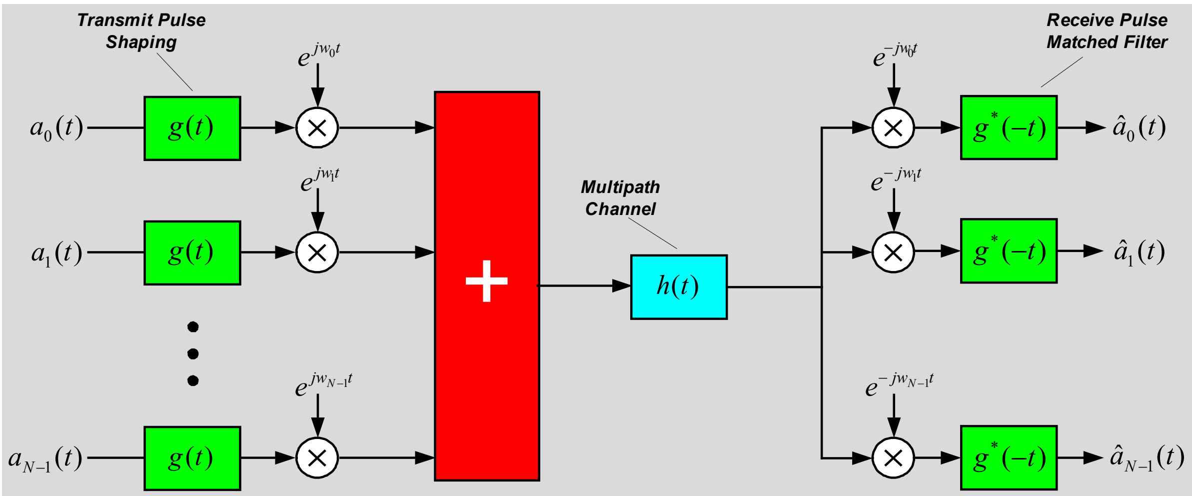


**Fig.3 Basic structure of OFDM [12]**

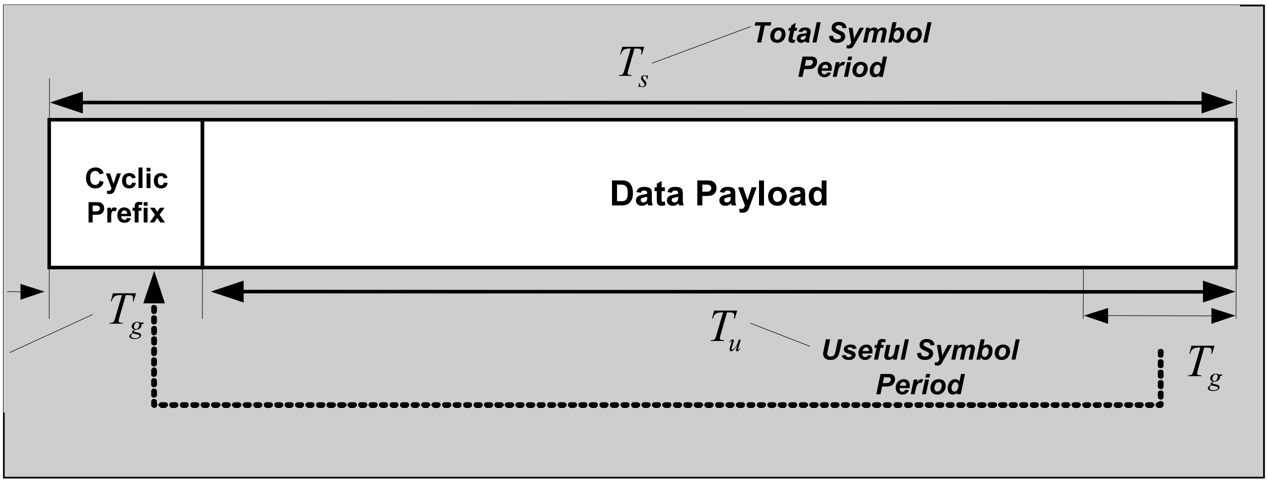


**Fig.4 OFDMA Packet Structure. [12]**

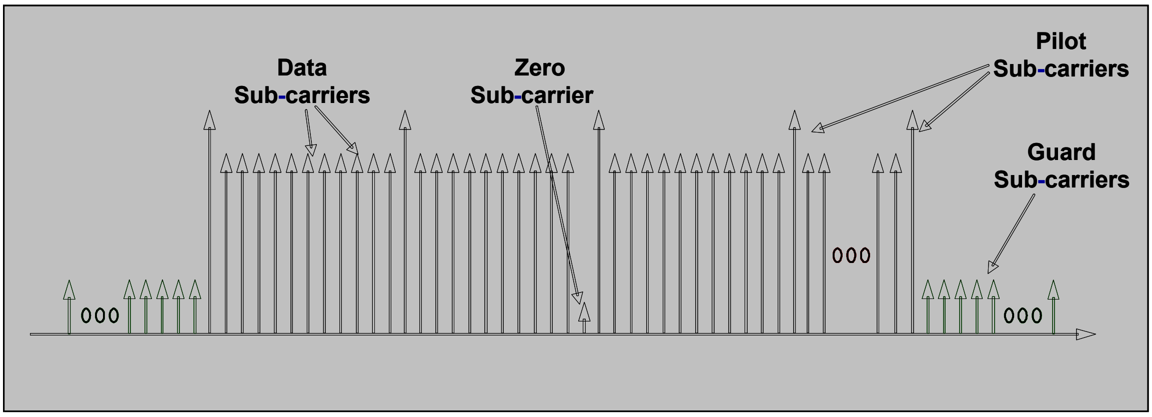


**Fig 5 OFDMA Symbol Structure and Sub channelization [12]**

**Key features of wiMAX are [12]**

***High Data Rates***: in a 10 MHz channel the addition of MIMO (Multiple input, Multiple output) antenna techniques along with flexible sub-channelization schemes, advanced Coding and modulation has further enabled the Mobile wiMAX technology to support DL data rates of up to 63 Mbps at a maximum and UL data rates up to 28 Mbps at a maximum.

***Mobility*****:** Mobile wiMAX supports optimized handover schemes with latencies less than 50 milliseconds to ensure real-time applications such as VoIP perform without service degradation. Flexible key management schemes assure that security is maintained during handover.

***Security*****:** Is a key feature of this technology. The security sub-layer is strategically positioned Between the MAC and PHY layers, some of the security features provided for in Mobile wiMAX security are EAP-based authentication, AES-CCM-based authenticated encryption, CMAC and

HMAC based control message protection schemes and diverse set of user credentials are supported; such as SIM/USIM cards, Smart Cards, Digital Certificates, and Username/Password schemes based on the relevant EAP methods for the credential type.

***Scalability**:* Despite an increasingly globalized economy, spectrum resources for wireless broadband worldwide are still quite disparate in its allocations. Mobil wiMAX technology is therefore essentially designed to be able to scale to different channelizations from 1.25 to 20 MHz to comply with varied regional requirements. This also allows different types of economies to realize the multi-faceted benefits of the Mobile wiMAX technology for their specific geographic needs such as providing affordable internet access in rural areas as well as enhancing the capacity of mobile broadband access in metro and suburban areas for both telecommunication and emergency relief services.

***Quality of Service (QoS):*** **[13]** is a fundamental aspect of the IEEE 802.16 MAC architecture. It defines Service Flows as a unidirectional flow of MAC PDU's on a transport connection that ensures specific QoS requirements. A service flow is identified by a 32-bit Service flow ID (SFID) and is associated to exactly one transport connection and QoS parameter set. The *service Flow Manager* component at the BS is responsible for creating and managing service flows.

The standard defines two mechanisms of service flow creation: a SS-initiated mechanism where SS requests the BS to create a service flow, and a BS-initiated mechanism where the BS automatically creates the service flow for the SS.

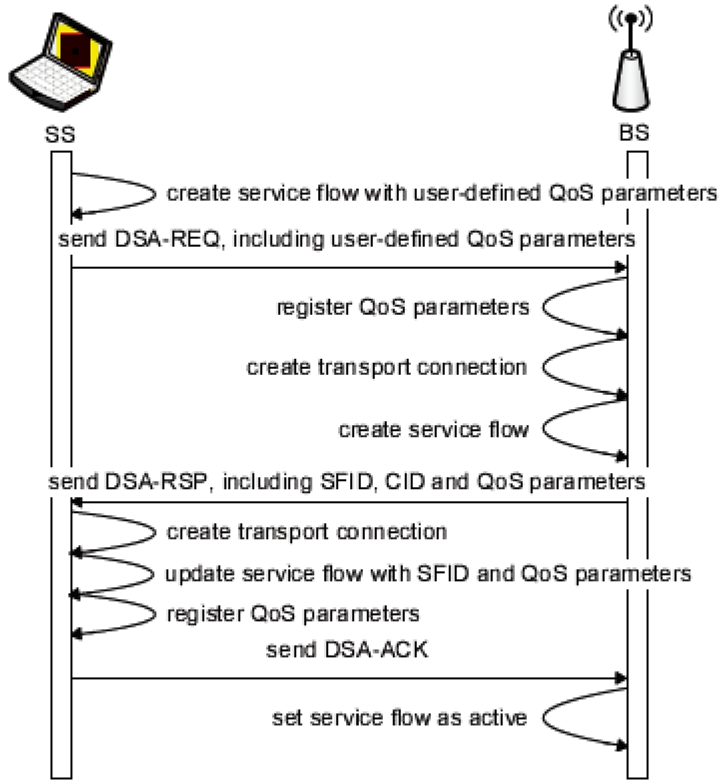


**Fig 6: Service Flow creation process. (SS initiated)**

The main focus of the MAC layer is to manage the resources of the air-link in an efficient manner. MAC layer is responsible for overall connection and session processing. The MAC layers at BS and SS communicate to set up an RF connection, and to set up, add and delete services on an as needed basis. The IEEE 802.16 MAC protocol is designed to support two network models.

-Point to Multipoint (PMP)

-Mesh Network Model

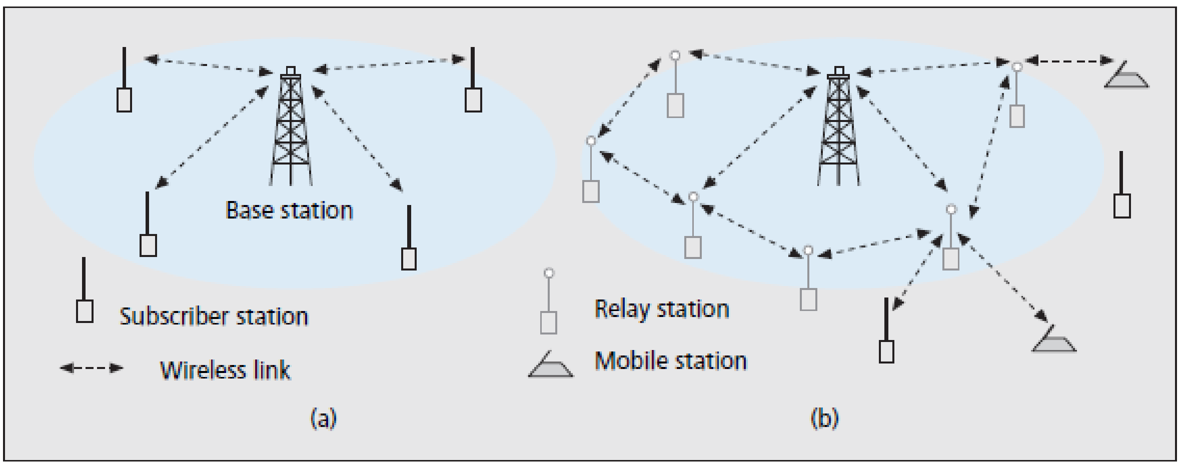


**Fig 7:** ***wiMAX Network architectures a) PMP model) mesh mode***

In the PMP mode, the nodes are organized into a cellular like structure, consisting of a base station (BS) and some subscriber stations (SS). The channels are divided into uplink (from SS to BS) and downlink (from BS to SS), both shared among the SS's. This type of network requires all subscriber stations to be within the transmission range. The IEEE 802.16 MAC protocol is connection oriented. Upon entering the network, each SS creates one or more connections over which their data are transmitted to and from the Base Station (BS). The MAC layer schedules the usage of the air link resources and provides Quality of Service (QoS) differentiation.

In the mesh mode, the nodes are organized in an ad-hoc fashion. All stations are peers and each node can act as routers to relay packets for its neighbours. In typical installations, there still can be certain nodes that provide the BS function of connecting the mesh network to backhaul links. However, there is no need to have direct link from SS to the BS of the mesh network. A node can choose the links with the best quality to transmit data; and with an intelligent routing protocol, the traffic can be routed to avoid the congested area. In this thesis, only PMP mode is to be considered.

In PMP mode, uplink (from SS to BS) and downlink (from BS to SS) data transmissions occur in separate time frames. In the downlink sub frame, the BS transmits a burst of MAC protocol data units (PDUs). Since the transmission is broadcast, all SSs listen to the data transmitted by the BS. However, an SS is only required to process PDUs that are addressed to it or that are explicitly intended for all the SSs. In the uplink sub frame, on the other hand, any SS transmits a burst of MAC PDUs to the BS in a time-division multiple access (TDMA) manner. SSs can be either full duplex (i.e., they can transmit and receive simultaneously) or half-duplex (i.e., transition and reception is at different time intervals).

The MAC protocol is connection-oriented. All data transmissions take place in the context of connections. A connection is a unidirectional logical link between

the MAC layer on the BS and the MAC layer of the SS. A service flow is mapped to a connection and the connection are associated with a level of QoS. Connections in the downlink direction is either unicast or multicast while uplink connections is always unicast. During initialization of an SS, three particular connections are established in both directions. The basic management connection is used for short time critical messages(C-SAP primitives). The primary management connection is used to exchange longer more delay tolerant messages (both C-SAP and M-SAP primitives). Finally the secondary management connection is intended for higher layer management messages and SS configuration data (M-SAP primitives). For actual user traffic, transport connection ID's are used. For each active service for a user, two connection ID's are created. Service flows may be provisioned when an SS is installed in the system.

Shortly after SS registration, transport connections are associated with this service flows. The outbound MAC then associates packets traversing the MAC interface into a service flow to be delivered over the connection. The QoS parameters associated with the service flow define the transmission ordering and scheduling on the air interface. The connection-oriented nature of QoS can provide accurate control over the air interface. Since the air interface is usually the bottleneck, the connection-oriented QoS can effectively enable the end-to-end QoS control. The service flow parameters can be dynamically managed through MAC messages to accommodate the dynamic service demand. The concept of a service flow on a transport connection is central to the operation of the MAC protocol. Service flows provide a mechanism for uplink and downlink QoS management. In particular, they are integral to the bandwidth allocation process. An SS requests uplink bandwidth on a per connection basis(Implicitly identifying the service flow through the connection ID). Bandwidth is granted by the BS to an SS as an aggregate of grants in response to per connection requests from the SS. wiMAX supports a wide range of data services and applications with varied QoS requirements.

The IEEE 802.16 standard has defined five service flows classes which have different QoS requirements:

Unsolicited Grant Service (UGS),real-Time Polling Service (rtPS), non-Real-Time Polling Service (nrtPS), Enhanced-real-Time Polling Service (ertPS) and best effort (BE). Each scheduling service is characterized by a mandatory set of QoS parameters, which is adapted to best describe the guarantees required by the applications that the scheduling service is designed for. Furthermore, for uplink connections, it also specifies which mechanisms to use in order to request bandwidth.

***UGS (Unsolicited Grant Service)*** is designed for real-time applications with fixed packet size. It guarantees bandwidth on real-time basis, by allocating the flow UL grants on regular basis. The key parameters for this service are *tolerated jitter*, *maximum latency* and *minimum reserved traffic rate*.

The *tolerated jitter* parameter specifies the maximum tolerated delay variation and *maximum latency* specifies the maximum delay between arrival and forwarding of the packet (at the BS side).

The *Minimum reserved traffic rate* parameter specifies the minimum bandwidth (in bps) that must be reserved for the flow. These parameters are supplied by the user while setting up the service flow. Based on these parameters, the BS calculates the interval, called *unsolicited grant interval*, when the service flow must be allocated a grant. The *Downlink scheduler*, while scheduling UGS traffic, must ensure that the latency requirements of the flows are met. Every time the scheduler is invoked, the queues are refreshed and packets whose latency deadlines have elapsed is dropped.***rtPS (Real-Time Polling Service***) is designed for real-time applications with variable packet size, and is assured UL grants on periodic basis. For a rtPS flow, the BS polls the SS with a special grant, referred to as *request opportunity*, allowing it to send bandwidth request. The BS in response allocates sufficient bandwidth in the subsequent frames. This mechanism enables the SS to specify the size of the grant, to meet flow's variable size traffic needs. However, compared to UGS, rtPS incurs more overhead due to the bandwidth request packets.

The key parameters for rtPS service are *maximum latency* and *minimum reserved traffic rate*.

The BS, based on these parameters, calculates the polling interval (*unsolicited polling interval*). The *Uplink Scheduler* ensures the flow of the minimum bandwidth. Like UGS, the *downlink Scheduler* makes sure that the latency requirements of the flow are met.

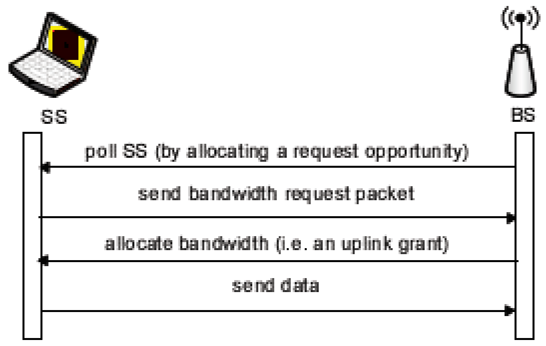


**Fig 8 Bandwidth Request/Grant Process (BS initiated)**

***nrtPS scheduling*** the only difference between rtPS and nrtPS (Non-Real Time Polling service) is that the latter is designed for non real-time applications. nrtPS flows are not polled on periodic basis and instead only when sufficient bandwidth is available. SS then specifies the size of the grant through the bandwidth request packet and BS responds by allocating sufficient bandwidth only when available. However the BS ensures the flow of the minimum bandwidth (based on *minimum reserved traffic rate* parameter) by periodically monitoring total bandwidth granted to the flow. Currently this period is set to one second. nrtPS does not use the *maximum latency* parameter.

***BE Scheduling (Best Effort) service*** is designed for the low priority best effort applications. for a BE flow, the scheduler allocates bandwidth only when sufficient bandwidth is available after servicing higher priority flows. No minimum bandwidth is guaranteed for this service.

***ertPS (Extended rtPS)*** [16] was introduced to support variable rate real-time services such as VoIP and video streaming. It has an advantage over UGS and rtPS for VoIP applications because it carries lower overhead than UGS and rtPS.

| | |
|---|---|
| UGS | Fixed packet size, user specifies minimum required bandwidth |
| rtPS | Variable packet size, Bandwidth allocation, using polling |
| ertPS | Time -Variable bandwidth allocation ,using polling |
| nrtPS | Fixed Bandwidth allocation, non-polling |
| BE | Bandwidth not guaranteed. |

**Table 2: Bandwidth allocation for service flows.**

## 2.3: Literature Analysis

The rapid growth in various wireless networks in recent years, has increased the demand for wireless data services and multimedia applications also over same media equally has seen increases. To ensure that the quality of service provided keeps up with the demand a lot of research in the field of QoS has been embarked upon all over the world. As the IEEE 802.16 standard did not address the scheduling algorithms and admission control issues, QoS research has sought to address this issue with many papers in recent journals. In this section, a brief summary of current work in the field of scheduling algorithms and admission control is presented.

Xiaojuan Xie et al [17] analyzed two representative wiMAX scheduling algorithms, and proposed a fair and Effective Queuing (FEQ) algorithm for wiMAX scheduling with a novel queuing analysis model. In this paper they conducted simulation studies for the FEQ algorithm to verify its effectiveness and the correctness of the queuing analysis model. Also they compared FEQ algorithmpsilas performance with two other representative wiMAX scheduling algorithms under poisson and busty traffic. Our simulation results show that FEQ algorithm has good performance in terms of fairness and packet drop rate, with the benefit of low algorithm complexity.

Ukil, A. et al [18] proposed priority indexed long-term (PILTPF) resource allocation algorithm, which dynamically allocates the OFDMA resources to the users to meet their QoS requirement, which is dependent on userpsilas, derived priority profile. It also considers the time diversity gain achieved in long-term computation of proportional fair (PF) metric. PILTPF algorithm emphasizes on individual userpsilas true priority in allocating system resources in order to maintain as well as optimize the requirements of different QoS classes, existing in the current and next generation broadband wireless networks like wiMAX.IEEE 802.16 standard defines the broadband wireless access specification for wiMAX, but it did not specify a standard scheme for scheduling algorithms and admission control.

As a result, Po-Chun Ting; Chia-Yu Yu; et al [19] proposed an uplink scheduling scheme called random Early Detection based Deficit Fair Priority Queue (RED-based DFPQ) for wiMAX. RED-based DFPQ adjusts the deficit counter of rtPS based on the current queue length. The proposed scheme claims to improve the transmission quality of rtPS traffic in wiMAX networks Ghazal, S.; Aoul, Y.H. et al [20] introduced new modules in both subscriber station (SS) and base station (BS), allowing more efficient handling of multi-service flows. They focused mainly on the design of a probabilistic and self-configuring AC algorithm, which prevents? Uplink and downlink congestions while guaranteeing QoS to rtPS and nrtPS flows respectively.

Chan, S. Yan Wang Haider . Et al [21] reviewed and compared various options for scheduling algorithms in wiMAX that are key to efficient operation of wiMAX technology. wiMAX multi-service environment is complex because of the various packet streams it needs to serve, their different quality of service (QoS) requirements and traffic behaviours. therefore, the packet schedulers operating at the MAC layer are very important for QoS delivery. Although these schemes may be applicable to wiMAX, there are still certain issues that require attention, e.g., how much information is required by the grant scheduler at the BS, or how to allocate time-slots, as well as sub-channels, to different SSs when orthogonal-frequency division multiple access (OFDMA) is used.

Jianfeng Song; et al [22] proposed a scheduling algorithm which could enhance the quality of service (QoS) provided by IEEE802.16e mobile wiMAX. In this paper, a cross-layer scheduling algorithm based on genetic algorithm (CLSAGA) under the network utility maximization (NUM) concept is proposed to allocate resources for each service flow. The real coded genetic algorithm is employed to solve NUM optimization problem. Adaptive modulation and coding (AMC) scheme and QoS category index of each service flow jointly decide the weights of utility functions to calculate the approach guarantees the QoS requirements and balances priorities of the mobile stations.

With this paper, Ahmad, I. Habibi, D. [23] proposed a mathematical model to estimate the bit error probability when the mobile station travels at different speeds (Multipath fading causes high bit error rate at the receiver end at high speed, which is a key reason for low throughput (*QoS factor*) at the receiver end). The estimated value of bit error probability is then taken into account to proactively compute the appropriate maximum packet size that offers the best chance to achieve improved throughput at different operating conditions. They simulated the proposed scheme for a centralized video surveillance system in a public train where the train is the mobile node and sends real-time video data to the base stations. The results show that the proposed scheme achieves significantly higher throughput and lower jitter compared to other standard schemes.

Looking to the future is the recent adaptation of satellite spectrum to be used for integrated terrestrial services in the footprint of the satellite derived from the recent FCC ancillary terrestrial component ruling. This is further explained by Ansari, A. et al [24].

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# CHAPTER THREE

## SIMULATIONS

In this chapter, the simulation setup used for experimentation is described. The setup used is further explained. The QoS parameters that are considered to analyze network performance are presented. It is followed by the descriptions of various simulations performed using the bandwidth designated for different flow types. The results obtained are presented in the next chapter, along with comments on each of the results.

Simulation network setup is also presented.

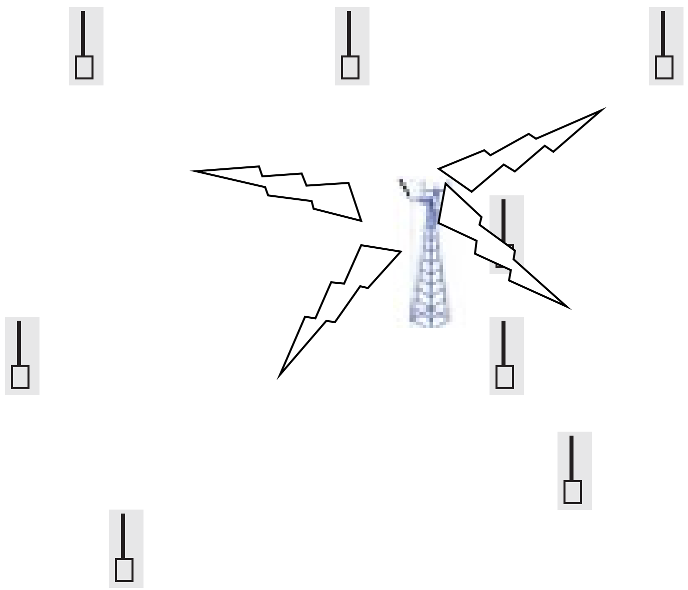

**Fig 9: Simulation set-up**

## 3.1: SIMULATIONS STRATEGY

This part basically explains the simulation environment in detail. The simulations were carried out by means of a prototypical simulator of the wireless LAN developed by C Xia[26] which was adequately tested and compared with opnet simulation software and found satisfactory, hence my confidence in its ability to provide approximate values, for my academic investigation with the queuing theorem being the foundation of its development, which equally apply to the wiMAX

setup, hence my choice of its usage. This simulator is a discrete event-simulator and was developed using J2sdk package based on the Java language [26] .Simulation set up would reflect the recommended size of packets, for VoIP, Video, and multimedia while travelling over the wiMAX network and their effects on the QoS performance metrics of the network such as throughput, delay/ jitter and blocked/lost packets.

Based on the network reference model described earlier in Chapter 2, Figure 9 shows the setup that will be used. There are 8 number SS's in the range of a base station.

The base station is connected to the core network. The focus of analysis will be the connection between the subscriber and the base station (Access Point) (AP).

## 3.2: DESCRETE EVENT SIMULATION [25]

Since the simulator to be used for this study is based on discrete events it's only proper to shed some light on exactly what this term really is here. Discrete event simulation is the process of modelling a system whose operation is a chronological sequence of events which occur at an instant in time and marks a change of state. A flow chart has been included that shows the chronological sequence of events in a discrete event simulation. Please find below.

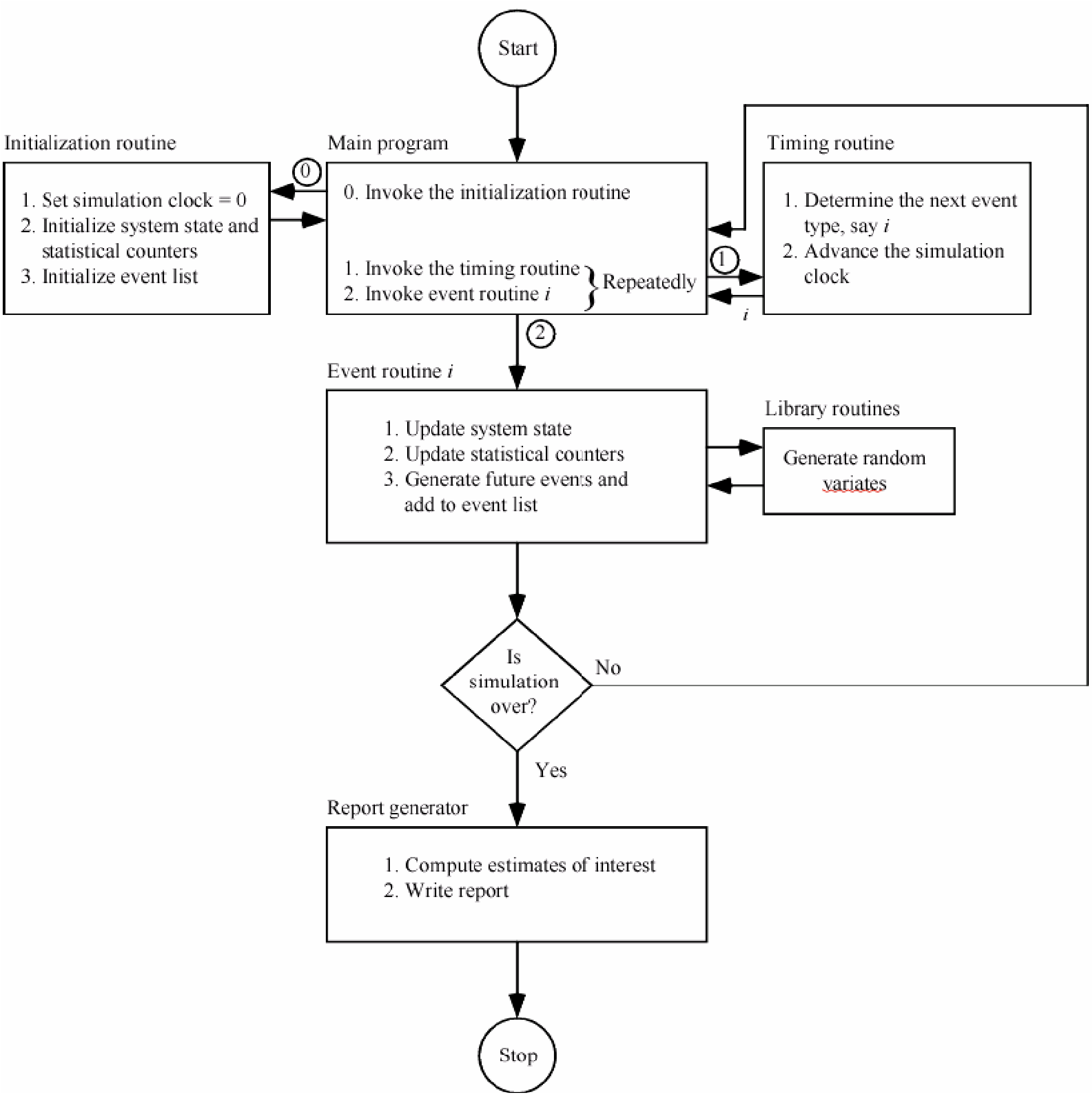


**Fig 10: Flow of Discrete Event Simulation**

## 3.3: THE BASIC MECHANISM — M/M/1/K & M/M/C/K QUEUING SYSTEMS

These queuing systems are widely applied in many areas to simulate and calculate a process which has a Poisson arrival and negative exponential service times? There are many kinds of queuing models, such as M/M/1, M/M/1/K, M/M/∞, M/M/C/C, M/M/C/K, M/G/1, etc. and M/M/C/K queuing systems.

I looked at two of these types of queues used in the simulation Java program

- ✓ M/M/1/K
- ✓ M/M/C/K

***M/M/1/K***

The *M/M/1/K* queue is a single server queue with a buffer/customer limit of size *K*.

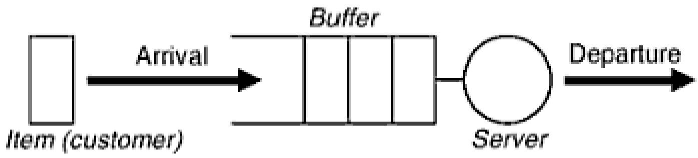


**Fig 11a: M/M/1/K Queue.**

If buffer/customer limit of size K is attained, the system begins to block other coming customers until there is free space in the queue. The demand for packet transmission is assumed to be a Poisson process, in networking and the buffer is used often as a finite queue. Therefore, the transmitting request in stations can be seen as an M/M/1/K queuing system. For M/M/1/K system, we can draw its state-transition diagram like **figure 10**. The average arrival rate is assumed to be λ and the average service rate is μ, then the steady state formulas could look like these. [25 pg 29, 26 pg(s) 56-61]

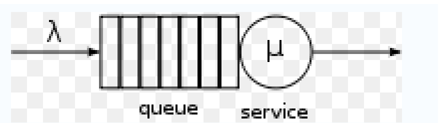


**Fig 11b: M/M/1/K QUEUE SCHEMA [28]**

$$p_n = \left(\frac{\lambda}{\mu}\right)^n p_0 \qquad for \qquad n = 0,1,2,\ldots,K,$$

$$p_0 = {(1-u)}\Big/{1-u^{K+1}}$$

$$\lambda_\alpha = \lambda(1 - p_K)$$

$$\rho = \frac{\lambda_\alpha}{\mu}$$

Here, $\lambda_a$ is the true average arrival rate to the queuing system; ρ is the server utilization; $P_n$ stands for the steady state probability that there are n customers in the queuing system.

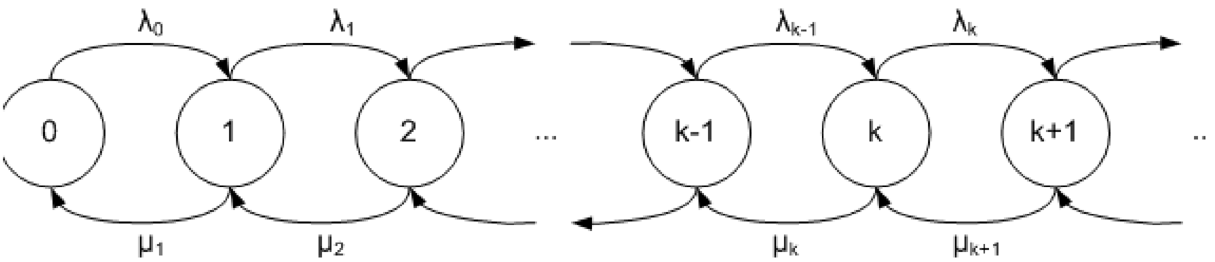


**Fig 12: M/M/1/K State Transition Diagram (Birth-Death Process) [27]**

***M/M/C/K***

Is a multi-server queue with C servers and customer/buffer size K. The arrival process and departure process are quite same as M/M/1/K except for the number of servers. When a new customer enters the system and there are idle servers, the customer enters an idle server and leaves after a random service time. If all servers are occupied, the coming customer should wait in the queue, and even if the queue is also full, the new customer would be retransmitted or blocked

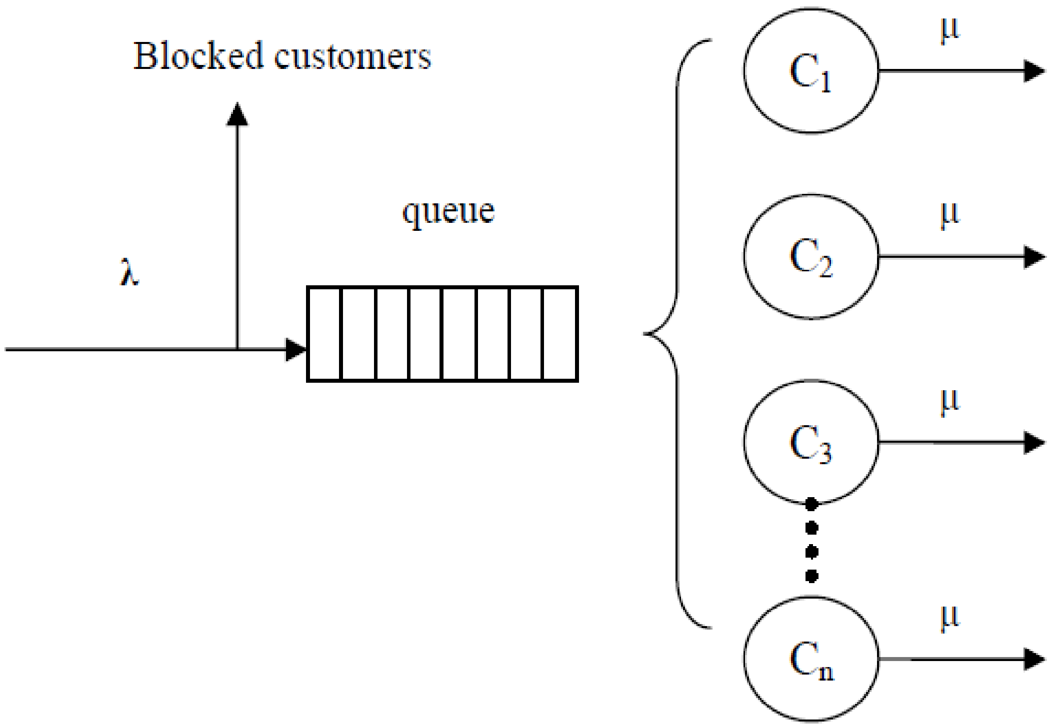


**Fig 13: M/M/C/K Queue**

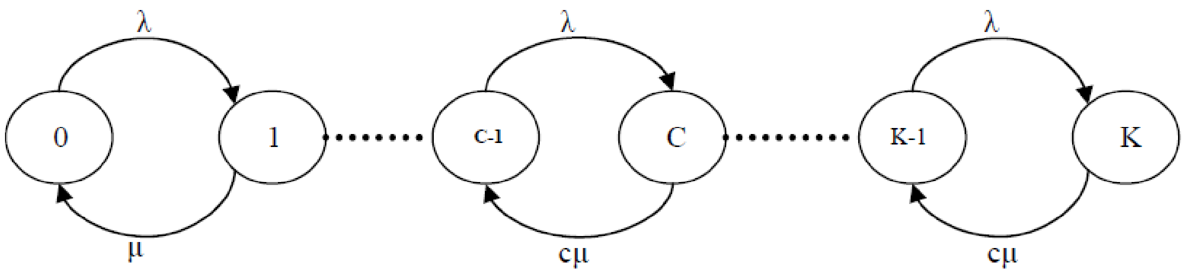


**Fig 14: M/M/C/K State Transition Diagram [26]**

The M/M/C/K system is somewhat different from M/M/1/K system. When the number (n) of customers in the system is less than the number of servers(C) (n<C), the departure rate will become n times of μ. If the number of customers in the system reaches to the total number of servers, the departure rate will be kept as Cμ.

Many formulas from the state-transition diagram were deduced, and were then used to calculate and describe the queuing system's features [26]

$$p_n = \begin{cases} \dfrac{u^n}{n!} p_0 & n = 1,2,\ldots,c \\ \\ \dfrac{u^c}{c!}\left(\dfrac{u}{c}\right)^{n-c} p_0 & n = c+1,\ldots,K, \end{cases}$$

$$\text{Where} \quad p_0 = \left[\sum_{n=0}^{c} \frac{u^n}{n!} + \frac{u^c}{c!}\sum_{n=1}^{K-c}\left(\frac{u}{c}\right)^n\right]^{-1}$$

$$\lambda_\alpha = \lambda(1 - p_K), \qquad \rho = (1 - p_K)u/c$$

$$L = L_q + E[N_s] = L_q + \sum_{n=0}^{c-1} np_n + c\left(1 - \sum_{n=0}^{c-1} p_n\right)$$

$$W = L/\lambda_\alpha$$

$\lambda_a$ is the true average arrival rate to the queuing system; ρ is the server utilization; $P_n$ refers to the steady state probability that there are n customers in the queuing system; L stands for average number in the queuing system when the system is in the steady state; W represents average time in the steady state system.

### 3.4: SIMULATOR PROCESS

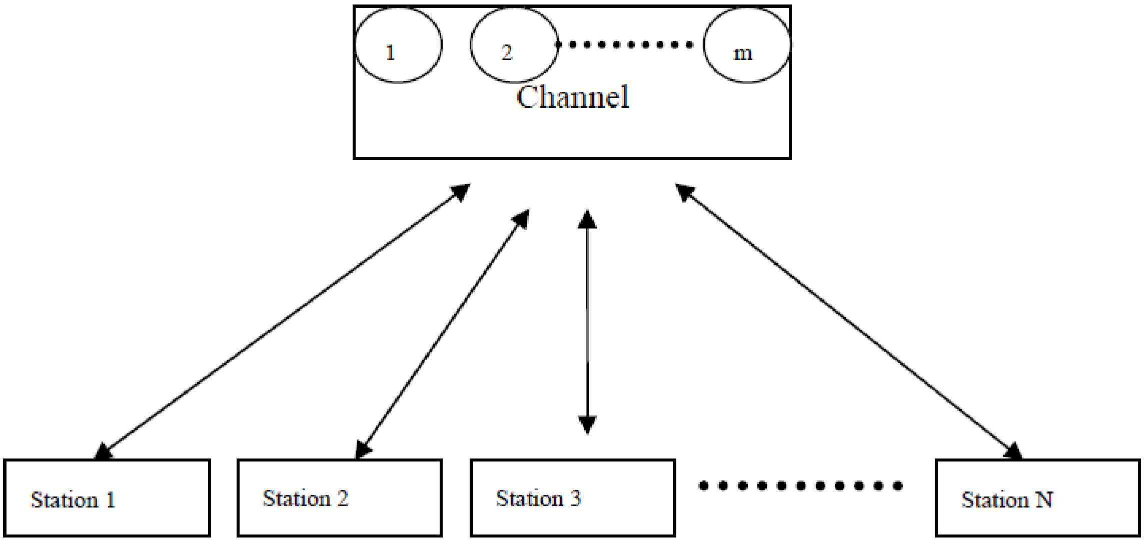


**Fig 15: Simplified framework for wiMAX Simulation.**

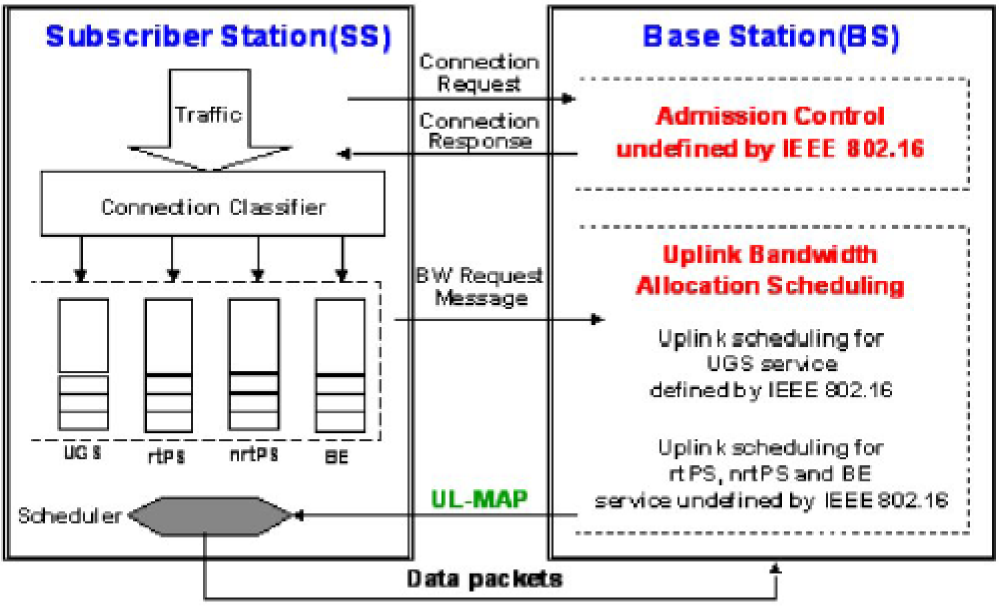


**Fig 16: QoS Architecture Of wiMAX [29]**

In wiMAX systems, the main framework can be assumed as the above **Figure 13** there are n stations, which share the m channels, allocated by AP (Base Station) **Fig 14** shows the QoS architecture to be implemented by **Fig 13**. To simulate the wiMAX system, we can assume that every station is an M/M/1/K queuing system, and the access point (AP) which is a base station in wiMax is an M/M/C/K queuing system with C channels. In each station, the data scheduler will randomly generate the data packets and send the requests. The packet's request process can be seen as a Poisson process, and there is a buffer used to store the coming requests.

If the request is undertaken, the sender will transmit it to the access point, which will distribute the idle channel to the data packets. We also can add a buffer to the access point, which will store the coming requests from different stations. If the buffer is full, the transmission should be requested again in the original station. Therefore, the wireless networks system can be simply modelled as the following figure.

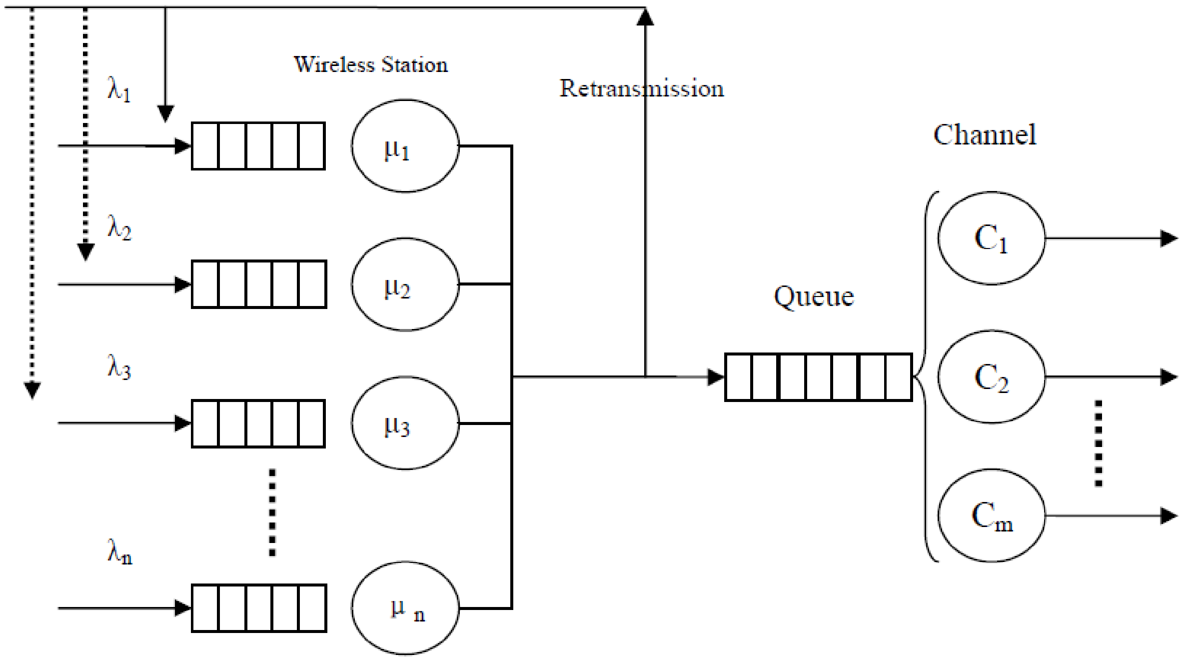


**Fig 17: Simple model of wiMAX[26]**

There are n stations (SS) and an access point (BS) with m channels, so it can be seen as a combination of an amount of M/M/1/K queuing systems at the SS and an M/M/C/K queuing system at the BS. The model above was implemented in Java by writing a program which simulated the M/M/1/K and M/M/C/K queuing system, and then combined them logically. In each queuing system, two main processes should be included, which are arrival process and departure process. Illustrated below are the two processes with logical flow diagram based on the M/M/1/K.[25]

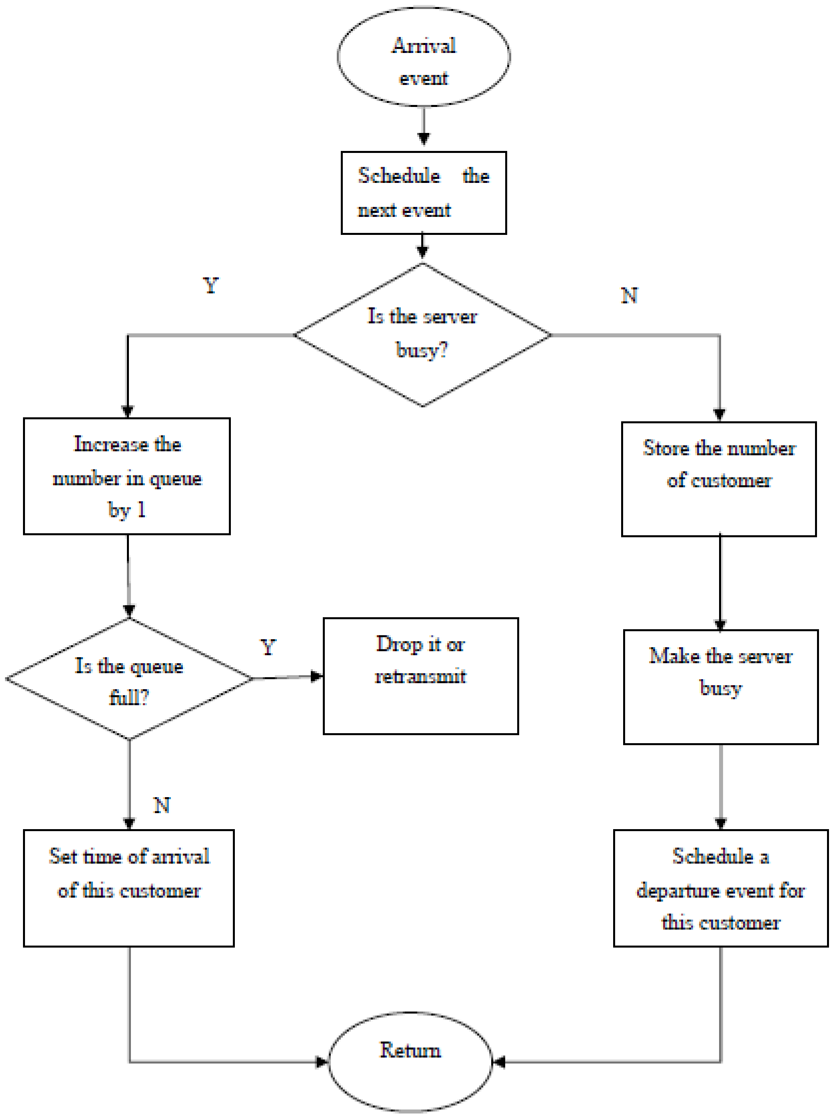


**Fig 18: Flow Diagram of M/M/1/K Arrival Process[26]**

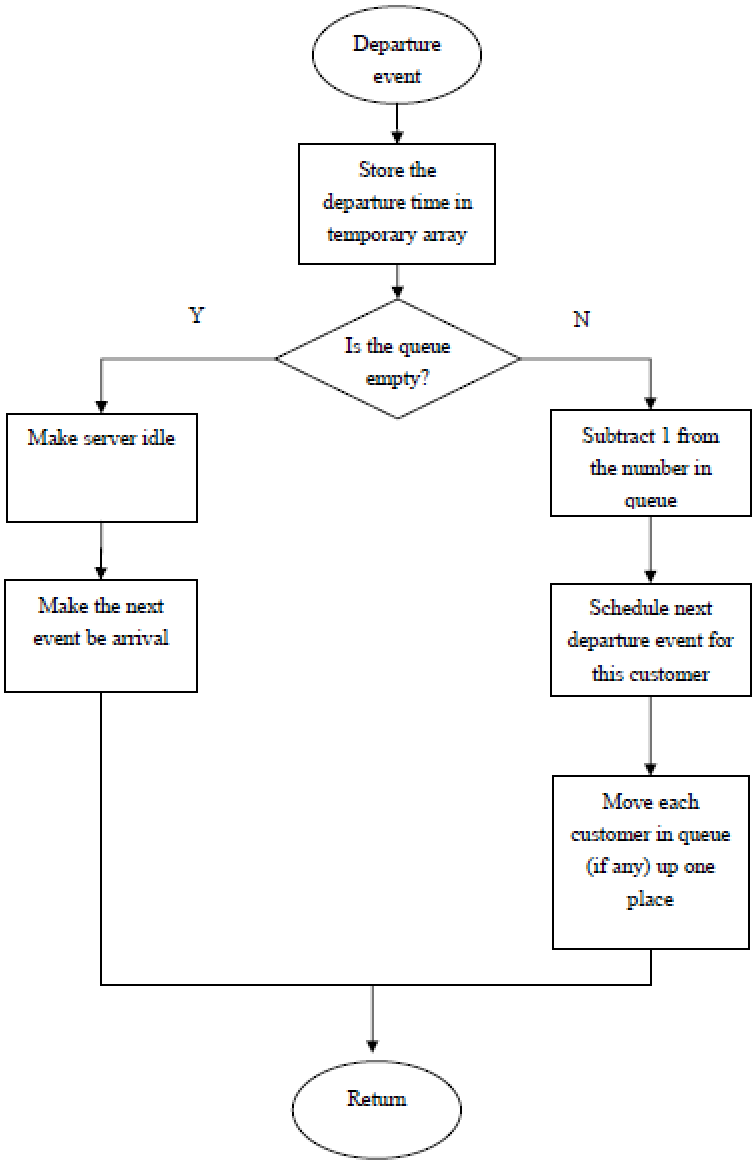


**Fig 19: Flow Diagram of Departure Process.[26]**

## 3.5: QoS METRICS

QoS includes providing Quality of Service to the end user. The perceived quality of service can be quantitatively measured in terms of several factors. In the analysis, the throughput, average delay, average jitter and packet loss were considered

***Throughput rate [6]***

Basically refers to the number of bits per second in a digital network, it's an important metric in the delivery of application services such as high resolution video, with large amounts of data to be delivered continuously.

***Delay (latency)***

Is quite undesirable in application services where real time response is required without a lag, such as voice telephony, conferencing, gaming online, and instructional applications where an individual or a class is guided through a process by an instructor maybe on another continent or straight from a database.

***Packet loss***

Is the failure of the network to deliver all the packets intended for a recipient. Packet loss is sometimes to be expected in a network, but the case of a wireless network is quite on the high side compared to the wired network, could be traceable to sudden fades because of multi-path and variable attenuation of the signal with changing weather conditions. Also buffers from time to time drop packets when completely full.

***Jitter***

Refers to a lag in or time difference in the the arrival time of a certain number of packets from the point of transmission over time or put simply the variation in average delay of successive packets, this is particularly not desirable for real time applications like the normal voice telephony. Jitter can be traded for delay by storing packets in a buffer and then sending them with equal time interval between them

### 3.6: SIMULATION PARAMETERS

| Class Description | Real Time? | Application Type | Bandwidth |
|---|---|---|---|
| Interactive Gaming | Yes | Interactive Gaming | 50 - 85 kbps |
| VoIP, Video Conference | Yes | VoIP | 4 - 64 kbps |
| | | Video Phone | 32 - 384 kbps |
| Streaming Media | Yes | Music/Speech | 5 - 128 kbps |
| | | Video Clips | 20 - 384 kbps |
| | | Movies Streaming | > 2 Mbps |
| Information Technology | No | Instant Messaging | < 250 byte messages |
| | | Web Browsing | > 500 kbps |
| | | Email (with attachments) | > 500 kbps |
| Media Content Download (Store and Forward) | No | Bulk Data, Movie Download | > 1 Mbps |
| | | Peer-to-Peer | > 500 kbps |

**FIG: 20 wiMAX Service classes [30]**

| Simulator Requirement | VoIP (codec G.711) | Video (H.262) | Multimedia (Web) |
|---|---|---|---|
| No of stations | 1,2-8 | 1,2-8 | 1,2-8 |
| Mean Inter-Arrival | 0.02 | 0.02 | 0.02 |
| Station mean service time | 0.05 | 0.05 | 0.05 |
| Channel mean time | 0.033 | 0.033 | 0.033 |
| No of packets | 50 | 50 | 50 |
| No of Channels | 2 | 2 | 2 |
| Packet size( bit) | 1280 | 1440 | 5900 |
| Queue size | 4 | 4 | 4 |

**Table 3: Simulation Data [29, 30, 31, 32, 33, 34]**

```java
public static void initialize() {
String input1, input2,input7, input8, input9, input10, input11,input12;
input1 = JOptionPane.showInputDialog("Input the number of stations");
input2 = JOptionPane.showInputDialog("Input the value of stations mean interarriva
input7 = JOptionPane.showInputDialog("Input the value of sta. mean service time");
input8 = JOptionPane.showInputDialog("Input the value of channel's mean time");
input9 = JOptionPane.showInputDialog("Input the number of packets required");
input10 = JOptionPane.showInputDialog("Input the number of channels");
input11 = JOptionPane.showInputDialog("Input the packet size/Bit");
input12 = JOptionPane.showInputDialog("Input the queue size");
num_sta = Integer. parseInt(input1);
mean_arrival = Double. parseDouble(input2);
mean_service = Double.parseDouble(input7);
mean_service_ap = Double.parseDouble(input8);
num_customer_required = Integer.parseInt(input9);
num_channel = Integer. parseInt(input10);
packet_size = Integer. parseInt(input11);
q_size = Integer. parseInt(input12);
time_next_event[0][0] = 0;
blocked_packet = 0;
total_packet = 0;
num_packet_ap = 0;
num_successful_transmission = 0;
total_delay_ap = 0;
```

A portion of the initialize() method of the simulator code that deals with taking the inputs listed in table 3 above from the user and passes it to other methods such as arrive(),arrive_ap(),depart(),depart_ap(),time(),timing_ap() etc in the simulator code for processing at the same time initialising required parameters.

```java
public static void arrive() {
int sta = station_arrival;
time_next_event[sta][1] = sim_time[sta] + expon(mean_interarrival[sta]);
if (server_status[sta] == 0) {
server_status[sta] = 1;
time_next_event[sta][2] = sim_time[sta] + expon(mean_service);
}
else {
num_in_q[sta]++;
time_arrival[sta][num_in_q[sta]] = sim_time[sta];
} //end if...else
}
```
The arrive() method depicting the M/M/1/K at the subscriber station.

```java
public static void report() {
double throughput, load;
average_delay = total_delay_ap/num_successful_transmission;
blocking_probability = blocked_packet/total_packet;
load = total_packet;
throughput = (num_successful_transmission * packet_size ) / sim_time_ap;
//Jitter= total_delay_ap-average_delay;
JOptionPane.showMessageDialog(null,
"The value of load is : " + load +
"\nThe value of blocking probability : " + blocking_probability +
"\nThe value of blocked packets : " + blocked_packet +
"\nThe value of successful transmission : " + num_successful_transmission +
        "\nThe value of average delay : " + average_delay +
"\nThe value of Throughput : " +throughput
   //"\nThe value of Jitter : " +Jitter
   , "Result",JOptionPane.INFORMATION_MESSAGE);
} // end method of report
```

The report method takes the inputs from other methods such as the arrive()/arrive_ap(),depart()/depart-ap() from the simulator code and does the automated calculations , and presents it in a result pane showing the following

1.average_delay

2.blocking_probability

3.load

4.throughput(Mb)

1. average_delay = total_delay_ap ÷ num_successful_transmission

   Total_delay_ap= sim_time_ap - time_arrival_ap

   sim_time_ap= time_arrival_ap[num_in_q_ap]

2. blocking_probability = blocked_packet ÷ total_packet

   total_packet = num_of_packet

3. load = total_packet

4. throughput = (num_successful_transmission * packet_size ) ÷sim_time_ap

   sim_time_ap= time_arrival_ap[num_in_q_ap]

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# CHAPTER FOUR

## SIMULATOR OUT-COME

Quality of service in wiMAX networks using real life application packet (applications level simulation) data were considered. wiMAX provides basic IP connectivity. Mobile devices capable of using wiMAX network will need to support voice calling over the internet protocol.

Thus, VoIP is the first application considered. With the availability of a larger data pipe, another common application these days is viewing videos over the internet. So video streaming is analyzed as well. In this chapter, the simulation results are presented. The QoS parameters obtained for VOIP traffic is presented first. The multimedia traffic results follow also.

### 4.1: RESULTS WITH VARIED NUMBER OF STATIONS

The numbers of stations were varied while other factors were kept constant and simulations were carried out using the appropriate bandwidth for VoIP, Video,and multimedia using table 3 below reproduced

| Simulator Requirement | VoIP (codec G.711) | Video (H.262/4) | Multimedia (Web/Streaming) |
|---|---|---|---|
| No of stations | 1,2-8 | 1,2-8 | 1,2-8 |
| Mean Inter-Arrival | 0.02 | 0.02 | 0.02 |
| Station mean service time | 0.05 | 0.05 | 0.05 |
| Channel mean time | 0.033 | 0.033 | 0.033 |
| No of packets | 50 | 50 | 50 |
| No of Channels | 2 | 2 | 2 |
| Packet size( bit) | 1280 | 1440 | 5900 |
| Queue size | 4 | 4 | 4 |

**Simulation (Sample Results) window with 1no station for VoIP traffic**

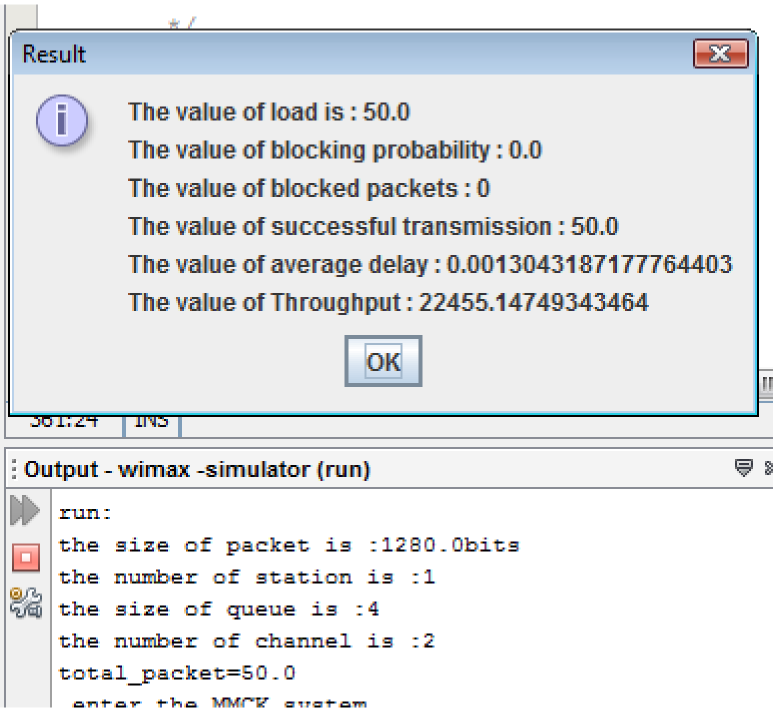


**Fig.21: A sample simulation output.**

**Results for VoIP with Number of Stations Varied**

| No of stations | Throughput (Mb) | Blocking probability | Average Delay |
|---|---|---|---|
| 1 | 0.02245 | 0.0 | 1.3E-3 |
| 2 | 0.04450 | 0.0 | 4.03E-3 |
| 4 | 0.05952 | 0.12 | 2.39E-2 |
| 6 | 0.07986 | 2.94E-1 | 2.66E-2 |
| 8 | 0.08759 | 2.94E-1 | 2.660E-2 |

**Table 4 Sample result table for VoIP Simulation**

Other results are shown as graphs with attempts at explanations made and conclusions inferred.

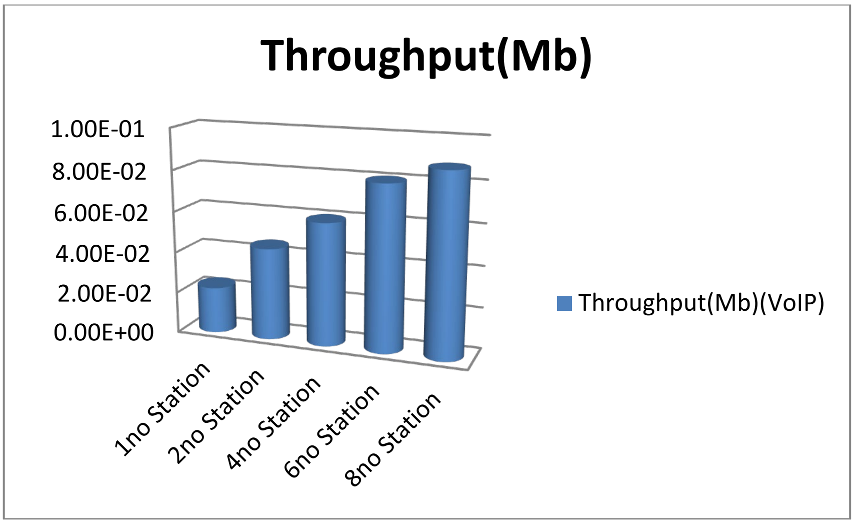


**Fig 22: VoIP- No of Stations –V-Throughput**

From the above diagram its quite clears at first glance that, the throughput of VoIP application increases steadily with the increase in number of stations up to eight number stations in this instance.

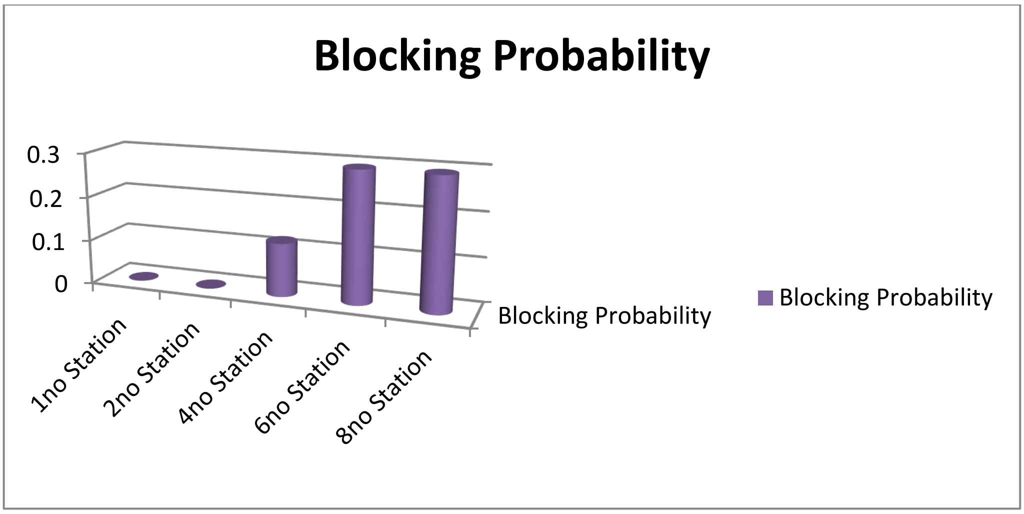


**Fig 23: VoIP -Blocking probability V No of stations**

In this figure the blocking probability for VoIP is effectively zero, when the no of stations is less than 2nos, at 4 nos there is a sharp increase in the blocking probability and a further sharp increase at 6nos station, It was observed that the blocking probability becomes stable after six stations have been introduced into the network

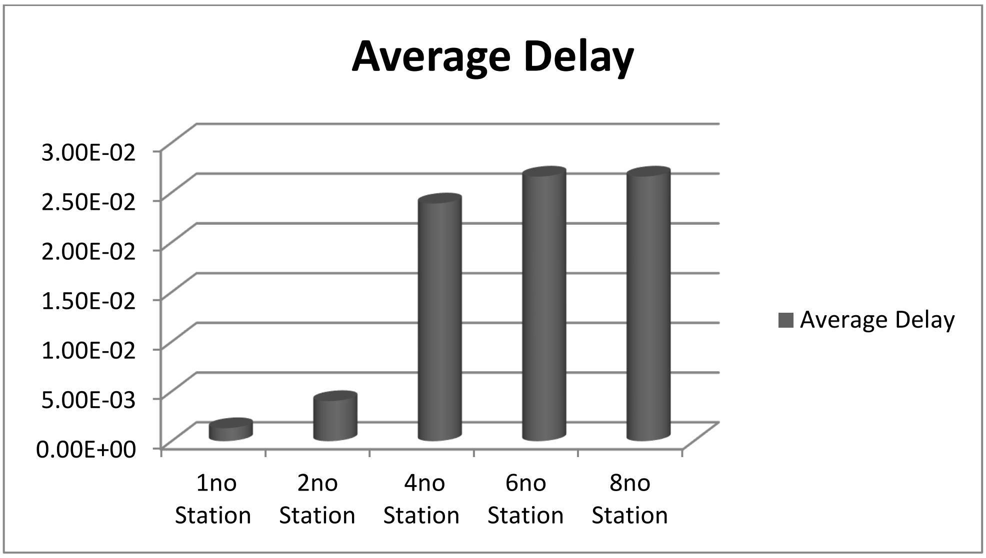


**Fig 24: VoIP- No of Stations –V-Throughput**

One can see from the chart that the delay of VoIP increases steadily with the increase in number of stations at first steadily and then sharply with a slight increase, thereafter the value of delay for VoIP application becomes stable.

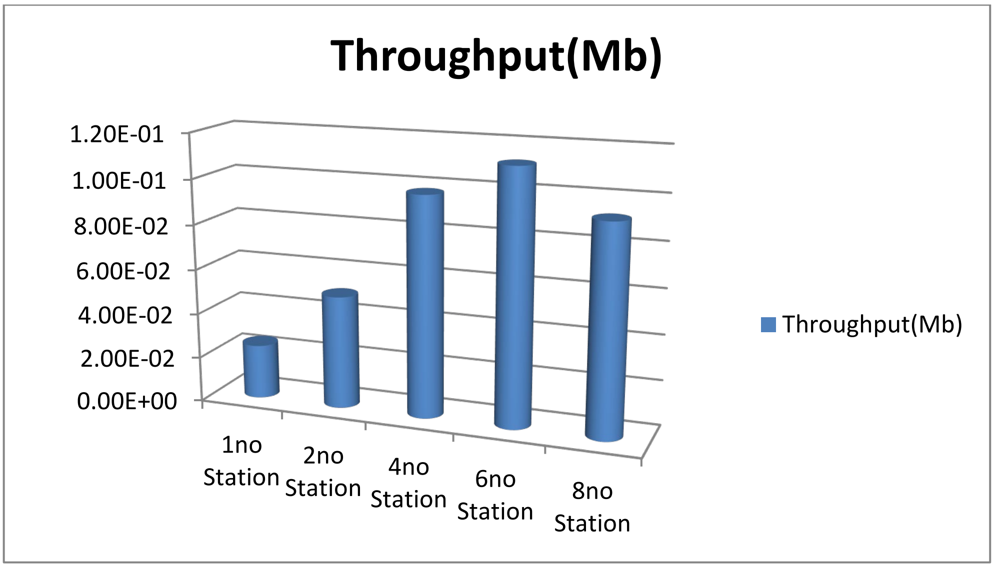


**Fig 25: Video-No Of Stations -V- Throughput**

The throughput for video application increases as the number of stations increase, till it reaches a maximum and after which the throughput values falls.

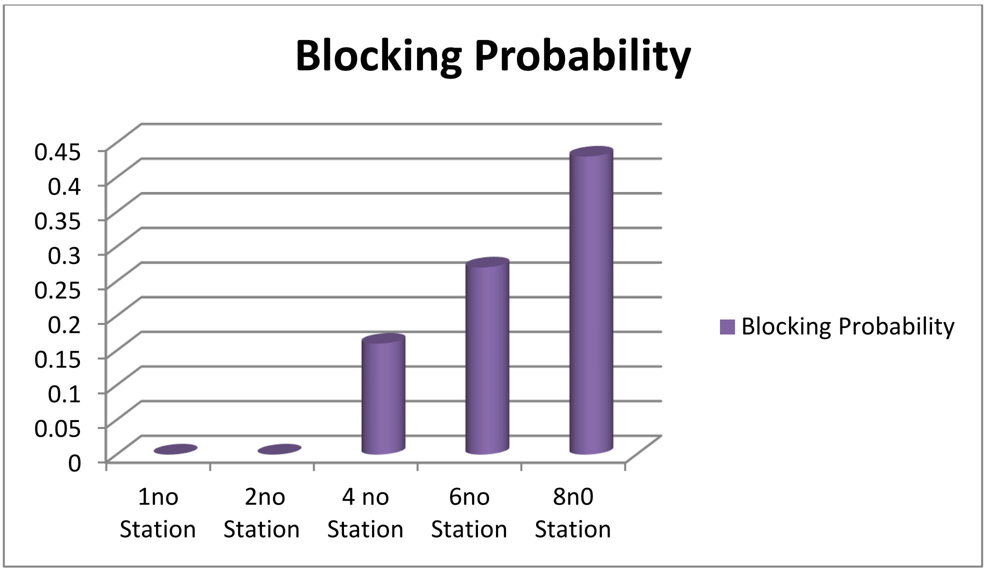


**Fig 26: Video-No Of Stations -V - Blocking Probability**

The blocking probability for video application steadily and strongly increases with an increase in the number of stations.

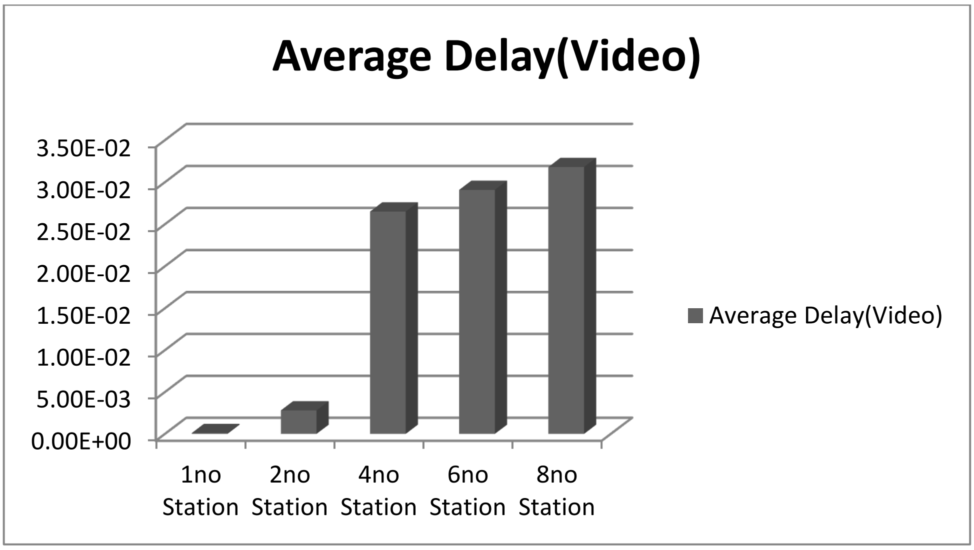


**Fig 27: Video –No of Stations-V-Average Delay**

The Video simulation shows no evidence of delay initially, but as the number of station doubles, average delay greatly increases and continues to do so steadily thereafter.

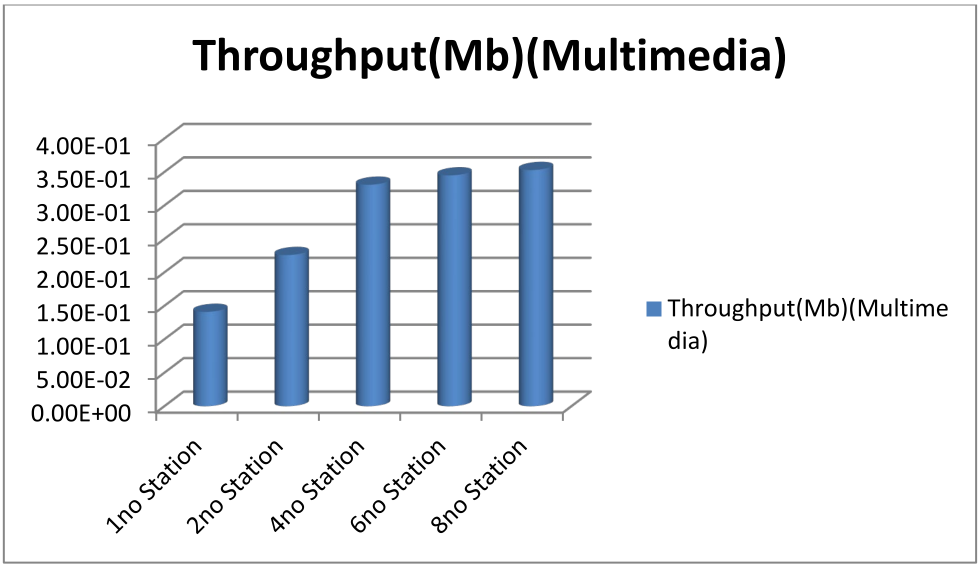


**Fig 28: Multimedia-No Of Stations –V-Throughput**

For multimedia applications simulation the throughput steadily increases and basically becomes fairly steady with increase in no of stations.

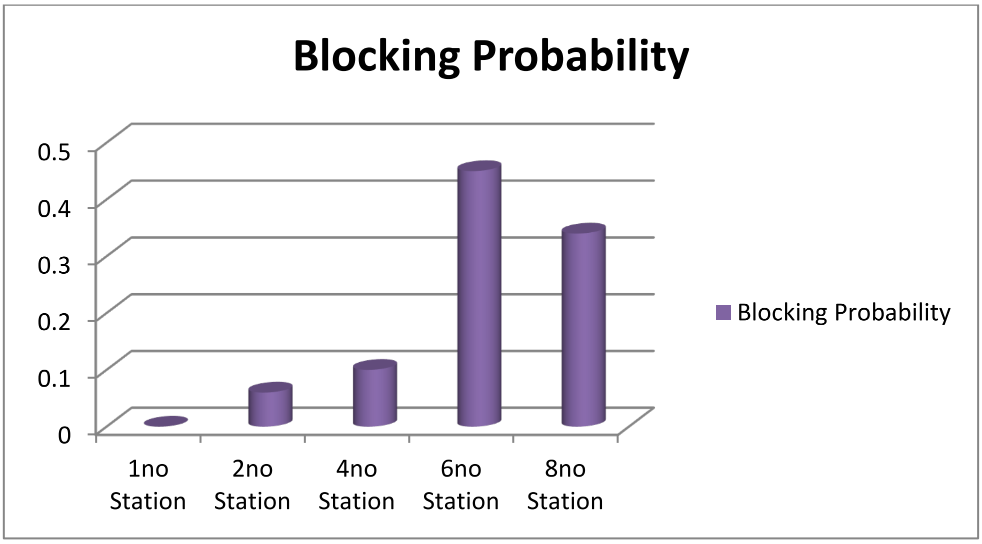


**Fig 29: Multimedia-No Of Stations – V- Blocking Probability**

Blocking probability for multimedia applications gradually increases and then drops sharply with a further increase in the number of stations the blocking probability decreases rather sharply.

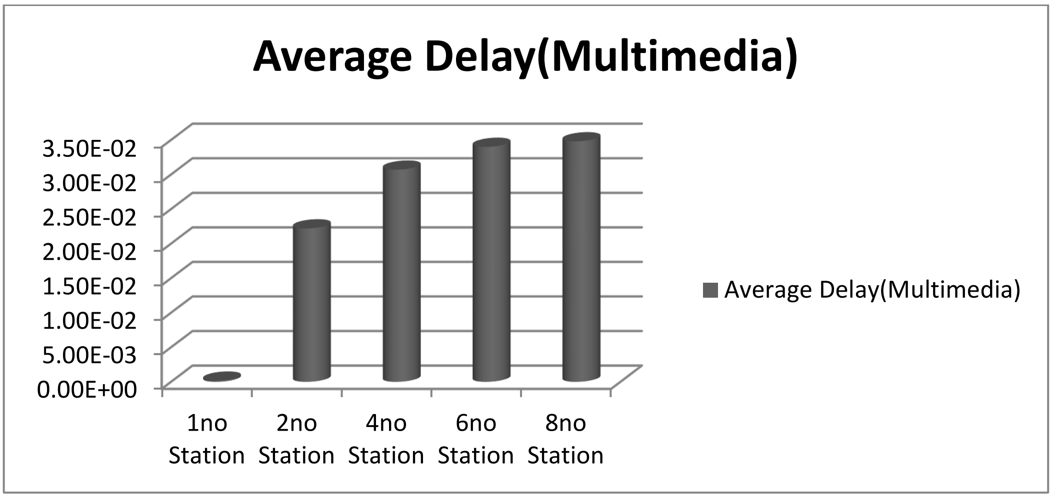


**Fig 30**: **Multimedia-No of Stations-V-Average Delay**

Average Delay for multimedia traffic shows a steady increase with the increase in the number stations.

### 4.2: RESULTS WITH VARIED NUMBER OF CHANNELS

For this set of simulations the channels were varied, the channels represent the number of servers in an actual network, and the effect on the QoS parameters for VoIP, Video, and multimedia were recorded, these parameters are throughput, blocking probability and average delay, they are presented as is here in this section. See sample simulation result window

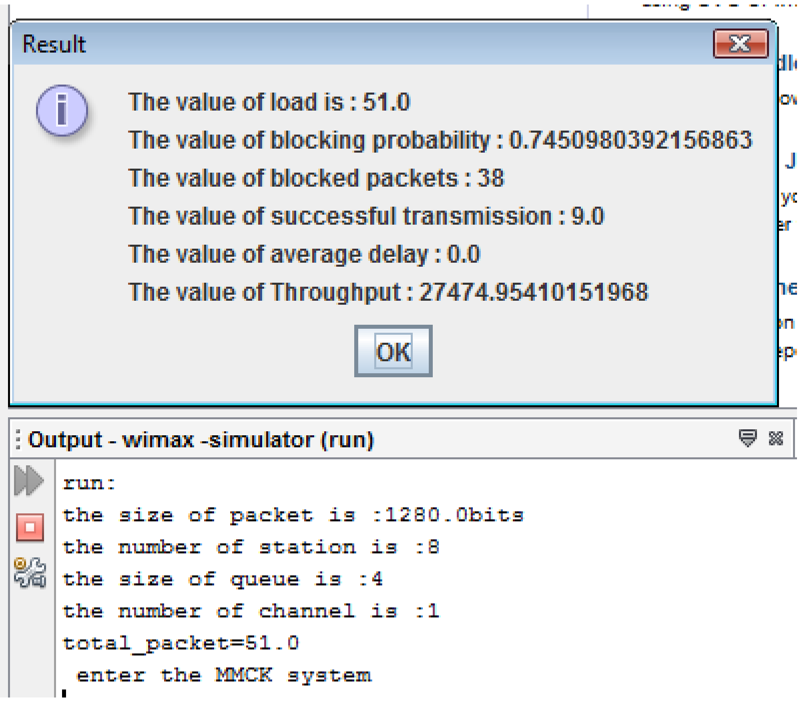


**Fig 31: A simulation result page with varied channel**

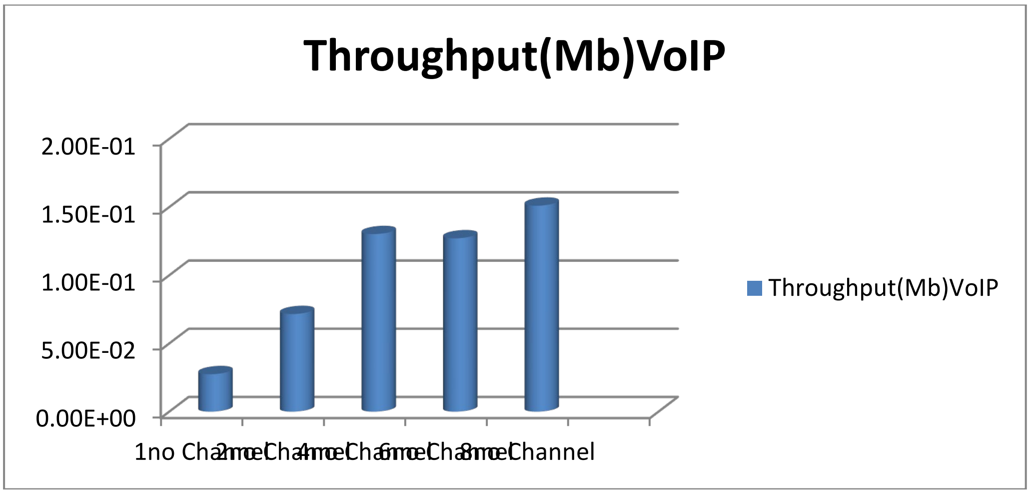


**Fig 32: VoIP –Varied Channel No –V-Throughput**

From the above diagram it can be seen that the throughput for VoIP strongly and steadily improves as the numbers of channels increase, it becomes steady and drops even slightly, before it continues on the upward climb again with increase in no channels.

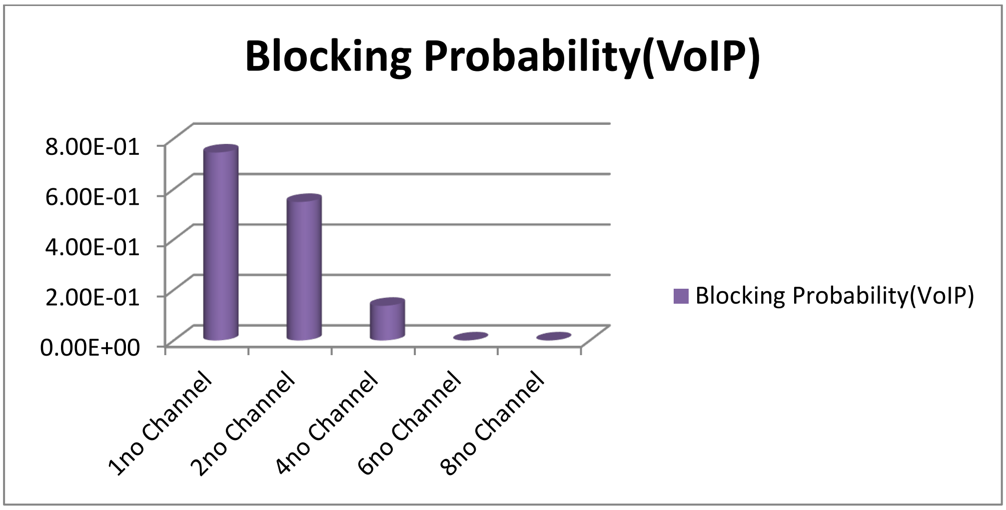


**Fig 33: VoIP-Varied Channel No-V-Blocking Probability**

As the number of channels increases, the value of the blocking probability gradually decreases until by the addition of six and above channels for this simulation the blocking probability becomes zero.

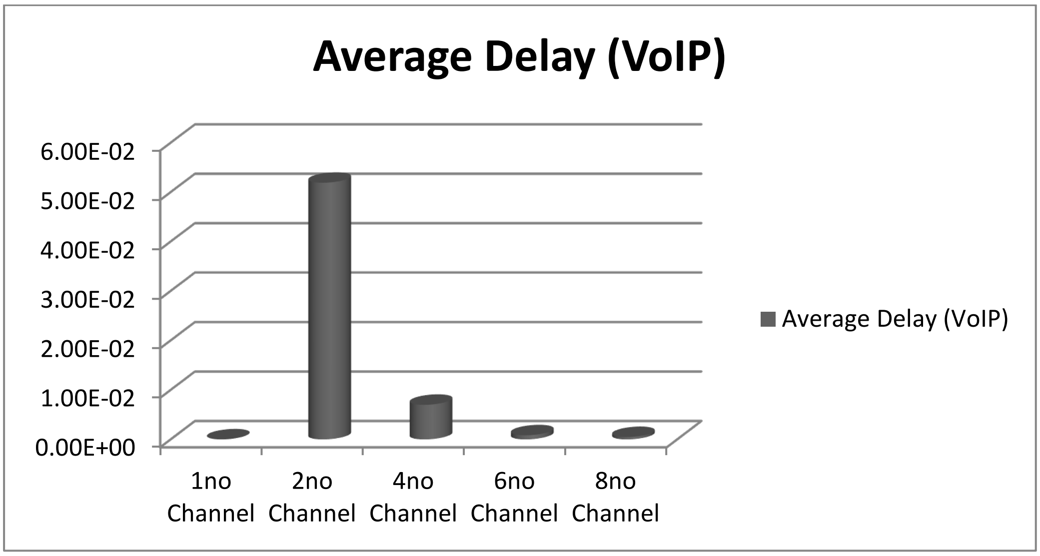


**Fig 34: VoIP-Channels Varied –V-Average Delay**

With one number channel the delay is non-existent or can be be said to be infinitesimal, but a significant increase is observed at two number channels, with a significant drop in the value of average delay with four number channels, and the value of average delay continues to drop steadily with a steady increase in the number of channels.

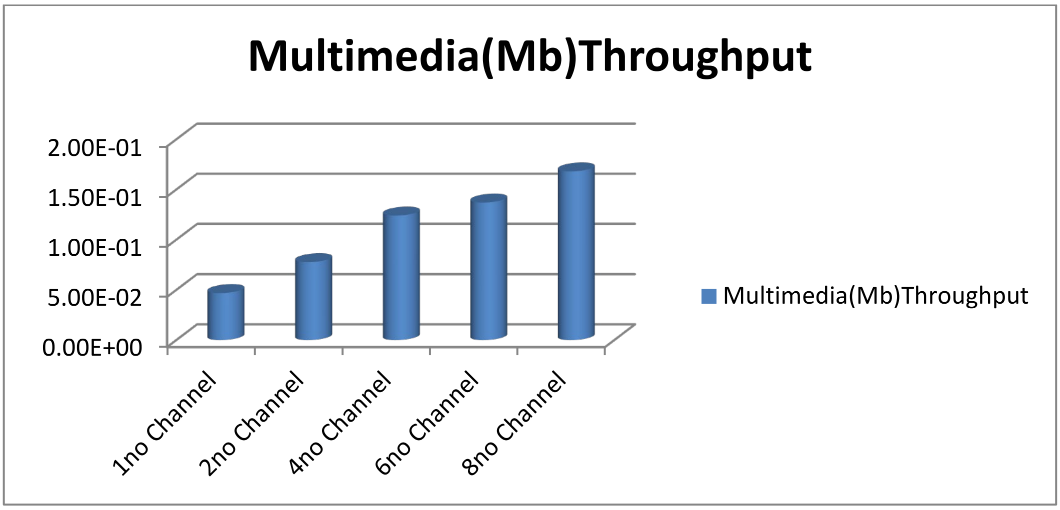


**Fig 35: Video-Channels Varied -V-Throughput (Video)**

For the video simulations it was observed that an increase in the number of channels resulted in a corresponding steady increase in the value of throughput.

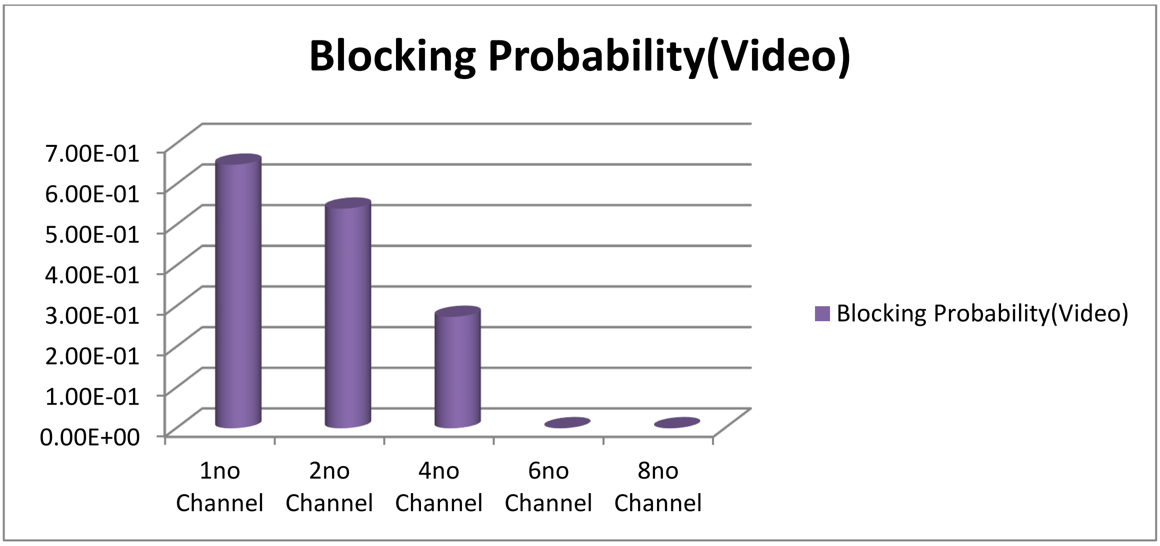


**Fig 36: Video-Channels Varied -V-Blocking Probability (Video)**

Here the blocking probability for simulated video traffic steadily decreases as the number of channels increases until at the six number of channels the blocking probability becomes zero, and stays that way till the end of this set of simulations for video traffic.

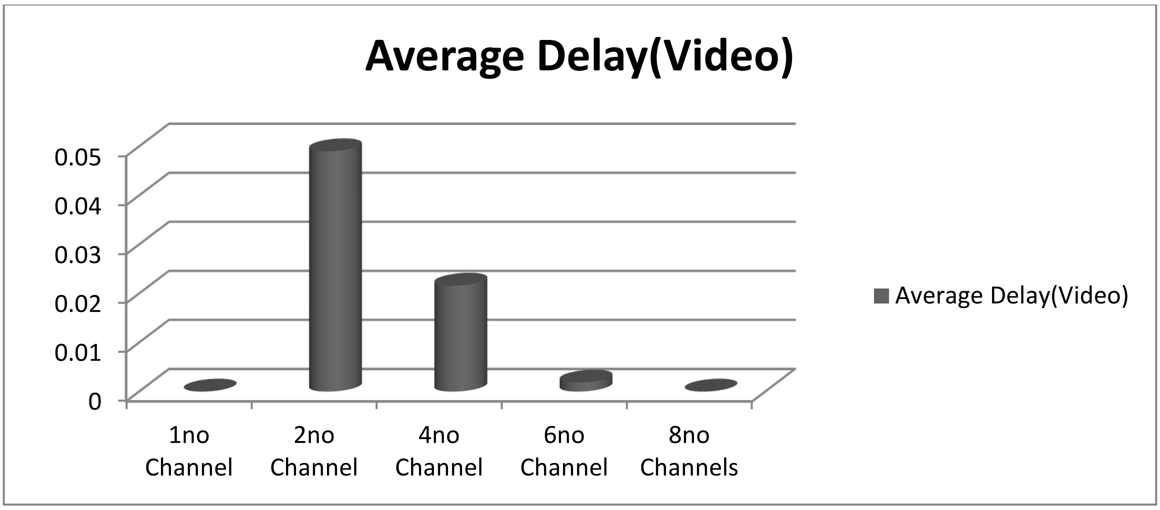


**Fig 37: Channels Varied -V-Average Delay (Video)**

The average delay increases sharply and drops rather quickly as the number of channels is increased until the average delay becomes zero.

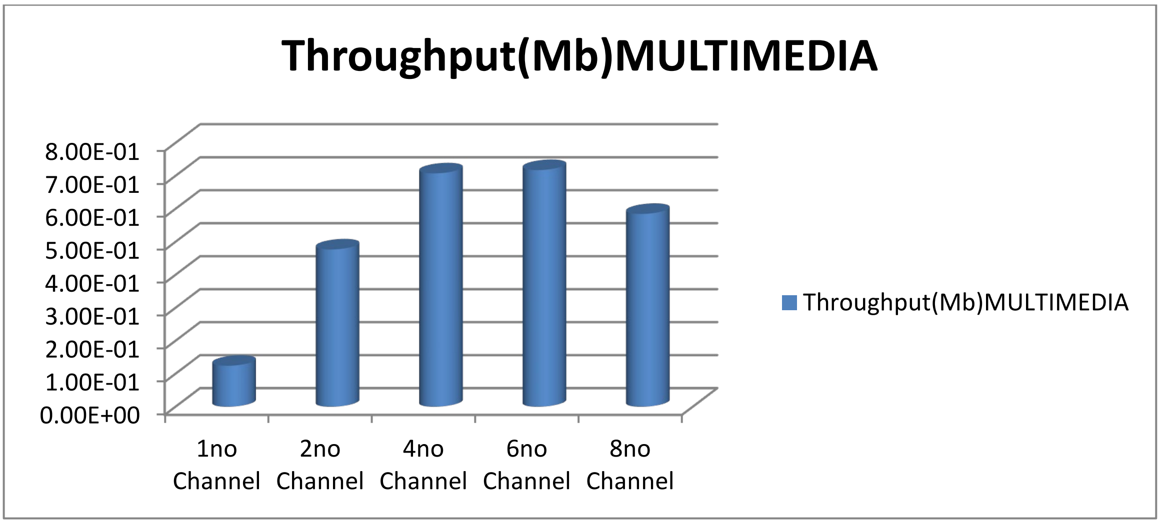


**Fig 38: Channels Varied -V-Throughput (Multimedia)**

The multimedia traffic simulations show from the chart that there is a steady increase in the value of throughput up until when the number of channels equals to four, the throughput seems to steady out at about the same value even with the increase to six channels, then a significant drop in throughput as the channel is further increased is observed.

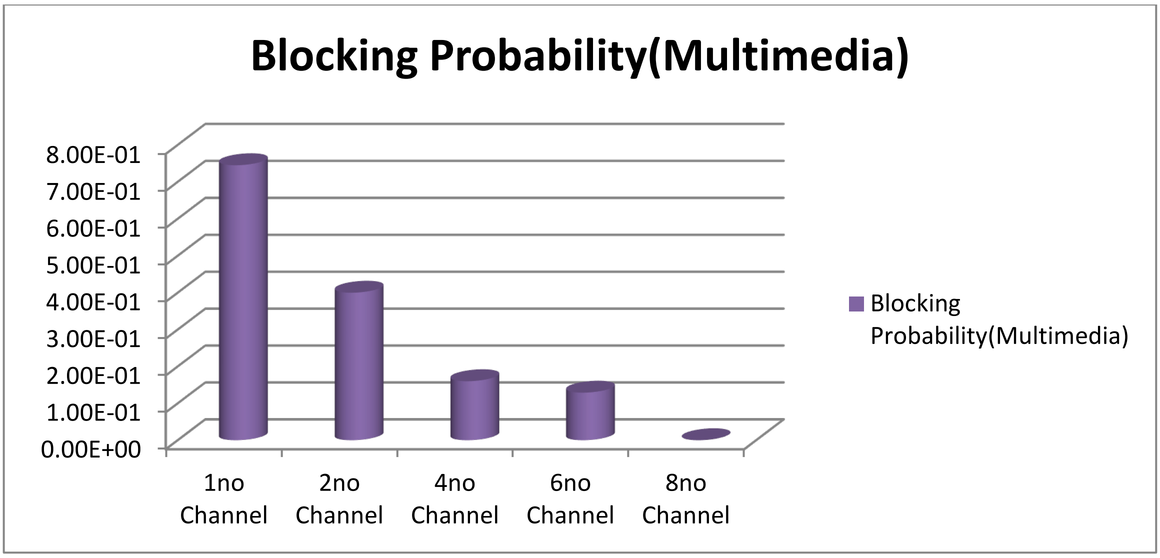


**Fig 39: Channels Varied -V-Blocking Probability (Multimedia)**

Here there is a steady decrease in the value of the blocking probability, for the multimedia traffic as the numbers of channels were steadily increased, it seems to show a slight decrease between four number channels and six but then drops to zero with an increase to eight.

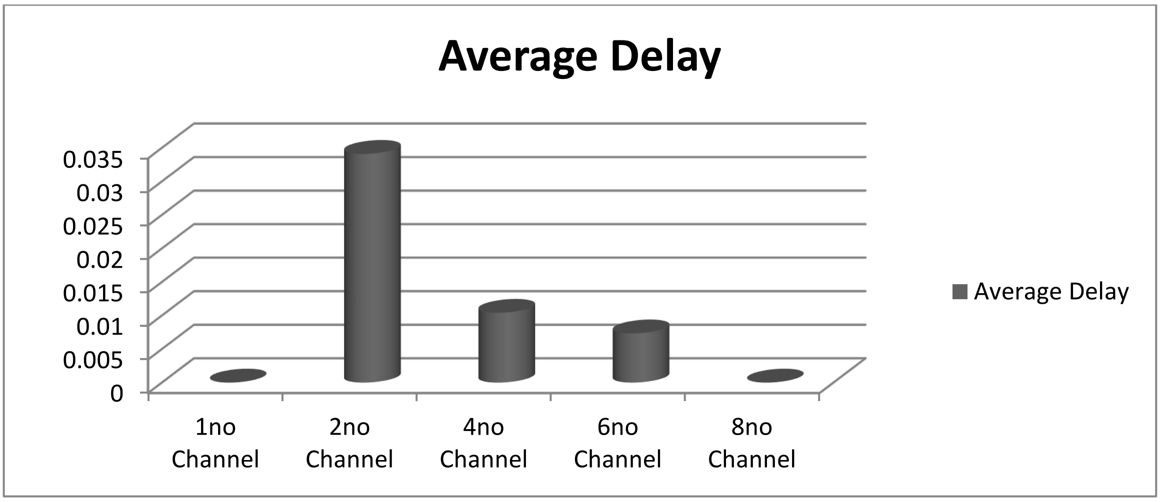


**Fig 40: Channels Varied -V-Average Delay (Multimedia)**

The value for average delay is zero with just one channel, but when the number of channels is doubled, the delay suddenly peaks, with a further doubling in the number of channels the average delay value drops more than half, but the sharp drop slows when the number of channels are increased to six. At eight number channels the average delay disappears entirely.

## 4.3: RESULTS WITH VARIED NUMBER OF QUEUES

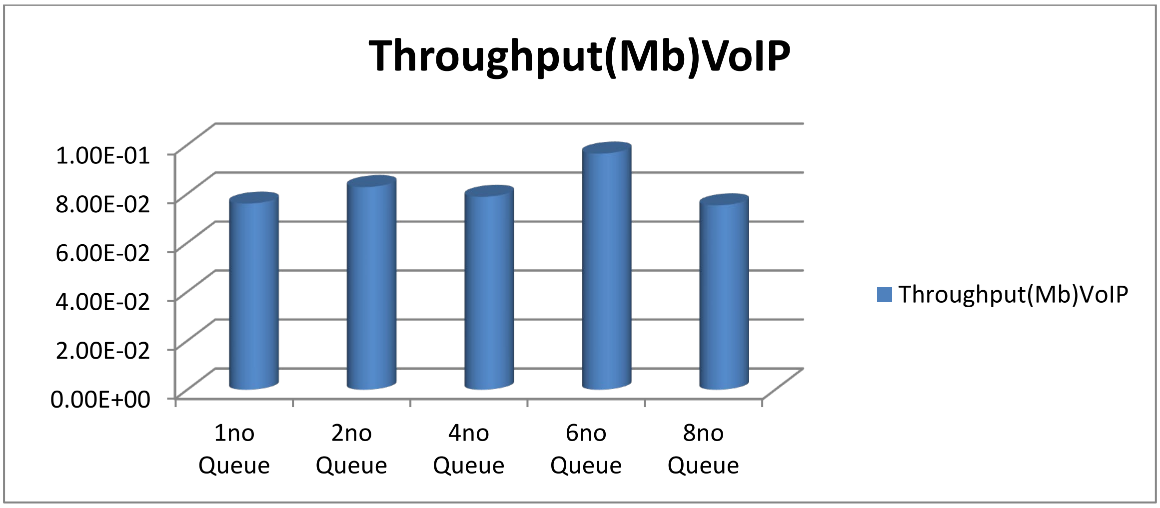


**Fig 41: No of Queues Varied-V-Throughput (VoIP)**

With number of queues varied for VoIP traffic, and other values kept constant, the throughput was found to hover basically within a range (6.00E-02 to 8.00E-02) showing only a significant increase when the queue number is six, at eight number queue sizes it goes back within the above specified range.

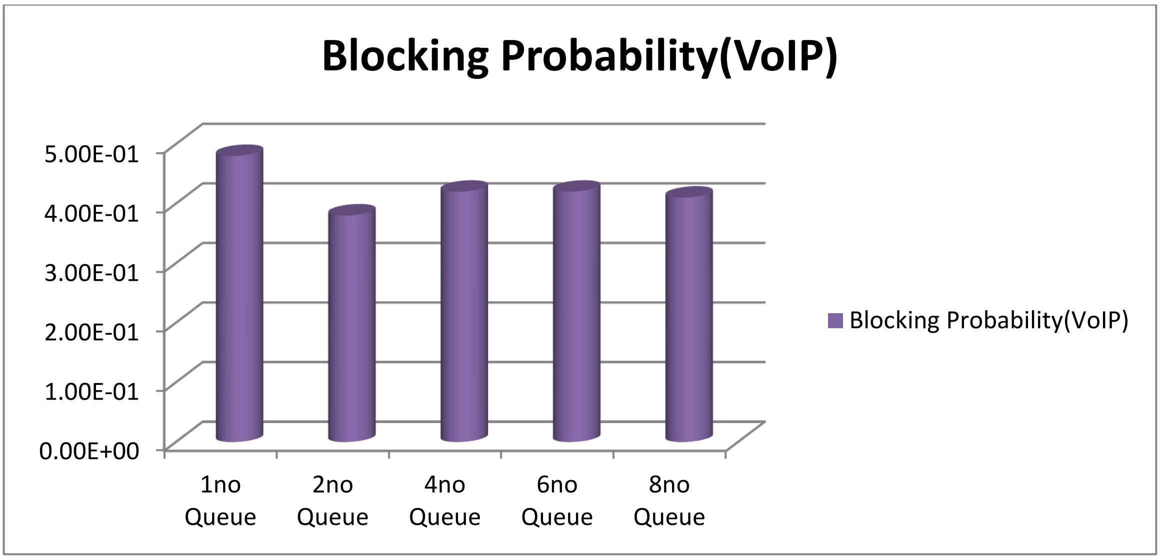


**Fig 42: No of Queues Varied-V-Blocking Probability (VoIP)**

For VoIP traffic the blocking probability peaks when the queue size is one, when this is doubled the blocking probability shows a decrease, further doubling of the queue size to four shows an increase in the blocking probability, but at six and queue sizes respectively the blocking probability does not change significantly.

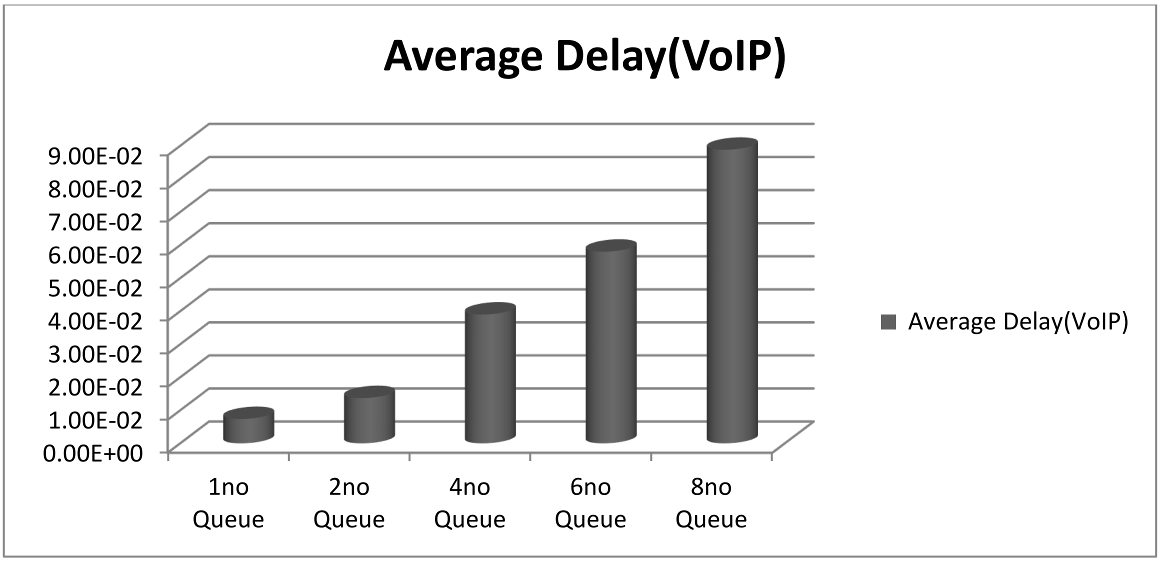


**Fig 43: No of Queues Varied-V-Average Delay (VoIP)**

The average delay for VoIP steadily increases as the size of the queue is gradually increased from one number queue size through to eight number queue size.

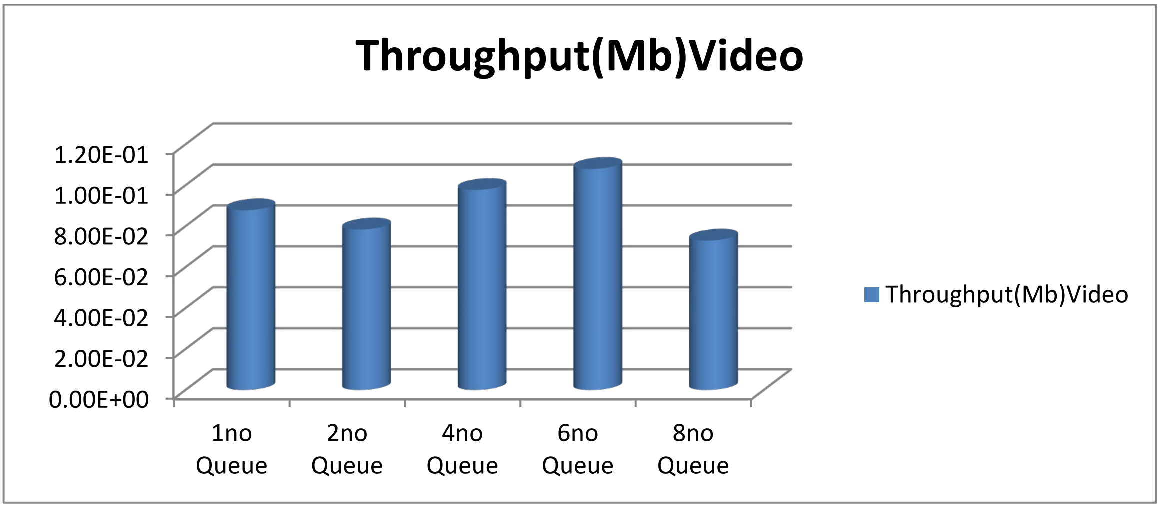


**Fig 44: No of Queues Varied-V-Throughput (Video)**

The throughput for the video traffic simulation when the queue size is varied, shows an initial decrease when the queue size is doubled, further doubling of queue size to four shows a slight increase and a further increase to six number queue sizes show a further increase in throughput, but an increase to eight number queue size shows a dramatic drop in queue size below the value of throughput even when the queue size was equivalent to just one.

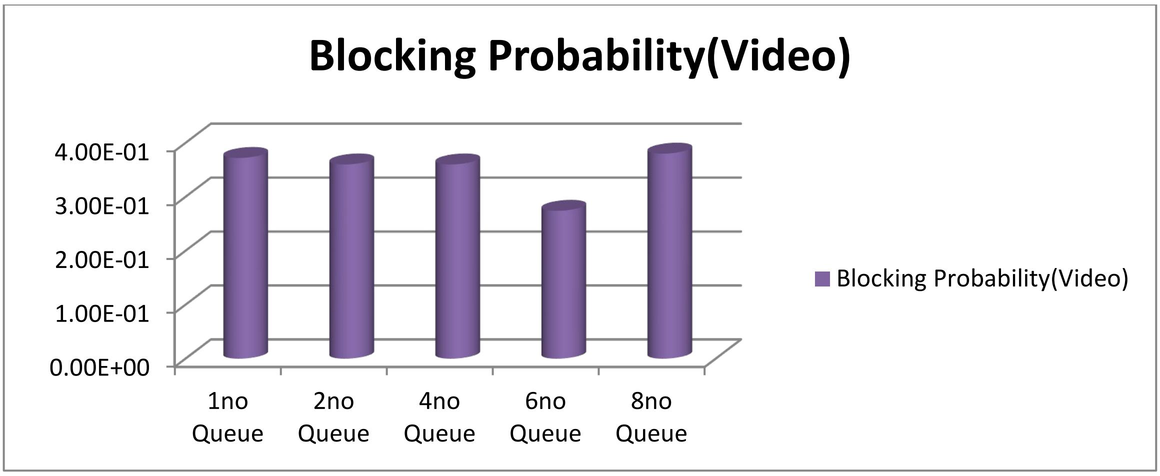


**Fig 45: No of Queues Varied-V-Blocking Probability (Video)**

The video traffic blocking probability stays fairly even as the queue size is varied upwardly up to four number queue size. There-after a slight decrease in the value of the blocking probability is noticed at the six number queue size point of the simulation, after which the value of the blocking probability climbs again as the queue size is increased to 8 number queue size.

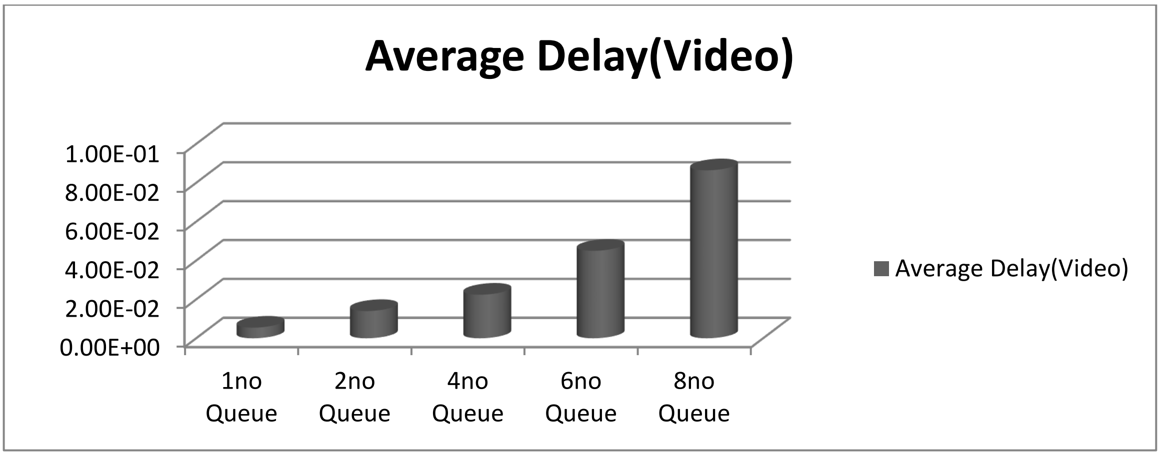


**Fig 46: No of Queues Varied-V-Average Delay (Video)**

The average delay for the video traffic as the queue size is steadily increased is observed to steadily increase to a peak value as the queue size reaches the eight number value.

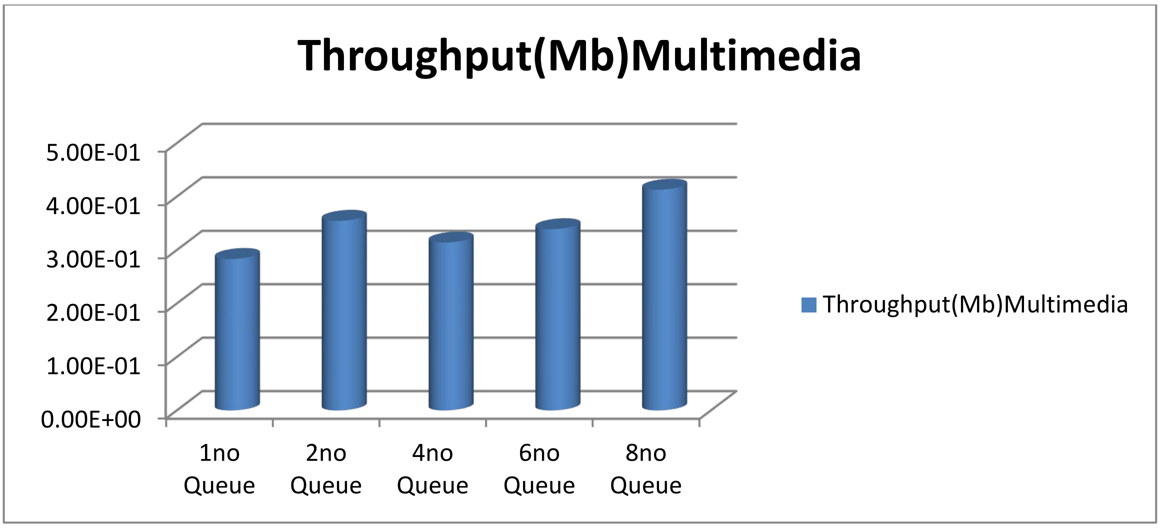


**Fig 47: No of Queues Varied-V-Throughput (Multimedia)**

It was noted that the throughput of the multimedia application traffic simulation experienced an increase in throughput as the queue size is doubled from one number through to two numbers, when further doubled to four numbers a decrease in throughput was observed, a further increase in the queue sizes to six and eight respectively results in a steady increase in the value of throughput also.

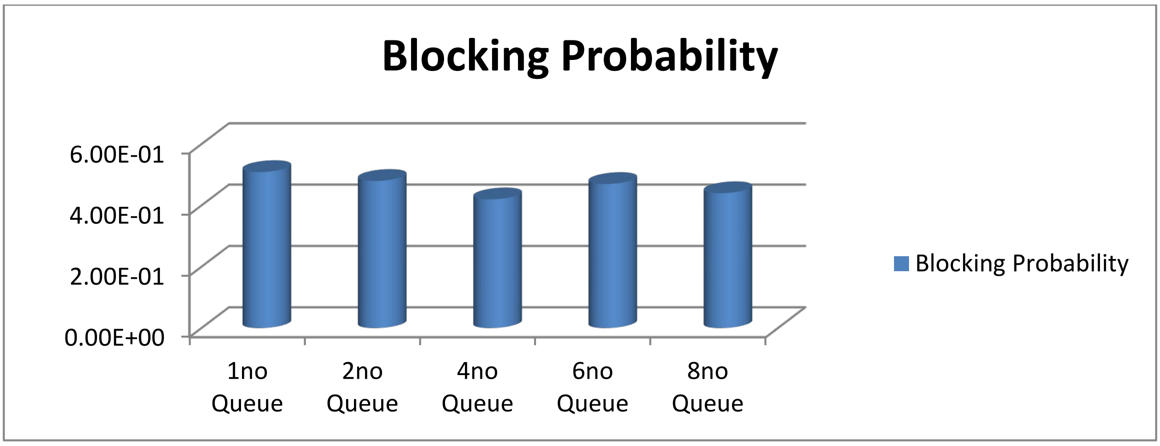


**Fig 48: No of Queues Varied-V-Blocking Probability (Multimedia)**

The blocking probability for the simulated multimedia traffic steadily decreases as the size of the queue decreases, up to 4 no queues, after which a slight increase is noticed when the size of The queue equals six, but on increase to eight number queues size the blocking probability drops once again.

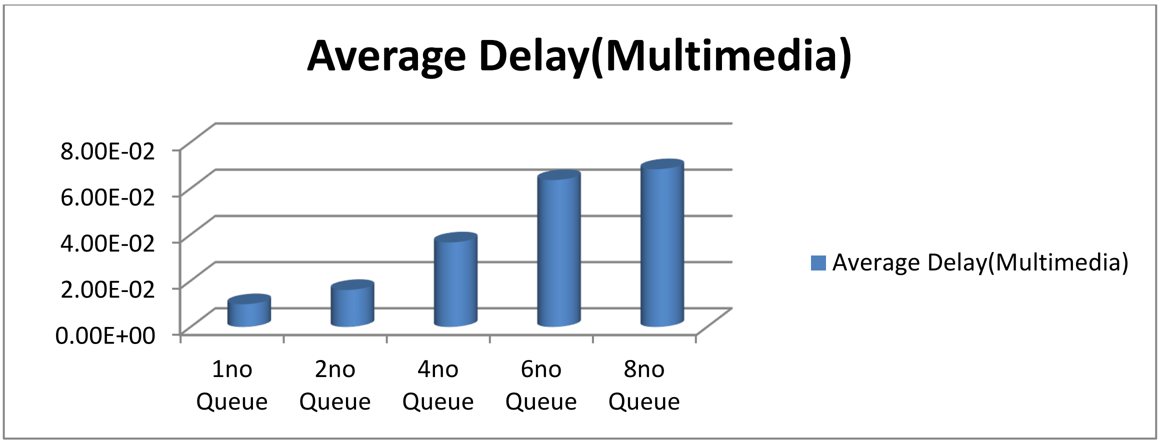


**Fig 49: No of Queues Varied-V-Average Delay (Multimedia)**

As the queue size is varied the average delay steadily increases and gets to a peak value when the queue size is highest, that is at the eight number queue sizes. The increases are quite large between 1 number queue size and six number queue size, but the increase between six number and eight number queue sizes is quite minimal.

# CHAPTER FIVE

# CONCLUSION

## 5.1: CONCLUSION

For this dissertation, the broad concepts of Quality of service (QoS) in wireless networks were studied with a view to understanding the effect of traffic types such as VoIP, Video, Multimedia enabled by QoS had on the network in terms of QoS parameters. The network architecture of IEEE 802.16/wiMAX was looked at with much emphasis on the MAC layer and the QoS mechanism characteristics present in these layer. Various services flows that are supported in wiMAX were studied. Literature analysis was carried out in detail to show current and ongoing thinking as regards QoS, and the future of a particular technology (wiMAX) as a result of its inherent QoS qualities.

VoIP, video, and multimedia traffic were analyzed using a Java simulation package, used prior to this time to analyse wifi by C .Xia,because of the similarities in the simplified uplink structure. The effect of different traffic types such as VoIP, Video, and Multimedia on QoS parameters like throughput, packet loss, and average delay were considered. In general, it was observed that video traffic experienced the most delay, closely followed by the multimedia traffic and the traffic with the least delay was the VoIP traffic .This goes to show that the video traffic has the most adverse effect on the network when being transmitted, closely followed by the multimedia traffic, the traffic with the least effect on the network is the VoIP traffic.

## 5.2: CHALLENGES ENCOUNTERED

The non- timely availability of opnet and ns-2 in the lab, so I could familiarise myself with the above mentioned software, proved the main challenge. But learning about wiMAX in such a short time span and writing a report was an exciting challenge of which I will cherish  for a long time

## 5.3: SUGGESTIONS FOR FUTURE RESEARCH

VoIP, Video and Multimedia Traffic were the three applications considered in this dissertation. Further analysis may be carried out on the effect of traffic types on mobile network, by further modifying the Java code used here to be able to depict not just the uplink (UL), but the downlink (DL) as well from the base station to the subscriber station by attempting to present code that will carry out the work of the classifier and scheduler at both the subscriber stations (SS) and the base station (BS), by choosing a particular scheduling algorithm of the several suggested see literature analysis.