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Quality of Service in WiMAX for Multimedia Services Qualité de Service en WiMAX pour les Services Multimédias

JURY

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Introduction

In the last two decades, we witnessed a huge increase in the usage of wireless communication technologies. With the advancements in these technologies, the service providers are now able to provide innovative multimedia services such as voice over IP (VoIP), IPTv, realtime gaming, and telemedicine etc. These services demand faster and more reliable wireless networks that could provide certain guarantees on throughput, delay, and other transmission parameters. This demand resulted in the development of several important wireless technologies, including the IEEE 802.16 standard. The standard is commercially known as WiMAX, which stands for Worldwide Interoperability for Microwave Access.

WiMAX is an important broadband wireless access standard that offers long-distance data transfer at high transmission rate. It aims to fulfill the requirements of IMT-Advanced specifications and thus becomes a fourth generation (4G) standard of digital cellular networks. It supports the triple-play of the Internet, telephony, and digital television. It is generally regarded as a disruptive technology [140]. Disruptive technologies are cheaper, simpler, and smaller. Therefore, they have the capability to completely replace established products [53]. The technologies and market sectors that are vulnerable to the disruption due to WiMAX include cable and satellite TV, third generation digital cellular networks, and backhaul industry [140].

In a point-to-multipoint (PMP) WiMAX network, a base station controls and manages the transmission of multiple subscriber stations. The QoS support is provided by a cooperation of the base station and subscriber stations. Ensuring QoS for various types of traffic is necessary to provide efficient transmission of both realtime and non-realtime traffics. However, furnishing QoS support in wireless networks is much more challenging as compared to wired networks. This is due to the unpredictable nature of wireless medium and the mobility of subscriber stations.

The standard provides support for both realtime and non-realtime traffics. Realtime traffic is divided into constant bit-rate (CBR) traffic (UGS class) and variable bit-rate (VBR) traffic (rtPS class). The scheduling of UGS class is straightforward and well-defined by the standard. However, the scheduling of rtPS class and non-realtime traffic (nrtPS and BE classes) is not defined in the standard. Scheduling rtPS class is the most challenging among all classes of traffic due to its bursty nature and strict delay constraints. Therefore the scheduler must ensure that the packets

are delivered before the deadlines are expired, otherwise the packets may be of no value to the receiver. Usually, video applications use rtPS class. These applications can tolerate some degree of packet loss. However, if many packets miss their deadlines and loss become significant, then it can seriously degrade the service level as perceived by the end-user. Therefore, this service type is given priority over non-realtime traffic. The scheduler must also make sure that lower priority classes also get acceptable level of service and no connection starve even under high load.

One of the most important functionalities of a QoS architecture is the scheduling of network traffic. A packet scheduling algorithm decides the order of transmission of data packets with the aim to ensure QoS for all service flows. The complex task of scheduling is performed by three distinct schedulers in WiMAX: base station uplink scheduler, base station downlink scheduler, and subscriber station scheduler. The most difficult part of scheduling is performed by the base station uplink scheduler. The base station uplink scheduler is responsible for scheduling packets from subscriber stations to the base station. These data packets are stored in queues that are maintained at subscriber stations. These queues are not directly accessible by the base station uplink scheduler and the scheduler cannot determine the exact sizes and deadlines of the stored packets. Therefore the uplink scheduler has to make decisions according to estimates. The functions of these schedulers are well-defined by the standard. However, the mechanisms to achieve this functionality are not defined by the standard. Therefore vendors and service providers can choose the scheduling algorithms that best suit their needs.

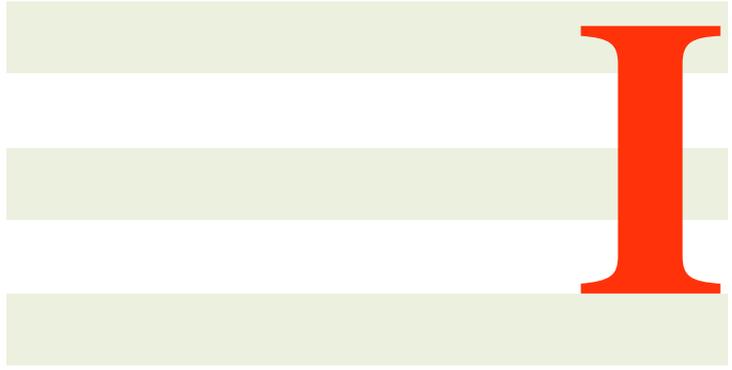
In this thesis, we provide a connection admission control and a two-level scheduling algorithm (TLSA) for the base station uplink scheduler. At the first level, uplink bandwidth is distributed among various service classes. Then at the second level, intra-class bandwidth distribution is done. TLSA aims to provide efficient and fair distribution of uplink bandwidth. The algorithm also prevents starvation of lower priority classes and ensures high resource utilization. The most important application of rtPS class is video streaming. Therefore, we also developed a video transmission framework that can be used to assess the performance of rtPS intra-class scheduling algorithm in transmitting video streams.

The dissertation is organized into two parts. In part 1, we present an overview of the IEEE 802.16 standard, its QoS architecture, and the uplink packet scheduling and connection admission control algorithms proposed for the standard. The part consists of two chapters. Chapter 1 presents an introduction of the 802.16 standard. After providing a brief description of wireless networks, the rest of the chapter deals with the 802.16 standard and the various technologies used in it that make it superior to existing wireless broadband technologies. Chapter 2 focuses on the QoS architecture provided by the IEEE 802.16 standard, and the packet scheduling schemes and admission control algorithms proposed for the standard. The chapter

begins by highlighting the importance of QoS in data networks and the various network components that are required to furnish QoS to user applications. Then it presents the details of the IEEE 802.16 QoS architecture. Next it provides a detailed classification of the connection admission control and packet scheduling algorithms proposed by the researchers.

In part II, we provide the details of TLSA and the associated connection admission control scheme. The part also provides theoretical and experimental analysis of the proposed algorithms. It ends by providing the details of a video transmission framework that is based on the proposed scheduling algorithm. The part consists of three chapters, i.e. chapters 3, 4, and 5. Chapter 3 provides the details of TLSA. It presents the working of TLSA and explains it with the help of flowcharts and illustrative examples. Furthermore, it provides an elaborative queuing model based on Markov chains that could be used to analyze the proposed algorithms. Chapter 4 presents the simulation analysis of TLSA. The simulations were performed by implementing the proposed algorithms in a commercial network simulator, namely Qualnet. The chapter provides results of in-depth simulation analysis of each scheduling class. In chapter 5, we provide the details of an H.264 compatible video transmission framework that is developed to assess the performance of TLSA in transmitting video streams.

Chapter 6 concludes the dissertation by providing the main conclusion and the paths for further research.

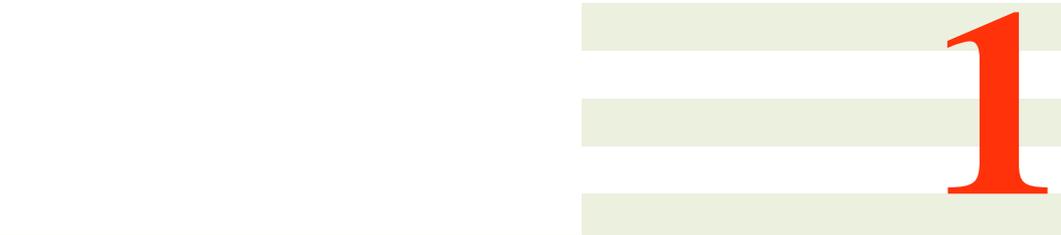


State of the Art

A communication network is a complex infrastructure that consists of several subsystems working together to enable communication between two or more entities. Traditionally, these networks used copper wires to provide connectivity to the communication devices. Later on, advancements in communication technologies enabled wireless networking. Wireless networks provide several benefits notably reachability and mobility. These advantages resulted in huge growth of mobile wireless networks. Modern broadband wireless access systems, such as WiMAX, provide support for data, realtime voice, high-quality video and many other services. Multimedia services, especially realtime services, have stringent QoS requirements. Therefore, furnishing adequate QoS support is crucial for the satisfactory operations of these networks.

The QoS in WiMAX is provided by the technologies implemented at the physical and MAC layers. To understand and appreciate the QoS architecture provided by WiMAX, it is important to understand the basics of communication networks in general and mobile wireless networks in particular. Therefore, we organize the state of the art into two chapters. The main focus of the first chapter is the building blocks of the IEEE 802.16 standard. The chapter begins by discussing the basis of communication systems and in particular wireless networks. It then provides the details of various procedures and technologies implemented at the WiMAX physical and MAC layers that work in coordination to ensure efficient working of these networks.

The second chapter is concerned with the QoS problem and the QoS architecture provided by the IEEE 802.16 standard. The two main elements of the QoS architecture, and the focus of our work, are connection admission control and packet scheduling. The IEEE 802.16 standard does not specify the algorithmic details of these components. Several researchers have proposed algorithms for these components. However, no proposal has been adapted yet by the standard and the area is still open for research. An extensive classification of the proposals on connection admission control and packet scheduling that are published in the literature is also provided in the second chapter. The two chapters furnish the background to understand the QoS problem in IEEE 802.16 standard and illuminates the significance for our work.



1

Digital Networks and the IEEE 802.16 Standard

The aim of this chapter is to present the IEEE 802.16 standard and the various technologies employed by the standard that make it superior to existing wireless broadband standards. However to facilitate the understanding of the standard, it is important to first discuss the basic building blocks of a communication network. Therefore, the chapter begins with an overview of data communication networks and their theoretical basis. Then, a description of wireless networks, their advantages and challenges are presented in Section 1.2. Next, we present an overview of the IEEE 802.16 standard. The historical evolution of the standard and various physical layer specifications are discussed in Section 1.3. Since the MAC layer plays a central role in providing QoS in 802.16 networks, therefore the design and working of the MAC layer is presented in detail in Section 1.4. In Section 1.5, we discuss various components and the interactions among them that enables end-to-end communication in 802.16 networks. Section 2.5 concludes the chapter.

1.1 Data Communication Networks

Data communication involves exchange of data between two or more devices through some sort of transmission medium such as a wire or radio waves. For data exchange to occur, the devices must be part of a communication systems called data network. A data communication system consists of five components [73],

Message The message is the data to be exchanged

Sender The sender is the device that sends the message

Transmission medium The transmission medium is the physical link that carries message from sender to the receiver

Receiver The receiver is the device that receives the message

Protocol A protocol is the set of rules that governs the process of data exchange

1.1.1 Network Reference Models

A communication network is a set of devices connected by communication links. A network is a complex system involving many hardware and software subsystems. Network reference models were introduced to facilitate the description of network protocols and services and the interactions among them. A reference model depicts how messages should be exchanged between any two devices in a telecommunication network.

The various subsystems in a network are normally provided by different manufacturers. In the absence of a common standard for communication networks, a communication device from one manufacturer may not be able to exchange information with devices from other manufacturers. The aim of network reference mod-

els is to guide manufacturers so that their products could interoperate with products from other manufacturers. The two most well-known network reference models are the Open System Interconnect (OSI) standard and the TCP/IP reference model. The details of these two models are presented below.

ISO Reference Model

In early 1980s, the International Standard Organization (ISO) developed the ISO standard 7498-1 [7] that concerns all the subsystems of a communication system. The standard is known as ISO reference model for open systems interconnection (OSI). The reference model was designed for computer networks with the aim to enable an application in a device to be able to communicate with an application in any other communication device irrespective of its manufacturer. The OSI model is probably the most influential network model.

To facilitate the design of a network, the ISO reference model divides network tasks into seven distinct layers. Each layer performs a well-defined task. Each layer has a communication interface with layers immediately above or below it. Each layer provides services to the layer immediately above it and uses the services of the layer immediately below it. The message sent by a particular layer is read by the same layer on the receiving side. The ISO reference model is shown in Figure 1.1. The seven layers of the ISO reference model are:

- Application layer
- Presentation layer
- Session layer
- Transport layer
- Network layer
- Data link layer
- Physical layer

Application Layer The application layer acts as a network entry point for user applications. It provides a well-defined interface to user applications and hide the details of communication system. The user applications in turn, provide interface to end-users and interact with the application layer to provide data transfer facilities to the users.

The services provided by the application layer protocols include message interchange, file transfer, access and management (authentication), and job transfer and manipulation. Several standard protocols are available to provide services at the application layer. Some examples of application layer protocols include File Transfer and Access Management (FTAM) protocol [6], X400 mail [9], and Common Management Information Protocol (CMIP) [8].

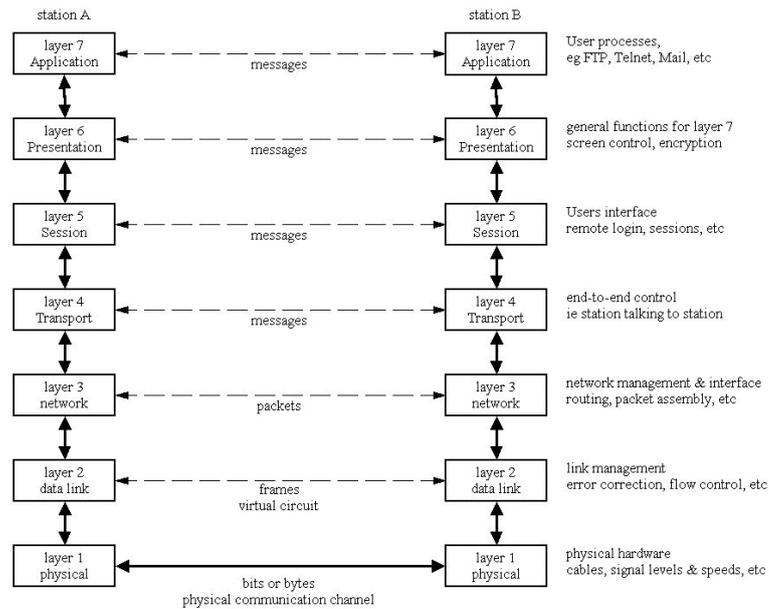


Figure 1.1: The ISO reference model

Presentation Layer The presentation layer converts the application layer data into a syntax-independent format, so that two application layer protocols can communicate irrespective of data syntax. In this way, it enables two communicating devices with different data formats to communicate with each other. The layer could also provide data security. For security, the data sent by the application layer is encrypted by the presentation layer. The encrypted data is then handed-over to the session layer.

Session Layer The session layer controls the connections between two communicating devices. It establishes, manages, and terminates the connection between two physical layer entities. The layer offers various services, such as authentication, authorization, dialog control, token management, synchronization, and exception reporting. Some examples of the protocols implemented at the session layer include Network Basic Input Output System (NetBIOS) [91], H.245 Call Control Protocol for Multimedia Communication [202], Remote Procedure Call Protocol (RPC) [197], Real-time Transport Control Protocol [171], and others.

Transport Layer The transport layer provides data transfer services to the upper layers. The layer provides various services to control the reliability of a communication link. These include flow control, segmentation of upper layer data, and error control.

The transport layer offers five classes of services ranging from class 0 (TP0) to class 4 (TP4). Class 0 provides only basic functionality such as connection establishment and data transfer. While class 4 provides more advanced features such

as error recovery and explicit flow control. The protocols that operate at the transport layer include Datagram Congestion Control Protocol (DCCP) [111], NetBIOS Frames protocol (NBF) [141], Reliable Datagram Protocol (RDP) [206], Stream Control Transmission Protocol (SCTP) [194], etc.

Network Layer The network layer provides end-to-end packet delivery and routing services, while maintaining the QoS requested by the transport layer. To ensure end-to-end packet delivery, the layer manages a host addressing scheme such that every device connected to the network has a unique identifier. The layer can optionally provide fragmentation/defragmentation and error reporting. Examples of the network layer protocol include Distance Vector Multicast Routing Protocol [209], Routing Information Protocol (RIP) [87], Internet Group Multicast Protocol (IGMP) [42].

Data Link Layer The data link layer offers control of the physical medium to provide the network layer with a reliable transmission medium. It is responsible for transferring data between two nodes on the same LAN or between two adjacent nodes in a Wide Area Network (WAN). The packets from the network layer are fragmented into more manageable units called frames.

The data link layer manages the local delivery of data between devices within a LAN. MAC frames do not cross the boundaries of a network. The data link layer focuses on local delivery, local addressing and media arbitration. It also provides services for error detection and correction. Some examples of the data link layer protocols are Address Resolution Protocol (ARP) [145], Asynchronous Transfer Mode (ATM) [15], Ethernet [5], Point-to-Point protocol (PPP) [185], Multiprotocol Label Switching (MPLS) [181], IEEE 802.11 wireless LAN [97], etc. The working of data link layer in wireless networks is discussed in Section 1.2.

Physical Layer The physical layer defines electrical and mechanical specifications of communication devices and the transmission medium. This includes pin configuration, voltage levels, cable specifications, network adapter etc. Another important function of the layer is modulation of digital data and then transmission of corresponding signals over the physical link. The details of the physical layer in wireless networks are provided in Section 1.2.

Internet Protocol Suite

The Internet Protocol suite, also known as transmission control protocol/Internet Protocol (TCP/IP) model, is the most commonly used practical network model. Similar to ISO reference model, the TCP/IP model separates networking tasks in

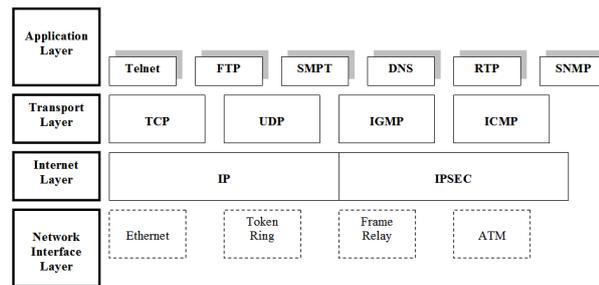


Figure 1.2: The Internet Protocol Suite [201]

various layers. Each layer provides services to the layer immediately above it and its working is transparent to the layers above and below it.

There are four layers in TCP/IP model: Application layer, Transport layer, Internet layer, and Network layer. The top most layer of the TCP/IP protocol suite provides functionality equivalent to the top three layers of the ISO model. Conceptually, the TCP/IP model can be represented as Figure 1.2.

Application Layer The application layer provides communication interface to the user applications. The functionality provided by the top three layers of the OSI reference model (application layer, presentation layer, and session layer) is combined in the application layer in the TCP/IP model. Some examples of the application layer protocols in the TCP/IP protocol stack include: Hypertext Transfer Protocol (HTTP) [71], File Transfer Protocol (FTP) [150], Telnet [149], and Simple Mail Transfer Protocol (SMTP) [110].

Transport Layer The transport layer provides end-to-end communication facility. The layer provides both connection-oriented (reliable) and connection-less (unreliable) data exchange between two end-points. Some examples of the transport layer protocols include Transmission Control Protocol (TCP) [99], User Datagram Protocol (UDP) [146], and Realtime Transport Protocol (RTP) [171].

Internet Layer The Internet layer transforms the user data into packets, which can be sent to the destination. The layer also provides a logical addressing scheme, such that every device connected to the network can be uniquely identified. Routing of data packets is also done at the Internet layer. Examples of the Internet layer protocols include Internet Protocol (IP), Internet Message Control Protocol (ICMP) [147], and Address Resolution Protocol (ARP) [145].

Network Access Layer The network access layer manages the physical transmission link. It specifies how the data is sent through the physical link. Ethernet [5],

token ring [131] and Neighbor Discovery Protocol (NDP) [135] are examples of protocols that work at the network access layer.

1.2 Wireless Networks

Wireless network refers to any type of computer networks that connect devices without using wires or cables [83]. Wireless networks are implemented and administered using radio waves. This implementation is done at the physical layer and the data link layer of the ISO reference model. Wireless networks offer many advantages: [105]:

Mobility A wireless network enable its users to access information from anywhere without having a wired connectivity

Reachability Wireless networks allow people to be better connected and reachable without the limitations of location

Easy to Deploy Often, wireless networks are much easier to deploy then wired networks

Low Initial Cost For big telecommunication networks, the initial setup cost of wireless networks is cheaper than wired networks

Maintainability Much easier to maintain than wired networks

New Services Wireless systems enable vendors to provide new services such as Short Message Service (SMS) and Multimedia Messaging Service(MMS)

Characteristics and Challenges

The characteristics of wireless medium are fundamentally different from that of wired networks. The main characteristics of wireless networks that make it different from wired networks are summarized below.

- Wireless medium is unpredictable
- Radio spectrum is regulated by government agencies and therefore it could be very expensive
- Radio spectrum is a scarce resource that must be distributed among various, often competing systems
- Users can be mobile
- Radio signals always experience some degree of attenuation and noise
- The nodes are spread in a three-dimensional space
- Due to limited battery size, wireless devices have limited energy

Due to these characteristics, designing a wireless communication system is very challenging. The major challenges faced by wireless network designers are given below.

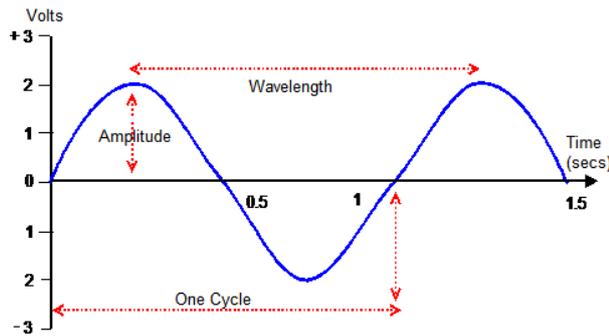


Figure 1.3: Basic characteristics of a radio wave

- To be portable, a wireless communication device must be small
- Transmission and signal processing must use the minimum possible energy
- Due to mobility and continuously changing medium conditions, users' demands change and it may not be easy to adapt to these changes
- Fair and efficient allocation of radio resources
- Due to broadcast nature of radio signals, security is much harder to achieve than wired networks
- The network must be able to locate a node among all its distributed subscriber stations and route messages to it which might be moving at high speed
- Support for multiple mode of operation for service and data

Physical Layer

In wireless communication, electromagnetic waves are used for data transmission. The basic characteristics of radio waves such as amplitude, frequency, and phase are altered continuously to encode and send data over radio waves. Amplitude is the measure of power of wave. It is measured in volts. Phase is the measure of shift in the position of a radio wave. It is measured in degrees. Wavelength is the distance between successive peaks of a periodic wave. The number of cycles a wave complete in one second is called frequency. Frequency is measured in hertz (Hz). Signals with longer wavelengths have lower frequencies and vice versa. Transmission is normally done using a band of frequencies. The frequency band needed to transmit a signal is called its bandwidth.

Wireless networks can be used to transmit both analog and digital data. For analog transmission the message is first encoded on a carrier wave by modifying one of its basic properties. The process is called modulation. A modulated radio wave is called a radio signal. The radio signal is then transmitted over air. When the signal is received, the information is extracted from it at the receiver by a process known as demodulation. Frequency Modulation (FM) and Amplitude Modulation (AM) are common examples of analog transmission. In digital transmission, the

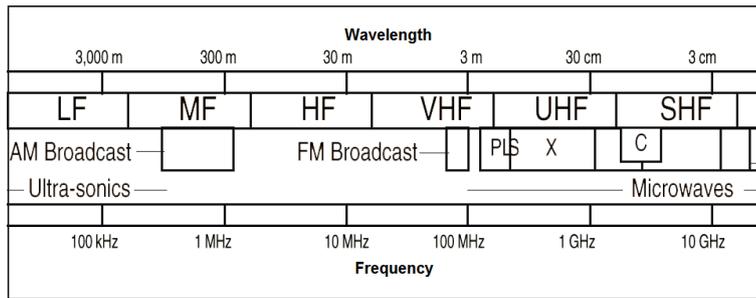


Figure 1.4: The radio spectrum [4]

message is represented as binary digits (0 and 1). These binary digits are then sent as series of ON/OFF electrical pulses.

Each radio signal is made up of a number of radio waves at various frequencies. The range of frequencies that made up a signal is called spectrum. The total radio spectrum is divided into groups of radio frequencies called “bands”. Most commonly used bands are usually identified by names such as very low frequency (VLF), high frequency (HF), very high frequency (VHF) bands, etc. The radio spectrum nomenclature is shown in Figure 1.4.

There are many factors that affect the transmission of radio waves such as attenuation, signal deviation, interference, and noise etc. Attenuation refers to decrease in signal strength as it moves through the atmosphere. The degree of attenuation increases in the presence of moisture in the air. Higher frequency signals experience more attenuation than lower frequency signals.

Radio waves bend, due to changes in the density of the air, as they move from one atmospheric layer to the next. The bending of radio signals is called refraction. Moreover, lower frequency waves are reflected back to the earth by the top-most layer of the atmosphere. This reflection enables low frequency radio signals to be transmitted to thousands of kilometers.

At frequencies above 30 MHz, radio signals experience almost no deviation from transmitter to receiver. This type of transmission is called line-of-sight transmission. For this type of transmission, differences in the temperature of the atmosphere or presence of high percentage of moisture in the air can cause radio signals to travel far beyond the normal distance. This phenomena is known as superrefraction. Interestingly, atmospheric conditions can cause a signal to bend away from the earth, which results in reduction of the transmission distance. These phenomenons complicates the design of radio systems and spectrum management.

The nature of high frequency waves is quite different from lower frequency waves. Higher frequencies experience more attenuation and these waves easily pass through the atmosphere. Therefore communication at these frequencies (between 30MHz and 30GHz) are limited to line-of-sight distances. This limitation is an advantage as the same frequencies can be reused beyond the transmission range.

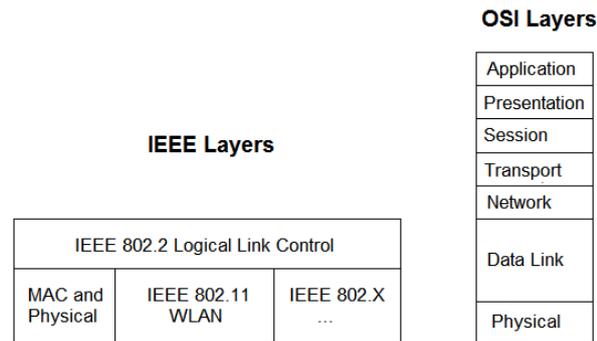


Figure 1.5: The IEEE 802 family of protocols

Frequencies in this range are used by many important services, such as television broadcast and cellular phone networks. The portion of the spectrum between 1 GHz and 3 GHz is particularly valuable for mobile communications and for new broadcasting technologies. Fourth Generation (4G) mobile phone standards such as IEEE 802.16 and Longterm Evolution Advanced (LTE-A) make use of this frequency range.

Data Link Layer

The data link layer is responsible for data link control and media access control. Data link control manages the communication between two adjacent nodes, while the media access control deals with sharing the physical link. One of the most commonly used wireless standard is the IEEE 802.11 standard [78], which is also known as WiFi. The structure of the data link layer of 802 family is shown in Figure 1.5.

The main functions of the data link control include framing, flow control, error control, and reliable transmission. Framing packages data sent by the network layer into small data units and then add information required for successful transmission. These data units are called frames. Frames can be either fixed-size or variable size. For example ATM networks use fixed-size frames, while the IEEE 802.3 local area networks uses variable size frames.

Flow control regulates the rate at which data can be sent from source to destination. Each receiver has a limited processing power and limited amount of memory. Therefore, the flow control is necessary to prevent sender to overwhelm the receiver.

Error control involves procedures to detect and correct errors. If the receiver detects an error, it can request the sender to resend the lost data. This procedure is called automatic repeat request (ARQ).

The second sublayer of the data link layer, that is the media access layer, resolves access to the shared media. The layer ensures only one node can send data at a time, other nodes do not interrupt the sender and no node could monopolize the

medium.

Wireless networks have inherent characteristics that make them different from wired networks. Due to particular characteristics of radio channel, the task of data link layer is much more complex in wireless networks. The main challenges faced by wireless data link control and MAC layer can be summarized as follow [26]:

1. In wireless networks, a node address does not equal to a node location. It is the duty of the data link layer to locate a node among millions of mobile stations.
2. As radio signals are not protected from other signals, therefore wireless systems must have robust error prevention and correction schemes.
3. Radio waves do not have definite boundaries, so security efforts could not be implemented only at boundaries.
4. The medium is significantly less reliable and it has time-varying and asymmetric propagation properties. Therefore, the data link layer must be capable of coping with the changing conditions.
5. Wireless networks have dynamic topologies.
6. Wireless networks lack full connectivity, therefore the sender and receiver may not be visible to each other.
7. For mobile stations, power management is an important issue

1.3 Overview of the 802.16 Standard

1.3.1 History

The IEEE 802.16 standard is commercially known as Worldwide Interoperability for Microwave Access (WiMAX). The name WiMAX was coined by WiMAX Forum [220] as a marketing term in 2001. Initially WiMAX was meant to provide high-speed wireless data communication to fixed subscriber stations. WiMAX Forum defined the first specification of WiMAX as “a standards-based technology enabling the delivery of last mile wireless broadband access as an alternative to cable and digital subscriber line (DSL)”.

The first specifications [92] of the IEEE 802.16 standard were published by the IEEE in 2001. The objective was to provide a high speed, point-to-point and line-of-sight connectivity between fixed stations. The targeted market was rural communication networks where cable connection was not feasible. Prior to the 802.16 standard, this market segment was dependent on proprietary applications and poor interoperability often resulted in unnecessary complications. The 802.16 standard provides this market segment the standardization and interoperability needed for faster and economical deployment and operation.

Standard	Description
802.16-2001	Fixed broadband wireless access (10-66 GHz)
802.16a-2003	Physical layer and MAC definitions for 2-11 GHz
802.16d-2004	Air interface for fixed broadband wireless access system
802.16e-2005	Mobile broadband wireless access system
802.16-2009	Air interface for both mobile and fixed broadband wireless access system
802.16m-2011	Advanced air interface to fulfill IMT-Advanced requirement on 4G mobile systems

Table 1.1: Evolution of IEEE 802.16

In 2003, the scope of the standard was expanded to provide direct services to end-users. This expansion in the scope was brought in by the IEEE 802.16a specifications [93]. The targeted market was wireless local loop (WLL) [215]. The standard aimed to provide end-users high speed broadband access by wireless means. Then in 2004, the IEEE 802.16 and the IEEE 802.16a specifications were unified into the IEEE 802.16d (802.16-2004) [94] specifications. This standard is known as Fixed WiMAX.

In 2005, the IEEE 802.16 working group further extended the scope of the standard to incorporate support for mobile users. These specifications are known as IEEE 802.16e [95] or Mobile WiMAX. Mobile WiMAX is based on wireless broadband technology called WiBro [115]. The extension brought mobile cellular communication market within WiMAX domain. This put WiMAX in direct competition with existing and upcoming cellular phone technologies. Then, in 2011, the IEEE ratified the 802.16m [96] air interface with 100 Mbit/sec data rates for mobile stations and up to 1 Gbit/sec data rates for fixed stations. This standard is known as WirelessMAN-Advanced or WiMAX Release 2. The aim of the current release is to fulfill the ITU-R IMT-Advanced [11], [153] requirements for 4G wireless cellular communication systems.

1.3.2 Physical Layer Specifications

WiMAX is designed to deliver high throughput over large distances with high degree of reliability. The IEEE 802.16 working Group D designers used a number of excellent technologies to achieve this performance level. The standard offers a variety of tools for tailoring the transmission for various operational environments including several level of modulations, variable cyclic prefix length, multiple error-correction coding types, and various multiple-input multiple-output technologies including spatial multiplexing and beam-forming to mention only a few [85]. In the subsequent paragraphs we provide an overview of the WiMAX physical layer and its different variants, and the major technologies implemented and supported by the WiMAX physical layer.

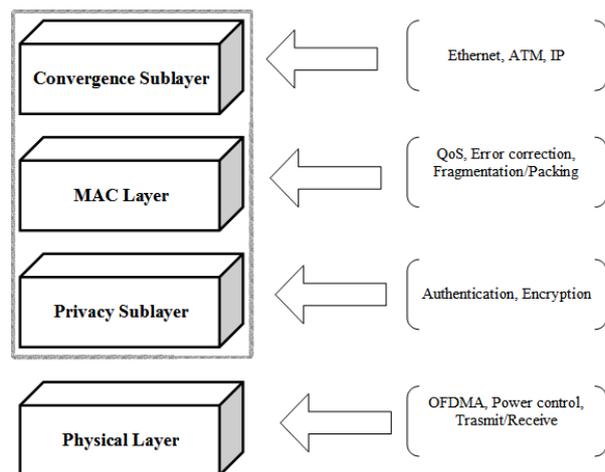


Figure 1.6: The Physical and data link Layers of IEEE 802.16 [140]

The 802.16 standard is an extension of IEEE 802, which is the major technology used in Ethernet. The major change is the use of wireless medium and the layer that controls the physical medium i.e. data link layer. These changes are transparent to the above layers of the ISO reference model. Figure 1.6 shows the 802.16 physical and data link layers.

Technologies used in WiMAX Physical Layer

The IEEE 802.16 standard uses many exciting technologies to provide very high data rates with high reliability. In this section, a brief description of the important technologies used by the standard are provided.

Modulation Modulation is the process of encoding information over a carrier wave by varying one or more properties of a periodic waveform. WiMAX physical layer supports phase shift keying and quadrature amplitude modulation techniques. A brief description of these technologies is given below.

Phase Shift Keying (PSK) In PSK, the phase of the carrier wave is varied according to data elements being modulated. Both peak amplitude and frequency of the wave remain constant. The simplest form of PSK is binary phase shift keying (BPSK), which employs two phases to represent one bit per signal element. One of the signal elements has a phase of 0 degree, while the other has a phase of 180 degrees. A more sophisticated scheme is quadrature PSK (QPSK). In QPSK, four phases are used to represent two bits per signal element. Thus for the same bit rate, the signal rate is effectively reduced to half. Conceptually, PSK modulation can be represented by constellation diagrams as shown in Figure 1.7.

PSK is simple and superior to amplitude shift keying and frequency shift keying. It is less susceptible to noise and requires only one carrier to modulate a binary data

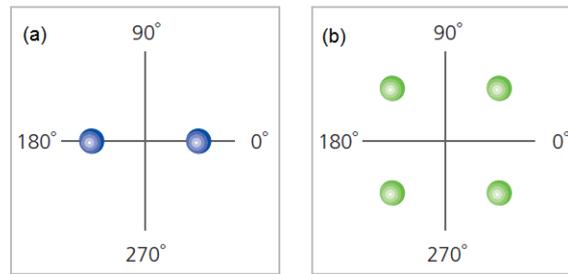


Figure 1.7: (a) Binary phase shift keying (b) Quadrature phase shift keying

stream. However, very high data rates cannot be achieved using PSK due to the fact that the receiving equipment must be very sensitive to distinguish small differences in phase.

Quadrature Amplitude Modulation (QAM) In QAM, both phase and amplitude of the carrier wave are varied simultaneously for modulation. The idea is to use two carriers simultaneously, one in-phase and other out-of-phase (quadrature), with different levels of amplitude. Due to the presence of both amplitude and phase variations, it may be considered as mixture of amplitude and phase modulations. High spectral efficiencies can be achieved through QAM. The only limiting factor to performance is the level of noise and linearity of the communications channel.

By using various levels of amplitudes and phases many variations of QAM are possible, such as 4-QAM, 16-QAM, 64-QAM, etc. The number in the prefix designates the total number of signal elements used for modulation. These schemes can be graphically represented by constellation diagrams, as shown in Figure 1.8. Modern telecommunication systems designed for high speed data transfer usually employ dense QAM constellations. The 802.16 physical layer usually use 64-QAM or 256-QAM. By using a high order QAM, high throughput can be achieved but with lower reliability. QAM is more susceptible to noise. Another drawback is the requirement of linear amplifiers in QAM, which results in greater power consumption and therefore may not be suitable for some networks.

Multiplexing Multiplexing is the process of combining more than one signals into one signal that could be sent over a shared medium. The two most common types of multiplexing are frequency-division multiplexing (FDM) and time-division multiplexing (TDM).

In FDM, multiple signals are transmitted on a shared channel by assigning them non-overlapping frequency bands (sub-channels). Thus, multiple signals can be sent and received at high speed. However, if many signals are to be transmitted, the necessary bandwidth could be large.

In TDM, multiple signals are transmitted on the same channel by assigning turns

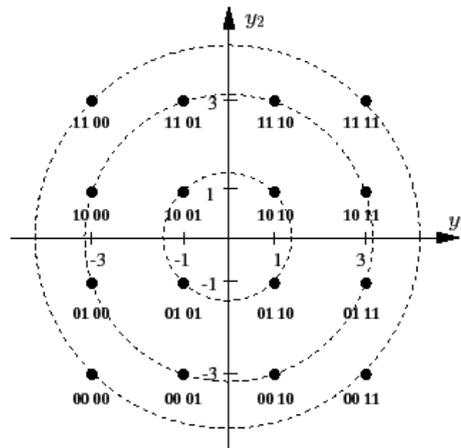


Figure 1.8: Constellation diagram of QAM

in succession. Each signal is given a specific time-slot in round-robin fashion. However, these schemes are not efficient enough to be used in very high speed data networks such as WiMAX. Instead WiMAX uses a much more efficient multiplexing scheme, called orthogonal frequency division multiplexing.

Orthogonal Frequency Division Multiplexing (OFDM) OFDM is the core technology that makes WiMAX a significant improvement over existing wireless communication technologies. OFDM was developed in 1970 by Bell Labs. It has become a popular scheme for wideband digital communication. OFDM has already been incorporated in various communication standards such as Digital Subscribers Line (DSL) [113], Digital Audio Broadcasting (DAB) [66], Digital Television DVB-T/T2 [67] (terrestrial), DVB-H [69] (handheld), DVB-C2 [12] (cable), Digital Multimedia Broadcasting (DMB-T/H) [51], and IEEE 802.11 Wireless LAN [78] just to mention a few.

OFDM is based on Fast Fourier Transform [210] that enables encoding data on multiple carrier frequencies such that the channels do not lose their individual characteristics. The property that multiple channels can be combined together in one signal without losing their individual characteristics is called orthogonality. The sub-channels are chosen so that they are orthogonal to each other, therefore crosstalk between the sub-channels is eliminated and inter-carrier guard bands are not required. This results in a much simplified design of both the transmitter and the receiver. The orthogonal sub-carriers are closely spaced and carry data on several parallel streams. Each sub-carrier is modulated with a conventional scheme (such as PSK or QAM) at a low symbol rate, keeping data rates similar to conventional modulation schemes in the same amount of bandwidth. The low symbol rate allows to introduce guard interval between symbols and to eliminate inter-symbol interference. Furthermore, time-spreading can be used to achieve high diversity gain and thus high signal-to-noise ratio.

The FFT algorithm on the sender and the inverse FFT on the receiver enable efficient and simplified modulator and demodulator implementations. Another major advantage of OFDM over single-carrier schemes is its higher resistance to interference, frequency-selective fading due to multipath propagation, attenuation and delay spread degradation. Therefore, with OFDM a wireless signal can be transmitted much farther than with competing technologies.

OFDM is used in conjunction with forward error correction coding and almost always uses frequency or time interleaving. Frequency interleaving increases resistance to frequency-selective fading. For example, when some parts of the channel fades, frequency interleaving makes sure that the bit errors that would result from those sub-carriers are spread out in the bit-stream. Similarly, time interleaving ensures that adjacent bits in the original bit-stream are transmitted far apart in time, thus reducing fading as would happen at high speed. In this way, interleaving spread the errors in the bit stream such that error correction decoders are not presented with high concentration of errors. In case of high concentration of errors, the decoders could not correct all errors and burst of uncorrected errors occurs.

OFDM requires accurate frequency synchronization between the transmitter and the receiver, otherwise the sub-carriers will no longer be orthogonal and therefore inter-carrier interference (ICI) would occur. Frequency offsets could be caused by mismatched transmitter and receiver or by Doppler shift [203]. Doppler shift with multipath propagation is very difficult to correct. This phenomena worsens as subscriber station speed is increased, and it is a major factor in limiting the use of OFDM in very high speed vehicles.

Duplexing Mode The 802.16 standard supports both time division duplex and frequency division duplex.

Time Division Duplex (TDD) In TDD, transmission and reception is done over same frequency channel by allocating separate time-slots for uplink and downlink in succession. Since only one frequency channel is used, therefore no band separation is required and the transmission medium can be used efficiently in both directions.

TDD emulates full-duplex communication over a half-duplex link. It is suitable when traffic patterns are variable and there is asymmetry in the uplink and downlink data rates. TDD is well-suited for Mobile WiMAX due to the following reasons [162]:

- Data traffic patterns in mobile networks are asymmetrical. TDD allows operators to adjust downlink and uplink rates according to evolving needs
- TDD is better suited for advanced antenna techniques such as adaptive antenna systems

However, TDD is not very suitable in case of symmetric traffic, because some bandwidth is wasted in shifting from transmit mode to receive mode and vice versa. Another drawback is that guard intervals are required between neighboring base stations, so that they can transmit and receive simultaneously. This results in reduced spectral efficiency and complex circuitry.

Frequency Division Duplex (FDD) In this scheme, separate channels are used for transmission and reception within the operating band i.e. the transmitter and the receiver operate at different carrier frequencies [73]. To prevent interference, the uplink and the downlink sub-channels are separated by a frequency offset.

FDD is more efficient when the pattern of traffic is symmetrical. It has lesser inherent latency and requires simpler circuit than that for TDD communication system. Unlike TDD, base stations communicate on different sub-channels and therefore do not interfere with each other. This makes radio planning simpler and more efficient. Paired band regulations may force some operators to deploy WiMAX in FDD mode only.

For satisfactory communication, the transmission and reception must be done on separate channels separated by guard band. This reduces the spectral efficiency. Furthermore, it is usually not possible to reallocate spectrum to alter the capacity in either directions if the bandwidth requirements change for each direction.

Media Access

Time Division Multiple Access (TDMA) In TDMA, subscriber stations share the resources of a shared channel in time. Access to the shared medium is provided to several users by dividing the channel into time-slots. The users communicate in succession in their allotted time-slots. This allows multiple users to share the same radio frequency channel, while only using a part of the channel capacity. Each user has full control of the medium in its allotted time-slots. To prevent interference guard bands are used. TDMA is simpler to implement and intuitive. TDMA is used in Global System for Mobile Communications (GSM) [14], Digital Enhanced Cordless Communications (DECT) [64], Personal Digital Cellular (PDC) [192], satellite systems, combat-net radio systems [102], and many others.

Frequency Division Multiple Access (FDMA) FDMA is a technique in which multiple access to a shared medium is provided by assigning separate frequency bands to individual users. The available frequency band is divided into separate sub-channels and a subset of the sub-channels is assigned to each user. In this way, a part of the frequency band is reserved for a specific user and it belongs to that user throughout the communication.

Each sub-channel is isolated from neighbouring sub-channels by guard bands to minimize the interference due to crosstalk. Unlike TDMA, FDMA is not vulnerable to synchronization problems. FDMA is commonly used in satellite communications.

Space Division Multiple Access (SDMA) SDMA is a channel access method that is enabled by employing directional antennas at the sender and/or receiver. Unlike traditional cellular technologies, where the base station radiates signals in all directions without any knowledge of the receiver, SDMA enables access to a channel by identifying the user location. Thus, it results in mitigation of interference and noise from adjacent sources. Furthermore, it avoids redundant transmission in areas where no active user is currently present. SDMA is used in 4G cellular systems including WiMAX and LTE-A.

Orthogonal Frequency Division Multiple Access (OFDMA) OFDMA is the preferred multi-access channel scheme in Mobile WiMAX. It has become the de-facto choice for broadband wireless technologies [162], such as 3GPP LTE [3] and 3GPP2 Ultra Mobile Broadband [13]. In OFDMA, multiple access is achieved by assigning subset of sub-carriers to each user. This allows several users to communicate simultaneously with the base station.

OFDMA is more efficient than OFDM [229] and it further enhances OFDM robustness to fading and interference. Furthermore, lower transmission power is required for low data rate users. In Mobile WiMAX, an advanced version of OFDMA called scalable OFDMA (SOFDMA) [224] is used. SOFDMA provides following benefits [162]:

1. Superior performance in none-line-of-sight (NLOS) multipath communication
2. Efficient use of available spectrum by time and frequency sub-channelization
3. Relatively simple transceiver structure
4. Good compatibility with advanced antenna techniques such as multiple-input multiple-output (MIMO) and adaptive antenna systems
5. High scalability that enables economical service customization

Performance Enhancement Technologies

Adaptive Antenna System (AAS) AAS is an optional feature in WiMAX that enables the transmission of downlink and uplink bursts using directed beams. AAS works by adjusting the power and direction of antenna radiation patterns. An array of antennas is used at the base station, which can focus the signal strength in the

direction of the intended receiver. This results in increased gain to the intended receiver, while reduced interference to and from other receivers and interference sources.

AAS enables SDMA, which allows multiple subscriber stations separated in space to communicate simultaneously on same subchannel. The technique is very effective in improving performance and coverage especially in Mobile WiMAX. Furthermore, it allows subscriber stations to deliver channel quality feedback to the base station.

Multiple-Input Multiple-Output (MIMO) MIMO is the use of multiple antennas at the sender and the receiver to achieve high data rates and higher level of reliability. It offers significant improvement in throughput and transmission range without additional bandwidth or increased transmit power. The total transmit power is spread over multiple antennas to achieve an array gain and diversity gain. This gives improved spectral efficiency and better link reliability. Mobile WiMAX was the first technology to incorporate MIMO technology [162].

Dynamic Frequency Selection (DFS) In wireless communication systems, channel allocation schemes are required to allocate frequency bands to base station and subscriber stations to achieve good spectral efficiency at the minimum possible interference levels. A channel allocation scheme can be either fixed or dynamic. Fixed channel allocation requires manual frequency planning and it could be tedious in big networks. DFS eliminates the tedious manual frequency planning and utilizes cellular radio resources much more efficiently.

Adaptive transmission The resilience to severe channel conditions can be further enhanced if the information about the channel state is sent back to the sender over a return-channel. Based on this feedback information, adaptive modulation, channel coding and power allocation may be applied across all sub-carriers, or individually to each sub-carrier. In the later case, if a particular range of frequencies suffers from interference or attenuation, the carriers within that range can be disabled or made to run slower by applying more robust modulation or error coding to those sub-carriers. Adaptive modulation allows almost instantaneous adaptation of modulation scheme for optimal data transfer. Thus it enables efficient use of bandwidth and support for a rich set of applications.

Antenna Diversity (Space Diversity) Diversity schemes are used for improving reliability of a signal by utilizing two or more communication channels with different characteristics. Alternatively, a forward error correction code (FEC) may

be added and different parts of the message transmitted on different channels. Different channels are effected by different levels of fading and interference, therefore the effect of fading and co-channel interference can be minimized.

Antenna diversity is a wireless diversity scheme in which multiple antennas are used to improve the quality and reliability of a wireless link. Thus, a receiver has several observations of the same signal, which may provide a robust link. Antenna diversity is particularly useful in mitigating the effects of multipath propagation.

Another diversity scheme, called transmit/receive diversity, uses two collocated antennas for transmit and receive functions. This eliminates the need for a duplexing module and it can protect sensitive receiver components from the high power used in transmission.

Frequency Hopping Spread Spectrum (FHSS) FHSS is a method of transmitting radio signals by rapidly switching a carrier among many frequencies. The switching is done according to a special sequence that is known only to the transmitter and the receiver.

The total bandwidth required to transmit a signal using FHSS is much higher as compared to single carrier transmission. However at any time, only a small portion of the bandwidth is used for transmission and thus the effective interference bandwidth is small. An advanced version of FHSS, called adaptive FHSS, improves resistance to radio frequency interference by selecting uncrowded frequency bands from a range of permissible bands.

FHSS has three main advantages [73]:

1. Highly resistant to narrow-band interference
2. Difficult to intercept and jam
3. Possibility of sharing a frequency band with many types of conventional transmission with minimal interference

Direct Sequence Spread Spectrum (DSSS) DSSS is a spread spectrum modulation technique, in which a sine wave is modulated by a continuously generated pseudorandom noise code symbols called chips. In fact, the data being transmitted is multiplied by pseudorandom sequence of 1 and -1 at a much higher frequency than that of original signal.

The resulted signal is white noise. The original signal can be reconstructed at the receiver by multiplying it with the same pseudorandom sequence. This process is known as de-spreading. DSSS provides process gain, which can be enhanced by employing a longer noise sequence and more chips per bit. If an undesired transmitter transmits on the same channel but with different code, then DSSS results

in no processing gain for that signal. Thus, it enables multiple transmitters to share the same channel.

Supported Frequency Bands The IEEE 802.16 standard uses both licensed and unlicensed bands. The licensed frequency bands used are 10-66 GHz and 2-11 GHz. While the unlicensed bands used by the standard are between 2 and 11GHz.

10-66 GHz Band This band uses very short wave lengths for line-of-sight communication. None-line-of-sight transmission is not feasible in this frequency range due to high multipath propagation losses. The motivation in using this band is the abundant availability of bandwidth. Channels within this bands are often more than 25MHz wide and most frequency bands can provide several hundred megahertz of bandwidth. Therefore, very high data rates are achievable with line-of-sight transmission. 10-66 GHz band is particularly suited for establishing backhaul links.

2-11 GHz Band In 2-11 GHz frequency range, WiMAX utilizes both licensed and license-exempt bands. All of the OFDM variants operate in this band. These variants are specially designed for non-line-of-sight operation such as residential use. OFDM based systems are more suitable for none-line-of-sight transmission due to the simplicity of the equalization for multi-carrier signals [81]. Communication in the license-exempt band experiences high interference due to crowding, therefore mechanisms like dynamic frequency selection is used to detect and avoid interference.

IEEE 802.16 Physical Layer Variants

There are five variants of WiMAX based on physical layer specifications. Two of the specifications are single-carrier based while others use OFDM. The details of the various physical layer specifications can be found in [98]. Table 1.2 provides an overview of 802.16 physical layer variants.

WirelessMAN-Single Carrier (WirelessMAN-SC) This is 10-66 GHz physical layer point-to-multipoint specification. In this variant, a single carrier is modulated with adaptive modulation and error coding schemes. Transmission in the downlink direction employs TDM, while access in the uplink direction is provided through a combination of TDMA and Demand Assigned Multiple Access (DAMA). An uplink subframe for WirelessMAN-SC is shown in Figure 1.9, while the structure of downlink subframe is shown in Figure 1.10.

The variant supports both TDD and FDD to separate uplink and downlink transmissions. Half-duplex FDD subscriber stations are also supported, which may

Specifications	Transmission	Function	Frequency Band	Duplexing Mode
WirelessMAN-SC	LOS	Point-to-point	10-66 GHz	TDD, FDD
WirelessMAN-SCa	NLOS	Point-to-point	2-11 GHz	TDD, FDD
WirelessMAN-OFDM	NLOS	Point-to-multipoint	2-11 GHz	TDD, FDD
WirelessMAN-OFDMA	NLOS	Point-to-multipoint	2-11 GHz	TDD, FDD
Wireless HUMAN	NLOS	Point-to-multipoint	2-11 GHz	TDD only

Table 1.2: Various implementations of the WiMAX physical layer

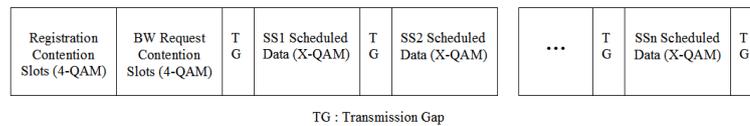


Figure 1.9: Uplink subframe of WirelessMAN-SC [94]

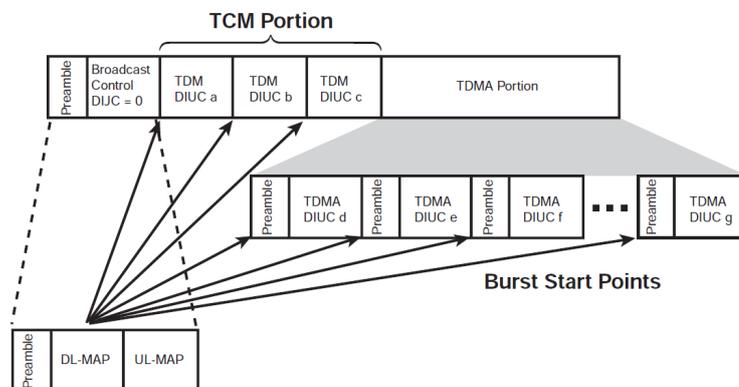


Figure 1.10: Downlink subframe of WirelessMAN-SC [94]

be less expensive than full-duplex FDD subscriber stations [129]. The standard also specifies details of error correction, modulation, and coding schemes. Reed-Solomon [217] is used in conjunction with inner block convolutional code [126] to robustly transmit critical data. Various degree of robustness and effectiveness could be achieved by combining FEC with QPSK, 16-QAM, and 64-QAM.

WirelessMAN-Single Carrier Access (WirelessMAN-SCa) WirelessMAN-SCa is designed for none-line-of-sight communication. A single-carrier modulation scheme in the frequency range 2-11 GHz is used. Both TDD and FDD duplexing modes are supported. Multiple access is enabled by TDMA in both uplink and downlink directions.

Block adaptive modulation is used with FEC coding for robustness and effectiveness. Both MAC layer and physical layer are designed to support optional AAS feature. The standard provides improved equalization and channel estimation performance over none-line-of-sight communication.

WirelessMAN-OFDM WirelessMAN-OFDM is based on OFDM with a 256-point transform to support multiple subscriber stations in 2-11 GHz frequency band. The air interface has been adopted as the physical layer specifications for broadband wireless access systems and it is the mandatory interface for license-exempt frequency bands. It is mainly developed to support fixed subscriber stations in residential and business areas.

Multiple access is provided through TDMA. Both TDD and FDD duplexing modes and half-duplex FDD subscriber stations are supported by the interface. BPSK, QPSK, 16-QAM, and 64-QAM modulation schemes are supported. The standard also supports transmit diversity and AAS with support to SDMA.

The interface supports transmit diversity using Space Time Coding (STC) [16] in the downlink direction. Two antennas are used at the base stations to transmit an STC-encoded signal. The antennas first transmit two different signals, and then transmit the complex conjugate of the same two symbols. The two signals are combined at the receiver to obtain a gain due to second-order diversity.

WirelessMAN-OFDMA The air interface uses OFDMA with 2048-point transform to provide both fixed and mobile broadband wireless access. Multiple access is provided by allocating a subset of the sub-carriers to individual users. The specification supports both TDD and FDD operations.

The standard supports five different sub-channelization schemes in both uplink and downlink directions. In addition to support for multiple modulation schemes (BPSK, QPSK, 16-QAM, and 64-QAM) and SDMA through AAS, MIMO is also supported. This results in much improved channel capacity and range.

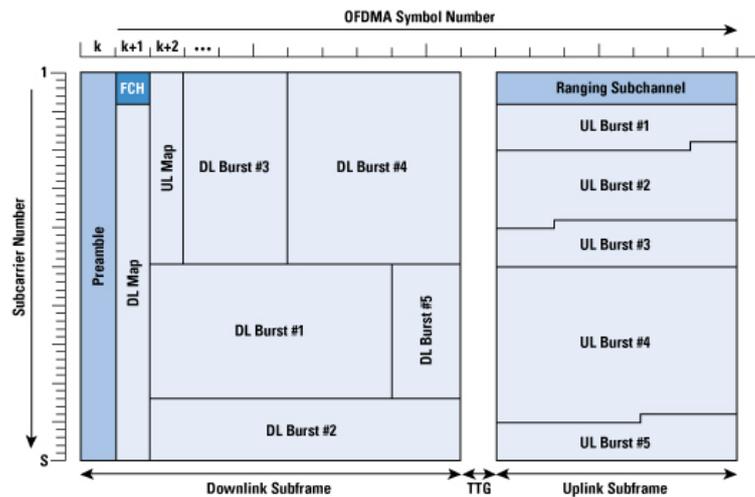


Figure 1.11: An example of WiMAXMAN-OFDMA frame

WirelessHUMAN WirelessHUMAN stands for wireless high speed unlicensed metro area network. It is a point-to-multipoint, none-line-of-sight physical layer specification that is designed for Unlicensed National Information Infrastructure (UNII) and other unlicensed bands [140]. It is similar to OFDM-based schemes, however the specifications only supports TDD for duplexing and therefore no FDD support is provided.

1.4 IEEE 802.16 Media Access Layer (MAC)

The 802.16 MAC layer controls the working of the physical layer. The physical layer is not aware of types of applications and their QoS requirements. The MAC layer provides intelligence to the physical layer by controlling multiple QoS enabled connections over the same physical link. It includes various features that enables it to provide high efficiency and flexibility. One of the features that distinguish it from other wireless MAC layers is the availability of dynamic bandwidth allocation. Dynamic bandwidth allocation reduces both latency and jitter.

WIMAX MAC could accommodate thousands of subscriber stations per channel. It provides support for both bursty and continuous traffic to enable various types of services. The standard was initially developed for backhaul industry and therefore it supports a variety of backhaul requirements including both ATM and packet-based protocols. The main functions of WiMAX MAC layer are summarized below [23]:

- Packing of one or more service data units (SDU) received from higher layer into a single protocol data unit (PDU)
- Selection of appropriate burst profiles and power levels
- Performing error detection and correction through ARQ
- Ensuring QoS according to the priorities of data flows

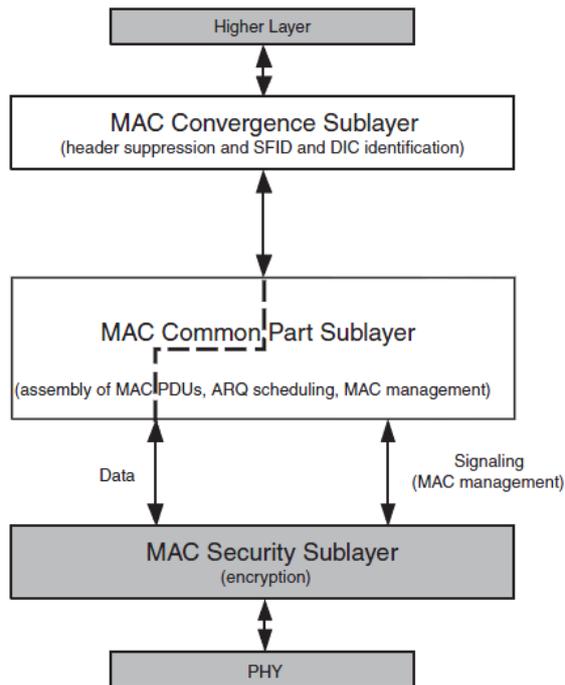


Figure 1.12: The IEEE 802.16 MAC Layer [23]

- Scheduling MAC PDUs over the physical link resources
- Mobility management
- Authentication and encryption
- Power saving and idle mode

1.4.1 MAC Architecture

The MAC layer of WiMAX is divided into three sublayers: the service-specific convergence sublayer, the common-part sublayer, and the security sublayer. The convergence sublayer takes the packet from the Network layer and performs upper layer protocol-dependent tasks such as header compression and address mapping. Thus the details of the higher layers are hidden from the MAC layer. The common-part sublayer is responsible of error handling, packet fragmentation, and packing. The security sublayer ensures secure transmission by providing encryption and authorization services. The architecture of WiMAX layer is shown in Figure 1.12.

Service-Specific Convergence Sublayer

The WiMAX standard includes service-specific convergence sublayers for mapping services to MAC connections. The sublayers support both ATM and packet based networks. The various convergence sublayers supported by the 802.16 standard are shown in Table 1.3. However, WiMAX Forum chose to implement only IP and Ethernet(802.3) sublayers in WiMAX.

Number	Service-Specific Convergence Sublayers
0	ATM
1	Packet IPv4
2	Packet IPv6
3	Packet 802.3 (Ethernet)
4	Packet 802.1/Q VLAN
5	Packet IPv4 over 802.3
6	Packet IPv6 over 802.3
7	Packet IPv4 over 802.1/Q VLAN
8	Packet IPv6 over 802.1/Q VLAN
9	Packet 802.3 with optional VLAN tags and ROHC header compression
10	Packet 802.3 with optional VLAN tags and ERTCP header compression
11	Packet IPv4 with ROHC header compression
12	Packet IPv6 with ROHC header compression
13-31	Reserved

Table 1.3: Service-specific convergence sublayers supported in the IEEE 802.16

The main task of the convergence sublayer is to map service data units (SDU) to proper MAC connections. This is required because higher-layer addresses are not available at the MAC and the physical layers. Therefore it is necessary to map higher layer SDUs to MAC level identifiers. After mapping, a connection identifier acts as a MAC layer identifier, which is assigned by the base station to carry both data and control traffic. The packets are classified in the convergence sublayer and assigned to specific connections based on the types of QoS parameters that the application needs.

In addition to these functions, the convergence sublayers also perform payload header suppression at the sender and reconstruction at the receiving end. Payload header suppression enhances the radio link efficiency by removing repetitive parts from the header of each SDU. Similarly, packet header reconstruction reinserts the removed parts in each SDU before giving it to higher layer protocols. WiMAX supports Robust Header Compression (ROHC) [38], which performs well where the packet loss is high such as in wireless networks.

Common Part Sublayer

The common part sublayer performs tasks that are not dependent at the protocols operating at higher layers. The most important functionality of the MAC layer is performed by this sublayer. The operations done at this layer includes scheduling, error control, bandwidth distribution, modulation, and encoding.

The sublayer is also responsible for transforming higher-layer SDUs into MAC PDUs. Depending on the size of the payload, one or more SDUs can be packed into

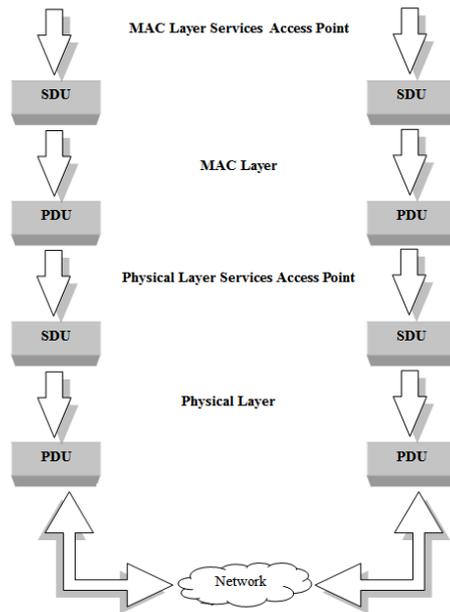


Figure 1.13: Handling of SDUs and PDUs

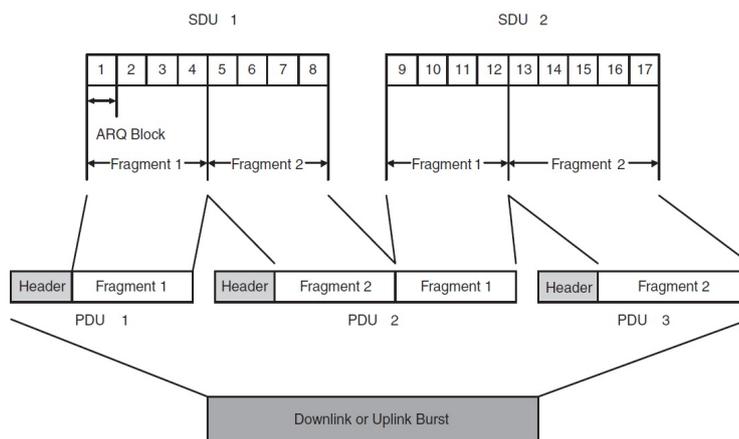


Figure 1.14: Fragmentation and packing of MAC SDUs in WiMAX

a single MAC PDU, or a large SDU can be fragmented to be carried over on several PDUs. This allows more efficient use of physical resources and makes MAC layer operation transparent to the receiving entity. Fragmentation and packing of SDUs in WiMAX is shown in Figure 1.14.

Security Sublayer

The main objective of the security sublayer is to protect service providers from theft of service. It provides authentication services for network access and connection establishment. The sublayer also provides services to exchange secure keys and data encryption to ensure data privacy. The security sublayer secures data at the data link layer. However, it does not protect interruption of the physical layer. Therefore, it is necessary to use mechanisms to secure physical layer and to provide

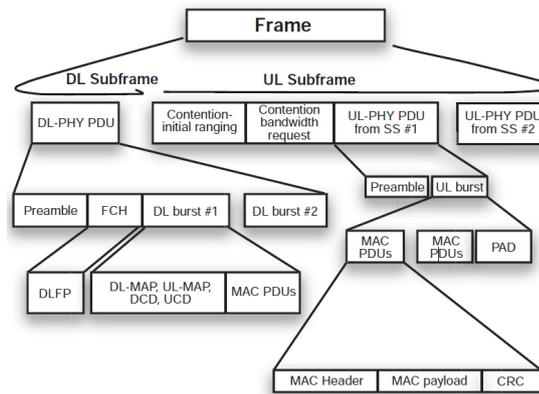


Figure 1.15: Frame structure of IEEE 802.16 MAC Frame [95]

security for higher layers [36].

1.4.2 MAC Frame Structure

A MAC frame is composed of several time-slots. A slot is the minimum physical link resource that could be allocated to a subscriber station. Each slot consists of one or more OFDM symbols. A contiguous series of slots allocated to a particular user is called its data region.

The WiMAX MAC supports both TDD and FDD operations. A TDD frame is divided into an uplink subframe and a downlink subframe. The two subframes are separated by a guard interval. In FDD networks, the frame structure is the same except that both subframes are transmitted simultaneously but over separate carriers. The structure of 802.16 MAC Frame is shown in Figure 1.15.

The downlink subframe starts with a downlink preamble, which contains the information for physical layer procedures. It is followed by a frame control section (FCH) containing the size of UL-MAP and DL-MAP messages, the modulation and the coding schemes. The MAP messages are broadcasted using the most robust profile by the base station after FCH. These messages specify the burst profile for each user. A burst profile is the smallest self-contained unit of data that has an associated modulation and coding scheme. The modulation and coding schemes are chosen adaptively for each burst to each subscriber station. More efficient burst profiles are chosen under favorable link condition, while more robust profiles are chosen under bad link conditions.

DL-MAP is always associated with the current frame and it is at least two FEC blocks long. The DL-MAP messages specify the physical layer level transitions and designate the portion of downlink subframe to individual users. Similarly, UL-MAP specifies bandwidth allocations and burst profiles for the uplink direction. The UL-MAP message provides bandwidth allocations starting no later than the next downlink frame.

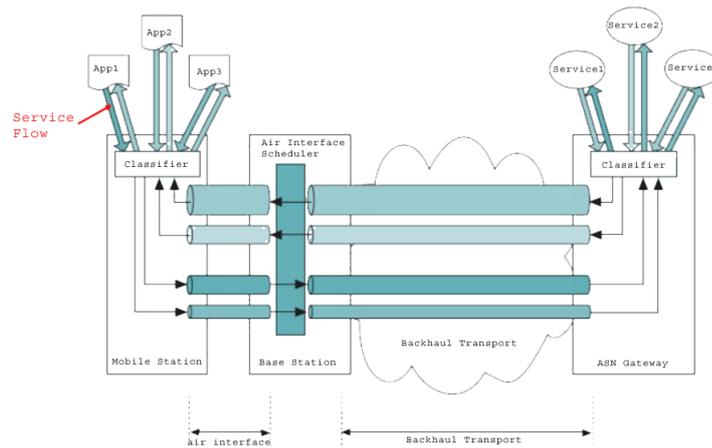


Figure 1.16: Service flows in WiMAX [20]

There are three categories of fields in an uplink sub-frame:

- Initial ranging slot: This slot is used by the subscriber stations to determine the optimum transmission power, and frequency and timing offsets to communicate with the base station.
- Bandwidth request contention slot: This slot is used by subscriber stations to transmit bandwidth requests to base station.
- Data slots: These slots are used to transmit payload.

1.4.3 Service Flows and Connections

WiMAX MAC is connection-oriented. Each connection is identified by a unique 16-bit connection identified (CID). A CID distinguishes multiple uplink data flows that are associated with the same downlink channel. The subscriber stations check the CIDs in the received data and process only those frames that are addressed to them. A service flow is a logical uni-directional flow of packets between a subscriber station and access service network gateway (ASN-GW) with an associated set of QoS parameters identified by a connection identifier, as shown in Figure 1.16.

The primary CID is allocated to the subscriber station during the network entry and initialization stage and it is used to transport all MAC-level signaling messages. When a base station admits a subscriber station to the network during the network entry and initialization stage, it assigns three connections to the subscriber station in each direction. These connections offer different levels of QoS and transport all MAC level signaling and control messages.

- **Basic Connection:** Basic connections are used to transfer short, time-critical control messages.
- **Primary Management Connection:** These connections are used to transmit delay-tolerant control messages such as authentication and connection setup messages.

- **Secondary Management Connections** These connections transmit protocol-based management messages such as Simple Network Management Protocol (SNMP) [43], File Transfer Protocol (FTP) [150], and Dynamic Host Configuration Protocol (DHCP) [62].

In addition to these management connections, subscriber stations can request transport connections to transmit data. These connections are unidirectional and typically assigned to services in pairs i.e. one connection for each direction.

In WiMAX a connection must be established before any communication can occur. When an application intends to exchange data a service flow is established between the subscriber station and the base station. The service flow specifies the required QoS parameters for the data exchange. Once the service flow is permitted, it is associated with a connection to provide a reference against which a bandwidth can be requested.

There are three sets of QoS parameters associated with each service flow. These sets are called *ProvisionedQoSParamSet*, *AdmittedQoSParamSet*, and *ActiveQoSParamSet*. These sets correspond to three possible service flow states, i.e. provisioned, admitted, and active. A service flow can be requested with any of these sets being null. A service flow cannot transfer data packets until it is switched to *active* state. The three sets can have different QoS parameters, however the following relation should always hold:

$$ActiveQoSParamset \subset AdmittedQoSParamset \subset ProvisionedQoSParamset \quad (1.1)$$

Provisioned service flows can be switched to *admitted* or *active* state, when *AdmittedQoSParamset* and *ActiveQoSParamset* become known through subsequent message exchange. IF *ActiveQoSParamset* is specified in the flow-creation request, the service flow can be provisioned and then immediately switched to the *active* state.

When a subscriber station intends to establish a connection, it sends the QoS requirements to the base station. The base station then pre-provision the service by entering the information into the service flow database. When the subscriber station is allowed to join the network, the base station fetches the flow information from the service flow database. The Service flow table, service class table, and the classifier rules table are designed to support auto-installation and auto-configuration.

The MAC reserves some connections for special purposes. A connection is reserved for broadcast transmission in the downlink direction. Another connection is reserved for broadcast contention-based polling. Similarly, a connection is reserved for contention-based initial access. Additional connections are reserved for

multicast contention-based polling.

1.4.4 Service Classes and QoS

The 802.16e standard offers five uplink scheduling classes to fulfill delay and bandwidth requirements of various types of user applications. The scheduling classes are very important for proper functioning of the packet schedulers and consequently the entire QoS framework. A brief description of these classes is presented here. More details of service classes and their associated QoS requirements are provided in Chapter 2.

Unsolicited Grant Services (UGS) UGS is designed to support constant bit rate (CBR) services, such as T1/E1 [77] emulation and Voice over IP (VoIP) without silence suppression. UGS supports realtime applications that generate fixed-size packets on periodic basis.

Extended Realtime Polling Services (ertPS) ertPS is designed to support applications that generate variable-size data packets on periodic basis with a sequence of active and idle intervals, such as VoIP with silence suppression. The service is designed on the efficiency of UGS and rtPS classes.

Realtime Polling Services (rtPS) rtPS is designed to support realtime services that generate variable-size data packets on a periodic basis, such as MPEG [144] video.

Non-Realtime Polling Services (nrtPS) nrtPS is designed to support non-realtime services that require variable-size data grants on regular basis. These applications are delay-tolerant but requires a minimum reserved traffic rate.

Best Effort (BE) Services BE services are typically provided by the Internet today for web surfing. These service do not require QoS support.

1.4.5 MAC PDU and SDU

A service data unit (SDU) is the data unit exchanged between two adjacent protocol layers. They constitute the data units received in downward direction from adjacent higher layer or the data units received in the upward direction from adjacent lower layer.

A MAC protocol data unit (PDU) is the data unit exchanged between the MAC layers of a base station and its subscriber station. It is the basic payload unit handled

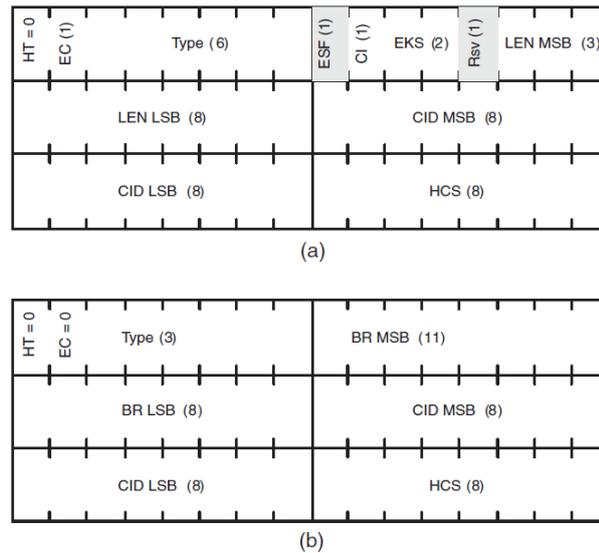


Figure 1.17: WiMAX PDU (a) Generic header (b) Bandwidth request header

by the MAC layer. It consists of three parts: a fixed-size MAC header, a variable-size payload, and an optional cyclic redundancy check (CRC). A MAC PDU can contain application data, management messages, or convergence sublayer data.

There are two types of PDUs in WiMAX i.e. the generic PDU and the bandwidth request PDU. The generic MAC PDU carries data and signalling messages, while the bandwidth request PDU carries uplink bandwidth requests from subscriber stations to the base station. Bandwidth request PDUs do not have any payload. The two PDU types have different header structure, as shown in Figure 1.17. A brief description of various fields of PDU header is given in Table 1.4.

Furthermore, a generic MAC PDU could have one of the following five sub-headers:

Grant management sub-header It is used by a subscriber station to convey bandwidth needs to the base station. The sub-header can be inserted into PDU immediately after the generic header. For an established connection, it is more efficient to use this sub-header rather than bandwidth request PDU. Therefore, bandwidth request PDUs are only used for initial requests.

Fragmentation sub-header The sub-header is used to manage SDU fragments present in the payload. It can be inserted into PDU immediately after the generic header.

Packing sub-header The sub-header indicates the presence of multiple SDUs into a single PDU. The packing sub-header is inserted before each MAC SDU that are contained within the PDU.

Mesh sub-header It is used only in WiMAX mesh networks

Fast-feedback allocation sub-header The sub-header indicates that the PDU contains feedback information about the downlink channel state.

Field	Description	Generic Header	Bandwidth Request Header
HT	Header type	value is 0	value is 1
EC	Encryption control	0 = no encryption, 1 = payload is encrypted	value is always 0
Type	PDU type		
ESF	Extended subheader field	0 = no ES, 1 = ES present	N/A
CI	CRC indicator	0 = No CRC, 1 = CRC Included	N/A
EKS	Encryption key sequence	index of the encryption key and the initialization vector used to encrypt the data	N/A
Rsv	Reserved		
LEN	Length	Size of the MAC PDU including the header	N/A
CID	Connection identifier	CID of the connection	CID of the connection
HCS	Header check sequence	present	present
BR	Bandwidth request	N/A	number of bytes requested for up-link transmission for a particular CID

Table 1.4: MAC PDU header fields [23]

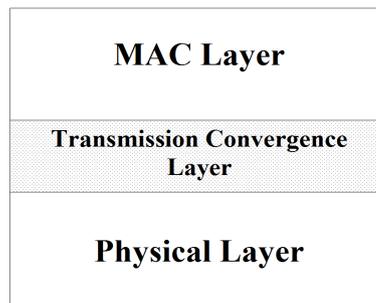


Figure 1.18: Transmission Convergence layer

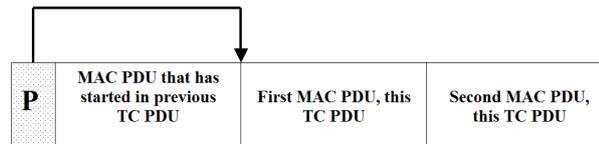


Figure 1.19: Transmission convergence PDU [140]

1.4.6 Transmission Convergence Layer

The transmission convergence layer resides between the physical layer and the MAC layer as shown in Figure 1.18. The layer transforms variable size MAC PDUs into fixed-size FEC blocks. The FEC block obtained at the end of each burst could be possibly shortened by the layer.

The transmission convergence layer resizes PDU to fit in the FEC block currently being used. It has a starting pointer that indicates where the next MAC header starts within the current FEC block, as shown in Figure 1.19. This enables resynchronization to the next PDU in case the previous FEC block had irrecoverable errors.

1.4.7 Working of WiMAX MAC

The WiMAX downlink operates on point-to-multipoint basis as shown in Figure 1.20. A base station, with sectorized antenna that could handle multiple sectors independently, controls all the transmission within its permissible range. Within a particular frequency band and antenna sector, all the subscriber stations receive the same transmission.

In the downlink, the base station is the only transmitter and the transmission is generally broadcast. However, the base station can designate portions of the downlink sub-frame to specific subscriber stations. In such a case, only intended subscriber stations receive the transmission. Otherwise, all the subscriber stations capable of listening to that portion would listen. Before a subscriber station could establish connection with the intended receiver, it must enter the network. The network entry process is shown in Figure 1.21 and explained below.

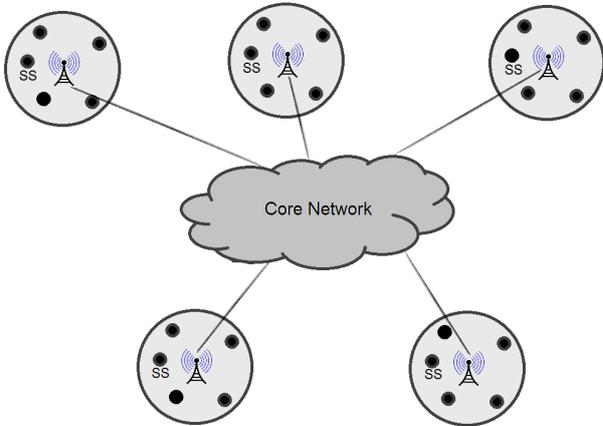


Figure 1.20: A point-to-multipoint WiMAX network

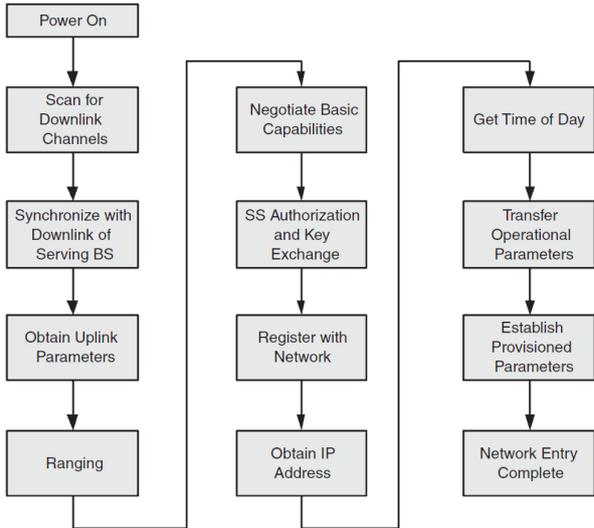


Figure 1.21: Network entry in a point-to-multipoint WiMAX network

Channel Acquisition

The MAC layer provides an initialization procedure that enables automatic configuration of subscriber stations. On startup, a subscriber station scans allowed frequency bands to find a suitable operating channel. Each subscriber station keeps a list of permissible downlink frequencies that are specified by the service provider's network. A subscriber station may be pre-configured by the vendor to use a specific base station or it may automatically select one of the available base stations. Pre-configuration is useful when the subscriber station might listen to another base station.

After detecting a suitable channel, the subscriber station synchronizes with the downlink transmission by detecting the downlink frame preambles. On detecting a preamble, the subscriber station synchronizes with the downlink transmission of the base station. Once the synchronization is done, the subscriber station determines the modulation scheme, FEC used on the channel, and other physical layer parameters by listening to downlink channel descriptor (DCD) and uplink channel descriptor (UCD) messages.

Initial Ranging

After determining the transmission parameters, the subscriber stations extract ranging information from UL-MAP messages. The process is required to determine relative timing and power-level adjustments required for uplink transmission. The initial ranging process is contention-based and therefore the subscriber station must use a backoff algorithm to choose an initial ranging slot to send ranging request. In Mobile WiMAX, a mobile station sends a Code Division Multiple Access (CDMA) [208] ranging code with the power level in the ranging slot. While in Fixed WiMAX, the subscriber station sends a RNG-REQ message in the corresponding ranging slot. In case of an unsuccessful ranging attempt, the subscriber station enters the contention-resolution phase and repeat the initial ranging process with increasingly higher power settings until a ranging response (RNG-RSP) is received. The process is shown in Figure 1.22.

The base station analyzes the power of the received signal and adjusts accordingly the timing advance and the power to the subscriber station. The ranging response contains the timing and power level offsets, and connection identifiers for the basic and primary management connections allocated to the subscriber station. Once the primary uplink connection is established, the subscriber station performs periodic ranging to adjust for timing and power-level fluctuations that may arise due to mobility, fading, and other such phenomena.

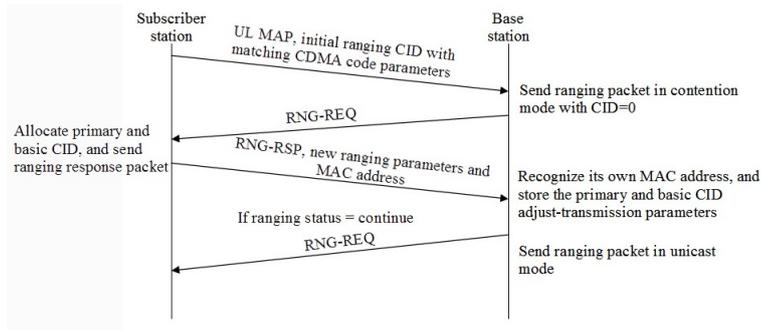


Figure 1.22: Ranging procedure [23]

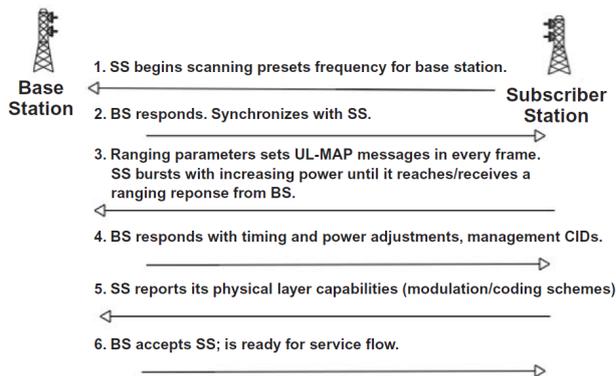


Figure 1.23: Channel acquisition, ranging, and exchange of basic capability set [140]

Subscriber Station Authentication and Registration

After initial ranging, the subscriber station reports to the base station its physical link capabilities and the modulation and coding schemes it supports. The parameters included in the basic capability set of subscriber station and the base station are shown in Table 1.5. In response, the base station provides the physical link parameters to be used for communication. The base station can permit or deny any capability reported by the subscriber station. The processes of channel acquisition, ranging, and registration are illustrated in Figure 1.23.

Each subscriber station is shipped with a pre-installed X.509 [57] digital certificate and a certificate issued for the manufacturer. These certificates are used during the authorization and authentication processes. The base station used the certificates to verify the identity of the subscriber station and its level of authorization. If the subscriber station is admitted to the network, the base station issues an authorization key encrypted with subscriber station’s public key. The authorization key is then used to secure further transactions. After negotiating the basic capabilities and receiving the authorization key, the subscriber station registers with the network and management connections are established with the base station.

Parameters	Description
Transmission gap	The transmission gap between the uplink and downlink sub-frame supported by the subscriber station for TDD and HF-FDD
Maximum transmit power	Maximum transmit power available for BPSK, QPSK, 16 QAM, and 64 QAM modulation
Current transmit power	The transmit power used for the current MAC PDU (containing the SBC-REQ message)
FFT size	The supported FFT sizes (128, 512, 1,024, and 2,048 for OFDMA mode; 256 for OFDM mode)
64 QAM support	Support for 64 QAM by the modulator and the demodulator
FEC support	Which optional FEC modes are supported: CTC, LDPC, and so on
HARQ support	Support for HARQ
STC and MIMO support	The various space/time coding and MIMO modes
AAS private MAP support	Support for various AAS private MAP
Uplink power-control support	Uplink power-control options (open loop, closed loop, and AAS preamble power control)
Subcarrier permutation support	Support for various optional PUSC, FUCSC, AMC, and TUSC modes
Half-duplex/full-duplex FDD support	Support for half-duplex and full-duplex FDD modes in case of FDD implementation

Table 1.5: Basic capability set of base station and subscriber station

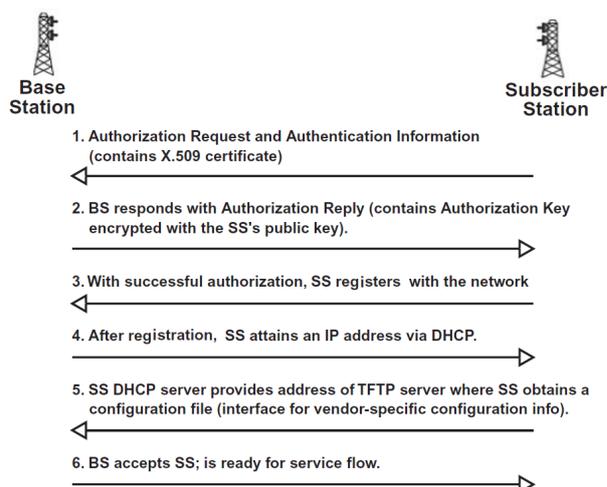


Figure 1.24: Subscriber station authentication and registration [140]

IP Connectivity

Once the registration process is completed, the subscriber station uses Dynamic Host Control Protocol (DHCP) to obtain an IP address. The subscriber station may report the IP protocol versions supported by it to the base station. The base station then chooses one of the supported protocol versions and informs the subscriber station. Then the subscriber station determines the network time using the Internet Time Protocol [148]. The DHCP server supplies the address of the TFTP server which the subscriber station can use to download a configuration file. The configuration files provides manufacturer-specific configuration information. The process is illustrated in Figure 1.24.

Connection Setup

To transmit data, service flows must be established first. A service flow is a one-way stream of packets either in uplink or downlink direction with an associated set of QoS parameters such as bandwidth, latency, and jitter. A service flow can be initiated either by the base station or the subscriber station. Network-initiated service flow creation is a mandatory feature of 802.16e. While, subscriber-initiated flow creation is an optional capability. A subscriber station could initiate service flows only if a dynamically signed connection, such as switched virtual connection (SVC), is present. A service flow is established via a three-way handshaking protocol. Firstly, a service flow establishment request is sent, which is responded by the receiving entity. The initiator then send an acknowledgement response.

For subscriber station initiated service flows, the subscriber station sends a Dynamic Service Addition request (DSA-REQ) message. The DSA-REQ message specifies the QoS parameters required by the service flow. If the base station could provide the desired level of QoS, it creates a service flow identifier (SFID) and send

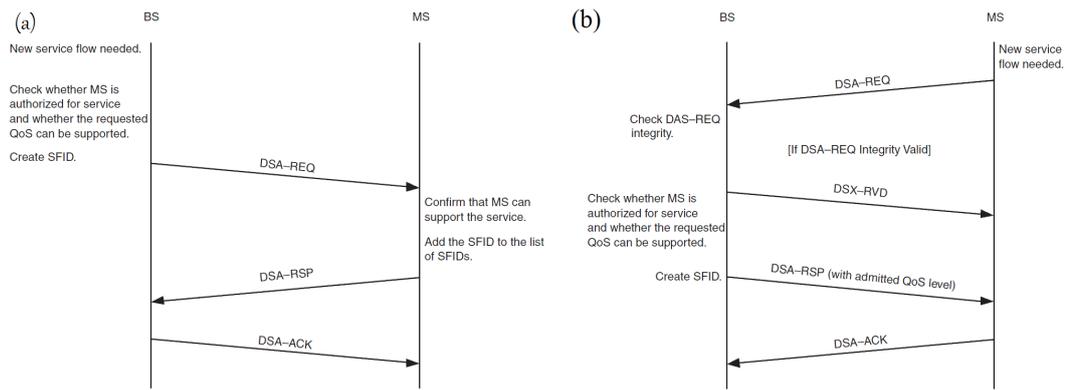


Figure 1.25: (a) Base station initiated service flow (b) Subscriber station initiated service flow

it to the subscriber station through a DSA-RSP message. The subscriber station then responds with the acknowledgment message (DSA-ACK).

For base station initiated flows, the base station first checks the authorization level of the subscriber station and whether or not the required QoS parameters could be supported. If the subscriber station is authorized for the requested service, the base station creates an SFID and sends a DSA-REQ message to the subscriber station. If it is acceptable for the subscriber station, it responds by a DSA-RSP message. The base station then responds with DSA-ACK message.

A service flow must be mapped to a connection before actual data transfer could take place. A connection is activated in two phases. The connection is first entered into admitted state and then to activated state. The resources assigned to an admitted connection may not be actually provided until the connection is activated. WiMAX also supports dynamic changes in a connection through renegotiation of service flow parameters. These changes follow the three-way handshaking protocol as described above.

Bandwidth Allocation

The WiMAX bandwidth request-grant mechanism is scalable, efficient, and self-correcting. The WiMAX MAC can efficiently support multiple connections with different QoS requirements per subscriber station.

Downlink In the downlink, the base station makes bandwidth allocation decision without any involvement of the subscriber stations. As data arrive for a particular connection, the scheduling mechanism at the base station allocates downlink bandwidth to the connection according to its QoS requirements. The base station then conveys the bandwidth allocation to subscriber stations through the DL-MAP message.

Uplink In the uplink, a subscriber station requests uplink bandwidth on per connection basis. Due to dynamic nature of wireless medium, all bandwidth requests are in terms of bytes rather than in terms of physical layer resources. Bandwidth requests can be either incremental or aggregate. When an incremental bandwidth request is received for a particular connection, the base station adds the requested bandwidth to its perception of the bandwidth needs for that connection. While in case of an aggregate bandwidth request, the base station replaces its perception of bandwidth needs by the requested bandwidth. Although a subscriber station requests bandwidth on per connection basis, the base station grants bandwidth to a subscriber station as an aggregate of grants for all connections that belongs to the subscriber station.

The standard supports a variety of uplink bandwidth request/grant mechanisms such as polling, unsolicited grant, contention-based mechanism, and piggybacking. When a subscriber station hosts a connection that is enabled to use polling mechanism, the base station allocates physical slots in MAC frame to allow the subscriber station to make a bandwidth request. Polling can be either unicast or multicast. In unicast polling, each subscriber station is polled individually. While in multicast polling, a group of subscriber stations is polled simultaneously. The base station indicates the allocation of uplink bandwidth to the subscriber station through the UL-MAP messages.

If the available bandwidth is not sufficient to poll each subscriber station individually, the base station can use multicast or broadcast polling to poll multiple subscriber stations simultaneously. The subscriber stations in the polled group with data to send then responds with a bandwidth request. To avoid transmission collision, exponential backoff algorithm [132] is used as contention-resolution procedure during a multicast or broadcast poll. For contention-resolution, the subscriber station first choose a random number CR between 0 and BACKOFF-WINDOW. The subscriber station then waits for CR number of transmission opportunities before sending its bandwidth request. In case of collision with another subscriber station, the subscriber station increases its waiting time by a factor of 2 and repeats the process. If it is unable to get bandwidth after a maximum number of tries, the MAC PDU is dropped.

Unsolicited grants are used for UGS connections only. Since these connections require a fixed amount of bandwidth on periodic basis, therefore the subscriber station does not need to provide its bandwidth requirements to the base station. This avoids the overhead associated with the polling mechanism.

In contention-based mechanism, there are no dedicated resources to make bandwidth requests. Instead subscriber stations uses a CDMA based mechanism to make bandwidth requests, so the base station can provide enough opportunities to make bandwidth request before sending the bandwidth request header.

The piggyback bandwidth request mechanism enables a subscriber station to make incremental bandwidth requests using the piggyback field in the grant management sub-header (GMSH). This avoids the need to separately allocate a bandwidth request header to each service flow.

When the base station receives a bandwidth request, the base station uplink scheduler allocates some bandwidth to the connection. However, during this time the state of uplink queues may have changed at the subscriber station due to arrival of new packets. Therefore the uplink scheduler allocates bandwidth on per subscriber station basis and leave the bandwidth distribution among service flows to subscriber station scheduler. This allows subscriber station scheduler to make decision on most recent queue states. Also this greatly reduces overhead as the base station needs not to send separate UL-MAP for each service flow.

Error Control

The error detection and correction mechanism in WiMAX is based on Automatic Repeat Request (ARQ). The ARQ mechanism is used to retransmit MAC SDU that have been lost or corrupted during transmission. The mechanism employs a sliding window approach that allows the sender to send up to a specific number of frames without receiving an acknowledgement. If an SDU is successfully received, then the receiver sends an acknowledgement to the sender, otherwise it sends a negative acknowledgement. In case of negative acknowledgement, the sender retransmits the corresponding SDUs and moves the sliding window forward.

In ARQ-enabled connections, the MAC SDU is first partitioned into fixed-length portions, called ARQ blocks. The length of the ARQ blocks for each connection is specified by a parameter called ARQBLOCKSIZE. Each ARQ block is identified by a block sequence number (BSN). The ARQ blocks are transmitted over the physical link, which are then received at the receiver and assembled into a MAC PDU.

Upon successful reception of ARQ blocks, the receiver sends an acknowledgement (ACK) indicating successful reception of one or more ARQ blocks. The acknowledgement can be sent either as a stand-alone message or through piggybacking. WiMAX supports two forms of feedbacks: selective acknowledgements and cumulative acknowledgements. In a selective acknowledgment, the receiver indicates the BSN of the ARQ block successfully received. In cumulative acknowledgements, the receiver sends back a BSN indicating that all ARQ blocks with the sequence number less than or equal to the BSN have been properly received.

1.4.8 Radio Link Control

WiMAX physical layer provides an advanced radio link control that enables the physical layer to adaptively switch from one burst profile to another. In addition, the

radio link control provides the functionality of power control and ranging. The radio link control sends periodic broadcasts of the burst profiles that are chosen for the uplink and downlink directions. Both atmospheric conditions and subscriber station capabilities are observed in choosing the most appropriate burst profile. Burst profiles for the downlink are tagged with Downlink Interval Usage Code (DIUC), while for the uplink are tagged with Uplink Interval Usage Code (UIUC).

During initial ranging, a subscriber station sends the specifications of the burst profile it intended to use in the downlink. The decision is based on the quality of downlink signals received. The base station may accept or deny the specified burst profile through RNG-RSP messages. Similarly, the base station continuously monitors the quality of the uplink signal it receives from each subscriber station. The base station could direct a subscriber station to use a particular burst profile by including an appropriate UIUC in the UL-MAP message when granting bandwidth. In this way, the subscriber station would always receive both the UIUC and the grant or none. Therefore the subscriber station does not need to send an acknowledgement. Furthermore, this reduces the possibility of uplink burst profile mismatch between the base station and the subscriber station.

The radio link control continues to monitor and modify the burst profiles according to channel conditions to achieve a best mix of efficiency and reliability. Negative changes in the environment can cause a subscriber station to use a more robust but less efficient profile. On the other hand, under good conditions a subscriber station can temporarily request a more efficient burst profile.

In the downlink, if a subscriber station needs to change the burst profile, it can request the change in two ways. The request method to be used depends upon whether the subscriber station uses grant per connection or grant per subscriber station mode. For grant per connection subscriber stations, the base station regularly allocates a station maintenance interval to the subscriber station. In the maintenance interval, the subscriber station could transmit a RNG-REQ message to request change in the downlink burst profile. For grant per subscriber case, the subscriber station sends a downlink burst profile change request (DBPC-REQ) message to the base station. The base station then responds with a DBPC-RSP message either accepting or rejecting the requested change.

1.4.9 Uplink Connection Classification

Each connection in the uplink direction is mapped to a corresponding scheduling service. The scheduling services have associated set of rules for allocating the uplink bandwidth. The scheduling services also specifies the details of request-grant mechanism between the subscriber station and the base station. The scheduling service to be mapped to a particular uplink connection is decided at the time of connec-

tion setup. The scheduling services used in WiMAX are based on Data-Over-Cable Service Interface Specifications (DOCSIS) [10] standard.

1.5 WiMAX Network Architecture

The design of an end-to-end system is not part of the IEEE 802.16 standard specifications. However, it is necessary to have an end-to-end architecture to address important issues such as IP connectivity, session management, end-to-end QoS support, and mobility management. Therefore, the WiMAX Network Working Group (NWG) developed and standardized these end-to-end networking issues. The details of the WiMAX network architecture are presented in this section.

1.5.1 Network Reference Model

The WiMAX network reference model is shown in Figure 1.26. It specifies the functional modules and the interactions among them. The model consists of three logical parts:

Mobile stations These are the devices used by the end-users to connect to the network

Access Service Network (ASN) An ASN consists of several base stations and gateways. WiMAX network model supports multiple profiles, each specifying different functional decomposition. In profile *A* and *C*, functionality is divided among base stations and gateways. While in profile *B* the functionality of base station and ASN is combined in a single functional entity. The comparison of various profiles is presented in Table 1.6 [23].

Connectivity Service Network (CSN) provides IP connectivity and the core IP network functionality. A CSN is managed by a network service provider (NSP). A mobile station is served by the CSN that is controlled by the visiting NSP, while the home NSP is where the mobile user belongs to.

1.5.2 Functions of Various Components of WiMAX Network

Access Service Network

The functions provided by ASN can be summarized as follows:

- Layer 2 connectivity with the mobile station
- Network discovery and selection
- Radio resource management and control according to QoS policy
- IP connectivity between the mobile station and CSN

Functional Category	Function	ASN Entity Name		
		Profile A	Profile B	Profile C
Security	Authenticator	ASN-GW	ASN	ASN-GW
	Authentication relay	BS	ASN	BS
	Key distributor	ASN-GW	ASN	ASN-GW
	Key receiver	BS	ASN	BS
	Data path function	ASN-GW	ASN	ASN-GW
Mobility		and BS		and BS
	Handover control	ASN-GW	ASN-GW	BS
	Context server and client	ASN-GW	ASN	ASN-GW
Radio resource management		and BS		and BS
	MIP foreign agent	ASN-GW	ASN	ASN-GW
	Radio resource controller	ASN-GW	ASN	BS
	Radio resource agent	BS	ASN	BS
Paging	Paging agent	BS	ASN	BS
	Paging controller	ASN-GW	ASN	BS
QoS	Service flow authorization	ASN-GW	ASN	ASN-GW
	Service flow manager	BS	ASN	BS

Table 1.6: Functional Decomposition of the ASN in Release 1 Profiles

- paging
- caching of subscriber profiles
- AAA client/proxy
- delivering RADIUS/DIAMETER messages to selected CSN AAA
- performs routing (IPv4 and IPv6) to selected CSNs.

Connectivity Service Network (CSN)

The CSN performs the following functions [23]:

- IP address allocation to subscriber stations for user sessions
- AAA proxy or server for user, device and services authentication, authorization, and accounting (AAA)
- Policy and QoS management based on the service level contract with the user
- Subscriber billing and interoperator settlement
- Inter-CSN tunneling to support roaming between NSPs
- Inter-ASN mobility management and mobile IP home agent functionality
- Connectivity infrastructure and policy control for such services as Internet access, access to other IP networks, ASPs, location-based services, peer-to-peer, VPN, IP multimedia services, law enforcement, and messaging

In this section, we provided an overview of the WiMAX network architecture. It is an end-to-end system that provides IP connectivity, session management, end-to-end QoS support and mobility management. The architecture consists of three main parts: mobile stations, access service network, and connectivity service network. Mobile stations are the devices used by the end-users. An access service network consists of one or more base stations and gateways. It enables mobile stations to communicate with the network. A connectivity service network provides IP connectivity and the core network functionality.

1.6 Conclusion

In this chapter, we focused on the specifications of the IEEE 802.16 standard. We discussed in detail the design of the physical and MAC layers of the standard. The standard was initially designed for fixed subscriber stations. Later on, the IEEE 802.16 Networking Group extended the scope of the standard to provide support for the mobile devices. It utilizes several robust and proven technologies at the physical layer to enable high-rate data transmission over large distances. These technologies include adaptive modulation, OFDM, OFDMA, adaptive antenna systems, multiple-input multiple-out and several others. However, the physical layer cannot determine the types of traffic and the associated QoS requirements. It is the MAC layer that provides intelligence to the physical layer by controlling multiple QoS-enabled connections over a physical link. The 802.16 MAC layer is divided into convergence sublayer and common-part sublayer. The convergence sublayer performs high layer protocol dependent tasks, while the common-part sublayer performs protocol independent tasks such as scheduling, error control, modulation, and encoding. The QoS mechanisms are implemented at the MAC layer and therefore it plays the central role in providing QoS to user applications. The understanding of the MAC layer is important for understanding the 802.16 QoS architecture, which is discussed in the next chapter.



2

QoS in 802.16 Networks

A well-designed QoS framework is important for efficient working of 802.16 networks. The two most important components of a connection-based QoS architecture are connection admission control (CAC) and packet schedulers. The IEEE 802.16 standard provides a rich QoS architecture, however it does not provide working details of these components. The aim of this chapter is to provide an overview of the IEEE 802.16 QoS architecture and the algorithms that have been proposed for CAC and packet scheduling.

The chapter is organized as follows. Section 2.1 gives a description of various perspectives of QoS in data networks, and the main components that are required to furnish QoS. Then, Section 2.2 provides a detailed overview of the 802.16 QoS architecture. Several researchers have proposed algorithms for 802.16 CAC and packet schedulers. Extensive classifications of the algorithms published in literature for CAC and packet scheduling are provided in Sections 2.3 and 2.4, respectively. Section 2.5 concludes the chapter.

2.1 Quality of Service

The term Quality of Service refers to the set of standards and techniques that ensures high quality performance of communication networks. In traditional networks, all types of traffic were treated equally. Therefore, an application could not get guarantees on throughput, delay, reliability and other transmission parameters. Consequently, a single bandwidth-intensive application could severely degrade the service level of the entire network. Modern networks support traditional data, high-quality video, realtime voice and other realtime applications. These applications require certain guarantees on transmission characteristics to ensure satisfactory user experience.

The problem of providing QoS is to determine efficient allocation of physical resources among users so that the performance characteristics such as delay, jitter, and throughput could be guaranteed. The main characteristics that are managed by a QoS framework are shown in Table 2.1. The QoS mechanisms have to ensure prioritized delivery of packets according to their QoS requirements, while maintaining a high utilization of network resources. A QoS-enabled network differentiates between different classes of traffic and provides preferential processing to critical applications. There are two broad perspectives of QoS [174] i.e. user-centric and network-centric.

- **User-Centric:** The European Telecommunication Standard Institution defines QoS as “the collective effect of service performances which determine the degree of satisfaction of a user of the service” [68].
- **Network-Centric:** “QoS refers to the mechanisms that give network man-

Transmission Parameter	Definition
Throughput	data transmission rate expressed in bit/sec
Latency	delay in transmission of data from source to destination
Jitter	the variation in latency
Packet loss	the percentage of packets lost during transmission. Smaller the packet loss, greater is the network reliability

Table 2.1: Important transmission parameters

agers the ability to control the mix of bandwidth, delay, variances in delay (jitter), and packet loss in the network in order to deliver a network service" [61].

QoS differentiation can be done at either per-user or per-connection basis. On this basis, QoS mechanisms can be grouped under the following two categories [174]:

Admission Control These mechanisms determine how and when a connection can be admitted to the network so that it could have access to network resources.

Traffic Control These mechanisms determine how packet classification, scheduling, flow rate control is done for traffic that enter the network. Typically, these decisions are done on per-packet basis.

When an application intends to send data, it must specify the required QoS parameters. An application can specify a subset of the following QoS parameters:

- **Minimum Reserved Traffic Rate (MRTR):** The minimum data transfer rate guaranteed to a service flow.
- **Maximum Sustained Traffic Rate (MSTR):** The upper limit of data transfer rate for a service flow
- **Maximum Tolerated Latency:** The maximum permitted packet delay from source to destination
- **Tolerated Jitter:** The maximum permitted variation of latency
- **Unsolicited Grant Interval:** The time interval between successive bandwidth grant opportunities for a connection over downlink
- **Unsolicited Polling Interval:** The maximum interval between successive polling bandwidth opportunities for a connection in the uplink direction.

QoS mechanisms enable network administrators to ensure the required level of service without over-provisioning the networks. A well-designed QoS framework allows the administrators to manage available resources to suit a particular requirement model, and to assure QoS especially for critical services. This results in high user satisfaction and enhanced efficiency without additional expenditures in network resources.

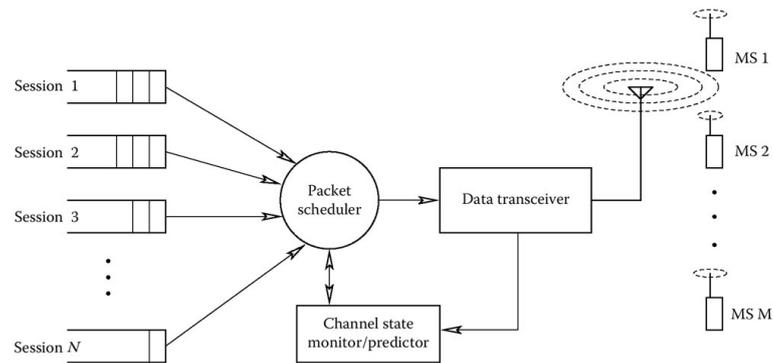


Figure 2.1: Scheduler in a wireless network [34]

2.1.1 Components of QoS

Connection admission control and packet scheduler are two main components of any QoS framework. These components must be integrated with the information of the physical layer for efficient operation. The main functions of these components are discussed below.

Connection Admission Control (CAC)

CAC is the first important component in a QoS framework. It accepts or denies connection requests such that the QoS could be guaranteed for the new connection, while the QoS of existing connections is not sacrificed. The decision is made according to the QoS requirements of the new connection and the available network resources.

To admit a new connection, CAC gets the information about the subscriber station and the requested QoS parameters. The CAC then analyzes the available network resources and admits the new connection only if QoS could be guaranteed to both new and existing connections. The reference of the accepted connection is then forwarded to the *packet scheduler*.

Scheduler

The scheduler is the most important QoS component at the MAC layer. A scheduler consists of two main parts, i.e. queues and scheduling algorithm. A schematic representation of a packet scheduler is shown in Figure 2.1. Queues could contain control information, user data, or bandwidth requests. By using multiple queues and an appropriate scheduling algorithm, preferential treatment can be provided to critical applications. A scheduling decision is based on various QoS factors and network state such as queue size, elapsed delay, minimum traffic rate, and radio link conditions. In general scheduling priority of service flows can be determined as follow [21]:

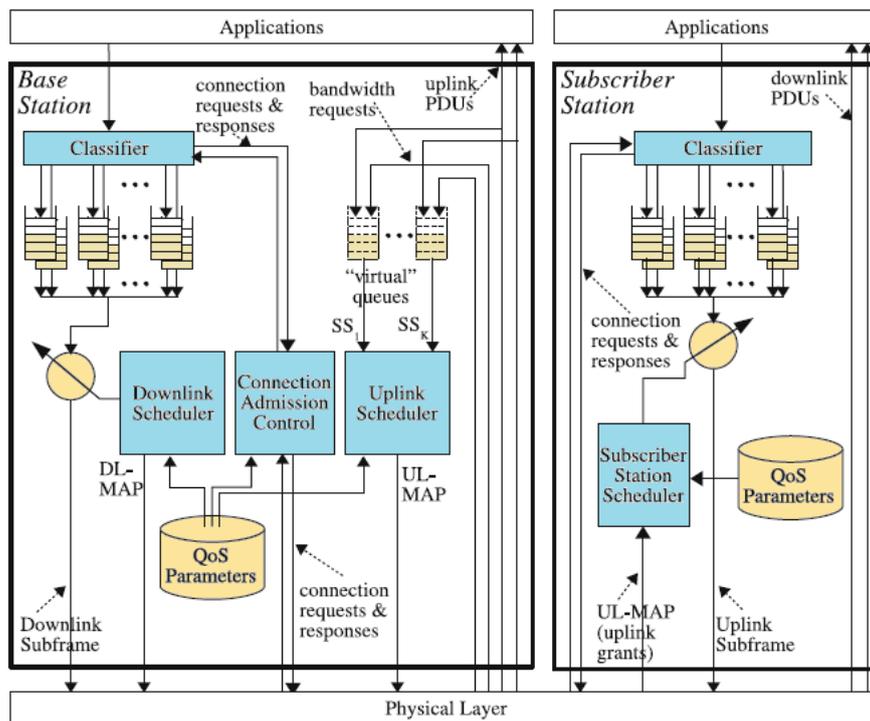


Figure 2.2: WiMAX QoS Architecture

- service flows with shorter maximum delay
- service flows with higher reserved rate
- service flows with large queues
- subscriber stations with better radio channel conditions

The scheduling algorithms decide the order of transmission of packets stored in the queues according to the QoS requirements of various connections. A scheduler takes into account the priority of different classes of services and the available resources to ensure satisfactory provisioning of services to critical applications. The Application Working Group (AWG) of WiMAX Forum considers scheduling critical for achieving optimal performance in WiMAX [191].

2.2 QoS Architecture of WiMAX

In WiMAX, the QoS mechanisms are implemented at the MAC layer. This allows the QoS mechanisms to directly interact with radio resource management to better manage physical layer operations. The WiMAX QoS architecture provides differentiated QoS to support both realtime and non-realtime applications. It supports simultaneous transmission of data, voice, and video by employing suitable technologies in the MAC and physical layers. Video and voice require a bound on latency but tolerate some packet loss, while data applications cannot tolerate packet loss but latency is not an issue. The applications can take advantage of its polling

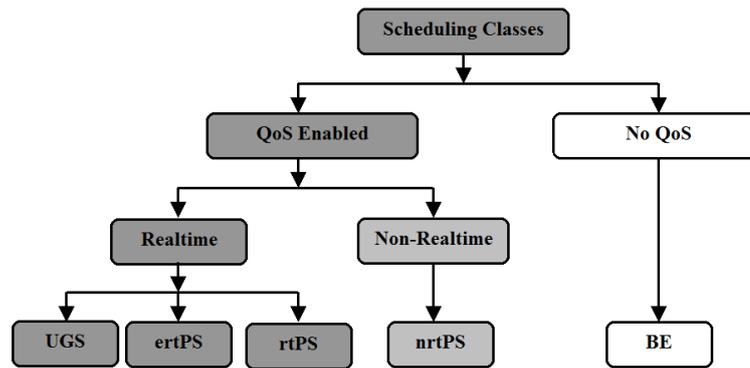


Figure 2.3: Categorization of IEEE 802.16 Scheduling Classes

mechanism and adaptive modulation in the physical layer.

Within a cell, the base station is responsible for providing QoS to all subscriber stations. To ensure continuous QoS, a CAC module at the base station admits or denies a new connection request. The bandwidth required for the uplink and the downlink applications is dynamically scheduled and granted by the base station. The complex task of scheduling is performed by three schedulers. Two of the three schedulers are implemented at the base station, while one scheduler at the subscriber station. Packet classifiers are present at both the base station and subscriber stations to classify and store packets into separate queues according to the QoS demands of the applications. The QoS architecture provided by WiMAX is shown in Figure 2.2. The details of CAC and scheduling in WiMAX are provided in Section 2.2.3 and 2.2.5, respectively.

2.2.1 Service Classes and Application Types

The WiMAX MAC layer provides various scheduling classes and corresponding data delivery services to ensure efficient transmission of MAC PDUs. A classification of the scheduling classes is shown in Figure 2.3. A scheduling service specifies the procedures that a network could use to allocate uplink and downlink transmission opportunities for the PDUs. The scheduling service classes are only for uplink flows, while the data delivery service classes are for both uplink and downlink flows. The 802.16d-2004 specifications only provide the details of scheduling classes for uplink flows. The 802.16e-2005 standard introduces scheduling for downlink flows as well, by proposing data delivery classes for the downlink flows.

The scheduling classes and the corresponding data delivery services supported by the IEEE 802.16e standard are shown in Table 2.2. When an application intends to initiate a communication, it requests one of the scheduling classes for the new service flow. When the service flow gets associated with scheduling class, it is assigned a pre-defined set of QoS parameters. The values of these parameters are later assigned using DSA and DSC commands. WiMAX Forum classifies applications

Scheduling service	Data delivery service	QoS parameters
Unsolicited grant service (UGS)	Unsolicited grant service	Maximum sustained traffic rate, latency tolerance, jitter tolerance
Extended realtime polling service (ertPS)	Extended realtime variable-rate service (ERT-VR)	Maximum sustained traffic rate, minimum reserved traffic rate, latency tolerance, jitter tolerance
Realtime polling service (rtPS)	Realtime variable-rate service (RT-VR)	Maximum sustained traffic rate, minimum reserved traffic rate, latency tolerance
Non-realtime polling service (nrtPS)	Non-realtime variable rate service (NRT-VR)	Maximum sustained traffic rate, minimum reserved traffic rate
Best effort service (BE)	Best effort service	Maximum sustained rate

Table 2.2: The scheduling and data delivery service classes provided by the IEEE 802.16e standard

into five categories as shown in Table 2.3. Each application requires a specific level of QoS to assure a good quality of user experience. The traffic models for these application types are provided in [219].

Unsolicited Grant Service (UGS)

The UGS class is designed to provide fixed bandwidth allocations to applications on periodic basis. This class is suited for constant-bit rate realtime applications that generate fixed amount of data at periodic intervals. After the connection setup, the subscriber station is not required to send any other request. Therefore, the overhead and latency associated with bandwidth request mechanism are eliminated. The QoS parameters associated with UGS service flows are MSTR¹, MRTR², latency, and jitter. MSTR is equal to MRTR for UGS flows. The applications that use UGS class include T1/E1 emulation over wireless link, VoIP without silence suppression, etc.

Extended Realtime Polling Service (ertPS)

This service class was introduced in the IEEE 802.16e-2005 specifications. It is designed to support realtime applications that generate variable-size data packets on periodic basis with active and idle periods. When an ertPS connection is in the active state, the base station allocates unsolicited unicast grants similar to UGS. During the idle period no bandwidth is allocated to the connection. This reduces complexity associated with some application types such as VoIP with silence

1. Maximum sustained traffic rate
2. Minimum reserved traffic rate

Class	Application	Bandwidth Guidelines	Latency Guidelines	Jitter Guidelines	QoS Classes
1	Multiplayer Interactive Gaming	Low	Low	N/A	rtPS and UGS
2	VoIP and Video Conferencing	Low	Low	Low	UGS and ertPS
3	Streaming Media	Low to high	N/A	Low	rtPS
4	Web Browsing and Instant Messaging	Moderate	N/A	N/A	nrtPS and BE
5	Media Content Downloads	High	N/A	N/A	nrtPS and BE

Table 2.3: Application Classes Supported in WiMAX [219]

suppression. The base station polls the subscriber station during the idle period to determine if the active period has been started. The associated QoS parameters are same as those for UGS class.

Realtime Polling Service (rtPS)

rtPS supports realtime applications that generate data at variable rate on regular basis, such as Movie Picture Expert Group (MPEG) videos. Due to the variable data generation rate and tight delay bounds, the base station periodically provides dedicated bandwidth request opportunities to rtPS flows. The associated QoS parameters are similar to that of ertPS class, but MRTR and MSTR have different values.

Non-Realtime Polling Service (nrtPS)

The nrtPS class is specified for delay tolerant applications that require guarantee on the minimum data transfer rate. These applications generate variable-size packets on regular basis with no delay requirement. The only parameters specified by the nrtPS service flows are MRTR and MSTR. Usually, these service flows do not tolerate packet loss. Example of applications that could take advantage of the nrtPS class are FTP and online backup systems.

Best-Effort (BE) Service

The BE service class is designed to support data applications that do not require service guarantees on throughput and delay. The network resources are allocated to BE flows only if other service classes do not need these resources. Most of the data

Types	Overhead	QoS Classes
Implicit Mechanisms		
Unsolicited request	N/A	UGS and ertPS
Poll-me bit	N/A	UGS
Bandwidth stealing	6 bytes MAC header	nrtPS and BE
Piggybacking	2 bytes sub-header	ertPS, rtPS, nrtPS and BE
Contention region	variable	ertPS, nrtPS and BE
Codeword over CQI Channel	N/A	ertPS
CDMA code-based	N/A	ertPS
Explicit Mechanisms		
Unicast polling	6 bytes per user	ertPS, rtPS, nrtPS and BE
Multicast polling	6 bytes per multicast	ertPS, nrtPS and BE
Group polling	6 bytes per group	ertPS, rtPS, nrtPS and BE
Broadcast polling	variable	ertPS, nrtPS and BE

Table 2.4: Bandwidth request mechanisms in WiMAX

applications, such as Hyper Text Transfer Protocol (HTTP) and Telnet, could use BE service.

2.2.2 Bandwidth Request Mechanisms

In a point-to-multipoint WiMAX network, a base station manages both the uplink and downlink resources. To transmit data in the uplink direction, a subscriber station needs to send bandwidth request to the base station. There are various ways in which a subscriber station can request bandwidth. A comparison of these mechanisms is provided in Table 2.4. As indicated in the table, the various bandwidth request mechanisms are unicast polling, multicast polling, broadcast polling, poll-me bit, unsolicited request, piggybacking, bandwidth stealing, CDMA code based request, and code word over channel quality indicator channel (CQICH). However, the optimal way to request the bandwidth is still an open area of research [191] [86] [136] [119]. These mechanisms can be grouped into following four categories:

Unsolicited Request

In this mechanism, the base station periodically assigns fixed amount of bandwidth to the subscriber station. Therefore, the subscriber station does not require to send bandwidth requests. In this way, the minimum latency can be guaranteed to the connection without additional overhead. However, this mechanism could result in wastage of resources in case of bandwidth allocation to a connection that has no data to send [55]. Unsolicited request is utilized by UGS and ertPS classes.

Polling

Polling-based mechanisms provide a dedicated bandwidth-request opportunity to a given subscriber station. These mechanisms guarantee that applications receive service on deterministic basis and avoid the possibility of transmission collisions. Usually, the base station uses polling-based mechanisms for high priority connections. Dedicated bandwidth request opportunities can be assigned to a group of subscriber stations (multicast polling) or to all subscriber stations belonging to a subnet (broadcast polling). Except UGS, all service classes can use polling.

Piggybacking

In this mechanism, a subscriber station sends bandwidth request in the time-slot allocated for data transmission. In this way, the overhead is reduced by avoiding the transmission of a complete bandwidth request message. Consequently, the waiting time is decreased. Except UGS, all service classes can use piggybacking. It is the preferred bandwidth request mechanism for established connections [151].

Contention

In contention-based mechanisms, the base station allocates time-slots in which any subscriber station can send its bandwidth requirements. The subscriber station uses CDMA code to send the bandwidth request. The subscriber station can choose the CDMA code and the transmission time randomly to send the request [114]. In this scenario, there is a possibility that two or more subscriber stations choose same CDMA code and transmission time, which results in collision. If collision occurs, the subscriber stations must use a backoff algorithm to reduce the probability of more collisions. The IEEE 802.16 standard uses truncated binary exponential back-off scheme [136] [52]. The ertPS, nrtPS, and BE classes can use the contention-based bandwidth request mechanism.

2.2.3 Connection Admission Control

In WiMAX, the subscriber station and the CAC module at the base station exchange *ranging* messages to determine channel conditions. To establish a new connection, *register* messages are used by the subscriber to inform the base station about the QoS requirements of the new connection. The details of *ranging* and *registration* processes have been provided in Section 1.4.7. Figure 2.4 shows the ranging and register messaging in WiMAX CAC. The algorithm to decide admissibility of new connections is not specified in the IEEE 802.16 specifications and the decision is left out for equipment manufacturers. A well-designed CAC is important for efficient working of the packet schedulers.

Scheduling Service	Bandwidth Request Opportunities	Bandwidth Request
UGS	The base station uplink scheduler offers fixed size uplink bandwidth grants on periodic basis	no explicit request
ertPS	The base station uplink scheduler offers periodic bandwidth request opportunities	Subscriber station request bandwidth through offered opportunities
rtPS	The base station uplink scheduler provides periodic bandwidth request opportunities	a subscriber station can request bandwidth in the offered opportunity, or it can use piggybacking to demand bandwidth
nrtPS	The base station uplink scheduler provides regular bandwidth request opportunities	A subscriber station can use the offered opportunity, piggybacking, or contention-based request
BE	none	piggybacking or contention based

Table 2.5: Bandwidth request mechanisms for various service classes

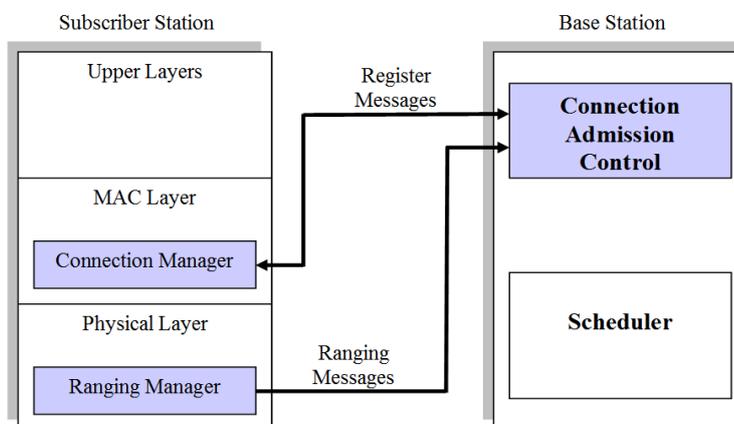


Figure 2.4: QoS Components

2.2.4 Packet Classification

The classification of packets into different categories is necessary before a scheduling mechanism could be applied to the traffic. The classification is done at the convergence sublayer. Higher-layer protocol properties such as IP address and application ports are used to map packets to specific connection identifiers (CID). The value of CID is subsequently used to classify MAC frames into appropriate transmission queues.

Packet mapping to service flows is done at the access service network gateway for downlink flows, and at the subscriber station for uplink flows. The classification and mapping is done according to packet matching rules based on 5-tuples i.e. source IP address, destination IP address, source port address, destination port address, and differentiated services codepoint (DSCP) [95].

2.2.5 Scheduling

In data networks, such as WiMAX, where the objective is to fulfill the QoS requirements of various types of applications, a major challenge is the optimal allocation and utilization of network resources. Resource utilization is considered optimal when there is no over-allocation or under-allocation. These problems could severely effect the network performance.

Beside bandwidth distribution, another important factor is latency. Some applications, such as email and websurfing, can tolerate some delay. While realtime applications have strict deadlines. Another important parameter to control is packet loss. Usually data services cannot tolerate packet loss, while realtime service can tolerate some packet loss. Taking these QoS requirements into account, a scheduler has to find an optimal trade-off between the conflicting goals of maximized user experience and efficient resource allocation.

The IEEE 802.16 standard divides the complex task of scheduling among three schedulers, namely base station downlink scheduler, base station uplink scheduler, and subscriber station scheduler. The standard specifies the functions of these schedulers. However, it does not provide details of actual scheduling mechanisms and left out these details for equipment manufacturers. Since scheduling algorithms have a profound impact on the overall capacity and performance of the system, it can be a key feature distinguishing among implementations of various equipment manufacturers [140].

Base station downlink scheduler The base station downlink scheduler manages data traffic from the base station to subscriber stations. Thus, it assures QoS for downlink service flows only. Downlink data queues are maintained at the base station. Therefore, the base station can determine the deadlines and sizes

of individual packets. This makes the job of the downlink scheduler relatively easier. The scheduler conveys the bandwidth allocation to subscriber stations through DL-MAP messages.

Base station uplink scheduler The base station uplink scheduler distributes bandwidth among uplink service flows. The uplink channel is the limiting factor in broadband wireless networks, which is critical for satisfactory provision of QoS to the user applications. Unlike the downlink queues, the uplink data queues reside in the subscriber stations and the uplink scheduler cannot directly know the length of queues and the deadlines and sizes of individual packets. Therefore, the uplink scheduling is to be done according to some estimates. Consequently, the task of the uplink scheduler is much more complex than that of the downlink scheduler. Uplink scheduling decisions are transmitted to subscriber stations using UL-MAP messages. For the uplink service flows, QoS is assured by cooperation between the uplink scheduler and the subscriber station scheduler.

Subscriber station scheduler The subscriber station scheduler is responsible for distributing bandwidth, which is granted by the uplink scheduler, among service flows. The UL-MAP contains uplink grants on per subscriber basis and not on per connection basis. It is the duty of the subscriber station scheduler to distribute uplink bandwidth among its connections. As a subscriber station has more accurate view of the uplink data queues than the base station, therefore this approach allows more accurate and timely scheduling decisions. Based on the QoS parameters and service class priority, the base station and subscriber stations always have a common order in which the service flows should be scheduled [140].

2.3 CAC Algorithms for WiMAX

The CAC schemes proposed for the IEEE 802.16 standard can be classified into two categories [133], i.e. flexible and conservative. A flexible admission control policy degrades the QoS of existing connections to obtain resources for a new connection. On the other hand, a conservative policy maintains the QoS of established connections and simply rejects a new connection if enough resources are not available.

2.3.1 Flexible CAC Policies

Flexible policies degrade the resources provided to the established connections if enough resources are not available to support the new connection. These policies

can be further classified into three categories [133]: (i) Service degradation (ii) bandwidth borrowing (iii) and bandwidth stealing.

Service degradation These policies decrease the bandwidth assigned to the existing connections that have lower service priority than that of the incoming connection. Y. Ge and G-S. Kuo suggested that these type of CAC policies should only be used for handover operation [79].

Bandwidth borrowing These admission control schemes decrease the bandwidth assigned to rate-adaptive connections in case of network congestion. D. Niyato and E. Hossain defined a bandwidth borrowing scheme [138] as a non-cooperative game strategy. The objective of the game is to find the optimal equilibrium point between rtPS and nrtPS connections to obtain resources for the incoming connection. H. Wang and W. Li integrated bandwidth borrowing with stepwise degradation [212]. In this scheme, a fixed amount of bandwidth is reserved for UGS connections. An incoming UGS connection is accepted if and only if enough bandwidth is available to provide QoS to both new and existing UGS connections. Wang et al. proposed a proportional bandwidth borrowing with a guard channel policy [213]. A guard channel is simply the portion of the channel capacity reserved for handoff operations. Proportional bandwidth borrowing is utilized when sufficient bandwidth is not available for a new connection.

Bandwidth stealing These CAC schemes are based on token-bucket approach. Jiang and Tsai provided a bandwidth stealing scheme [100] in which an uplink connection is characterized by two token-bucket parameters, i.e. token rate and bucket size. Realtime connections have a delay parameter as well. The scheme combines bandwidth stealing with a threshold-based policy. A threshold is defined for each class of service. Bandwidth stealing is done from classes that are using bandwidth more than their threshold.

2.3.2 Conservative CAC Policies

Degradation of QoS of the established service flows should be avoided because this causes reduced user satisfaction [39]. The degradation in the QoS level may eventually result in revenue loss [70]. Therefore, in the conservative schemes [22] an incoming connection is only accepted if the QoS guarantee could be provided to the new connection without sacrificing the QoS of the established connections. S. Chandra and A. Sahoo suggested the use of a hyper interval to test the admissibility of new requests [46]. This is advantageous because the QoS requirements of connections change over time. They proposed to block a new connection if the delay and jitter requirements of existing connections could not be guaranteed. A similar CAC scheme [225] was provided by O. Yang and J. Lu.

K. Eunhyun et al. proposed a solution [65] based on Markovian model. The proposed solution takes into account handover and change in modulation schemes of the existing connections. The model supports only two modulation schemes and assumes that all connections have fixed and equal requirements. These assumptions limit the applicability of this solution [133]. A conservative CAC scheme [163] that considers the power of radio channel was proposed by Rong et al. The authors proposed to use adaptive power allocation to maximize the usage of base station power. However, the proposed scheme is limited to Fixed WiMAX and subscriber stations with non-variable bandwidth needs.

2.4 Scheduling Algorithms for WiMAX

Chakchai So-In et al. reported [191] that scheduling algorithms such as Fair Scheduling [124], Min-Max Fair Scheduling [200], Distributed Fair Scheduling [204], Channel-State Dependent Round Robin [34], Feasible Earliest Due Data [178], and Energy Efficient Scheduling [101] could not provide QoS in WiMAX networks due to the specific features of the technology. A packet scheduling algorithm for WiMAX must take into account the service class and QoS requirements of each connection. A scheduler can choose to optimize one or more transmission characteristics such as throughput, fairness, and maximum operator revenue[8]. Usually, the scheduling schemes proposed for WiMAX first distributes the available bandwidth among different service classes. Then class-specific algorithms are used to distribute the bandwidth, allocated in the first step, among service flows.

Inter-class scheduling The algorithms that distribute network resources among various classes of traffic are called inter-class scheduling algorithms. These algorithms consider the various characteristics of traffic classes and the associated QoS parameters to determine the optimal resource allocation. In inter-class scheduling, an important issue is to decide the number of queues maintained by the scheduler. Many schedulers [56] [168] [45] [47] [49] proposed for WiMAX maintain separate queues for each class of traffic. However in [214], Y. Wang et al. proposed to combine rtPS, ertPS, and UGS queues to simplify scheduling and therefore lower computational complexity. Similarly, J. Borin and N. da Fonseca proposed [37] to use a single queue for rtPS and nrtPS classes. A second queue, which is of higher priority is used for UGS connections. If an rtPS packet reaches the deadline then it is moved to the UGS queue. The interclass scheduling schemes are normally priority-based [214] [59] [118] [221] [54]. The priority determines the order of resource allocation to each service class. Priority can be based on many factors such as service class, deadline, traffic generation rate, or queue

length [137]. In [120], UGS and rtPS classes are given the same priority, while non-realtime classes are given lower priority.

Intra-class scheduling These algorithms distribute resources within the same service class. The simplest intra-class scheduling algorithm is Round Robin (RR) [184]. However RR may be non-workconserving, as it could allocate resources to connections that do not have packets to send. Since RR cannot assure QoS, therefore some researchers [56] [168] [204] proposed to use Weighted Round Robin (WRR) for WiMAX. The weights in WRR can be determined according to various parameters such as queue length, packet delay, or the size of bandwidth request [120]. Similarly, the ratio of a connections data rate to the total average rate [221], the minimum reserved rate [86], or the pricing [167] can also be used as connection weights. The weights can be adjusted dynamically to achieve the desired level of throughput and delay. The main advantage of RR and WRR is their simplicity and constant computational complexity.

WRR does not consider the urgency of data packets and therefore it is not suitable for realtime services. Delay based algorithms are particularly suitable for realtime applications. Earliest Deadline First (EDF) [24] is the simplest delay based algorithm. It always schedules the most critical packet i.e. the packet with the earliest deadline. EDF could guarantee the delay requirements of realtime traffic. Some delay based algorithms, such as Largest Weighted Delay First (LWDF), can also be used to guarantee the minimum reserved traffic rate [154]. Another delay based algorithm is Delay Threshold Priority Queueing (DTPQ) [108]. DTPQ takes the deadlines of realtime traffic into account only when the head-of-line packet exceeds a specific delay threshold. This scheme tries to find a good mix of packet loss rate for rtPS traffic and average throughput for nrtPS traffic. The main drawback of these schemes is their high computational complexity, which may be a limiting factor in high speed networks.

Based on the usage of channel state information, the scheduling algorithms at the MAC layer can be categorized into channel-aware and channel-unaware algorithms [74] [182]. Channel-unaware algorithms do not consider channel state in decision making. While, channel-aware algorithms make decisions according to the channel state information.

2.4.1 Channel-Aware Algorithms

Channel-aware algorithms are also known as opportunistic algorithms or cross-layer algorithms. These algorithms make scheduling decisions according to channel

station information. The packets are transmitted only when the channel condition is good and therefore the packet loss probability is below a pre-defined threshold. There are mathematical [63] [193] and heuristic [75] [84] [123] models that could be used to predict future channel state information [17]. The base station could retrieve downlink channel state through Channel Quality Indicator (CQI) channel. The uplink channel state can be measured directly on previous transmissions.

Channel-aware algorithms improve throughput for delay tolerant traffic where the physical link has time-varying characteristics. With delay tolerant traffic, the scheduler could choose the best available channel to transmit data at high rate [198]. However with these schemes, the delay experienced by users with poor channel conditions may not be tolerable and consequently it may not be possible to meet the minimum service levels of these users. These algorithms give higher priority to subscriber stations with better channel conditions, which may result in under-allocation or no allocation at all to some subscriber stations with high error rate. Channel-aware algorithms can be classified on the basis of transmission parameter they aim to optimize [191]. On this basis, channel-aware algorithms can be classified as shown in Table 2.6.

Fairness

The fairness of resource allocation is an important issue in scheduling. Fairness can be either long-term or short-term. Short-term fairness guarantees long-term fairness but not vice-versa [112]. Various opportunistic scheduling schemes [222] [31] [109] [90] [164] were proposed that aim to maximize fairness of resource allocation. These schemes mainly ensure fairness of resource distribution among BE connections only. One of the fundamental fair algorithms for BE class is Proportional Fairness Scheme (PFS) [31] [109]. PFS calculates the ratio of channel capacity to the long-term throughput for each connection and then allocates bandwidth to the connection with the highest ratio. Thus, the algorithm ensures long-term fairness. The long-term throughput is determined over a time window. The performance of PFS with time-windows of various sizes was provided in [90]. F. Hou et al. proposed [90] to determine long-term throughput according to connection delays. J. Qiu and T. Huang proposed a timer [152] to prevent starvation of a subscriber station longer than a specified interval. C. So-In et al. proposed to use Generalized Weighted Fairness (GWF) [190] approach. GWF equalizes a weighted sum of the time-slots and bytes. This assures minimal level of QoS with acceptable system performance. Based on the selected set of weights, either slot fairness or throughput fairness could be achieved [191]. However, these schemes do not provide delay guarantees and therefore could not be used for realtime applications.

Optimization parameter	Algorithms	Advantages	Drawbacks
Fairness	Proportional Fairness Scheme and its variations [31] [109] [90], Generalized Weighted Fairness [190]	Ensures fairness of resource allocation among BE flows	Not appropriate for realtime applications
Throughput	Algorithms based on SNR [207] [178] [189] [30], Exponential rule [178], Temporary Removal Scheduler [28], Opportunistic Deficit Round Robin [156], Frame Registry Tree Scheduler (FRTS) [223], Cross-layer scheduling algorithm [121]	Efficient usage of network resources	(i) Not appropriate for realtime applications, (ii) Starvation
Minimized Delay	Modified Largest Weighted Delay [25], Adaptive rtPS scheduler [134], Modified M-LWDF [142], Urgency and Efficiency based Packet Scheduling [165]	support realtime applications	(i) Unfair distribution, (ii) Low throughput

Table 2.6: Classification of channel-aware packet schedulers

Throughput Maximization

The schedulers that aim to maximize system throughput [207] [178] [189] [30] use signal-to-noise ratio (SNR) as the decision making criteria. Priority is given to subscriber stations with the highest value of SNR. S. Shakkottai and R. Srikant suggested the use of exponential rule [178] to achieve maximum system throughput. According to the exponential rule, a scheduler can assign the maximum number of slots derived from the minimum modulation scheme to each connection. Next, the weights can be adjusted according to the exponent(p) of the instant modulation scheme over the minimum modulation scheme. In this way, the users with better modulation scheme get more bandwidth.

Temporary Removal Scheduler (TRS) [28] is an example of such algorithms. It analyzes radio conditions and then temporarily removes packets from the scheduling list, for a specified duration TR , associated with subscriber stations experiencing bad channel conditions. Then, opportunities are provided to the packets still in the scheduling list. After TR expires, the removed packets are examined and are returned to the scheduling list if there is an improvement in the channel conditions. After a certain number of retries, a removed packet is unconditionally added at the top of the scheduling list.

Another channel-aware scheduler is Opportunistic Deficit Round Robin (O-DRR) [156]. In O-DRR, the base station periodically polls to determine the subscriber stations to be served. A subscriber station is given an opportunity to send data, if it has data to transmit and the SNR is above a certain threshold value. TRS and O-DRR do not consider the various service classes and make scheduling decisions according to channel state information only.

S.A. Xergias et al. provided an opportunistic scheduler called Frame Registry Tree Scheduler (FRTS) [223]. FRTS make decisions according to packet deadlines for rtPS and UGS classes. While for nrtPS and BE classes, deadlines are considered to be infinite. Thus, higher priorities are assigned to UGS and rtPS flows. If a subscriber station changes the modulation scheme or QoS requirements, the scheduler re-evaluate its decision criteria.

Q. Liu et al. introduced a cross-layer scheduling algorithm [121] in which transmission opportunities and time-slots are assigned according to priorities. Priority of each subscriber station is determined according to its channel conditions and the service class requested. In [189], a heuristic approach was provided to maximize throughput. In this approach, a complete subchannel is allocated to a mobile station so that it could transmit the maximum amount of data. The goal is to minimize the total unsatisfied demand that is

$$\text{Min}[\sum_{1 \leq i \leq n} \lambda_i - (\sum_{1 \leq j \leq m} R_{ij} N_{ij})] \quad (2.1)$$

where m is the number of subchannels, n is the number of mobile stations, R_{ij} is the data rate of mobile station i on channel j , N_{ij} is the number of slots allocated to station i on subchannel j . The main drawback of this scheme is its polynomial computational complexity i.e. $O(nmN)$. These schemes could not provide delay guarantees. Furthermore, they could lead to starvation of subscriber stations with poor channel conditions.

Minimized Delay

Some scheduling schemes [25] [142] [165] [143] [107] aim to assure QoS to realtime applications by minimizing transmission delays. These schemes are particularly suited for realtime classes such as UGS, rtPS, and ertPS. However, unfair allocation of resources is a major drawback of these schemes. Moreover, these schemes provide relatively lower total throughput. One such scheme is Modified Largest Weighted Delay (M-LWDF) [25]. M-LWDF can assure QoS by providing the minimum throughput guarantee and delays smaller than a pre-specified threshold with a given probability. The algorithm assigns a weight to each class and takes into account both the channel conditions and queue length. The main challenge is to determine the optimal set of class weights.

An opportunistic algorithm specifically proposed for rtPS class is adaptive rtPS scheduler [134]. The algorithm tries to reduce the delay incurred during multiple bandwidth request/grant operations. It assigns bandwidth to the subscriber stations for the packets currently in the data queue and for the packets expected to arrive before next time frame. The algorithm utilizes a prediction method to estimate packet arrival times.

P. Parag et al. proposed a modified version [142] of M-LWDF. The modified algorithm selects users on each sub-carrier for every time-slot using the following expression:

$$\text{Max}[\text{channel_gain}(i, k) \times \text{HOL_delay}(i) \times \frac{a(i)}{d(i)}] \quad (2.2)$$

where a is the arrival rate and d is the throughput. The values of a and b are averaged over a sliding window. HOL_delay is the value of head of line delay. A similar scheduling scheme called Urgency and Efficiency based Packet Scheduling (UEPS) was proposed in [165]. UEPS takes into account the efficiency of radio resources and the urgency of enqueued packets. The algorithm first assigns priority to each user based on the time-utility function ($U'_i(t)$) and the ratio of current channel conditions ($R_i(t)$) to the average channel conditions ($R'_i(t)$). The value of i can be determined as shown in Equation 2.3. The main drawback of UEPS is that it could actually assign higher priority to nrtPS and BE flows than to rtPS flows.

$$i = \max | U'_i(t) | \times \frac{R_i(t)}{R'_i(t)} \quad (2.3)$$

2.4.2 Channel-Unaware Algorithms

Channel-unaware schedulers do not consider channel state in making scheduling decisions. The channel is assumed free of errors and the variable nature of wireless link is ignored. Channel-unaware algorithms for WiMAX can be further classified into two categories [44]: homogenous and hybrid.

Homogeneous Schedulers

These schedulers assign one classical scheduling algorithm for all classes of service and do not take into consideration the various characteristics of WiMAX service classes. Examples of such algorithms are RR, DRR, WRR, and EDF. None of the homogeneous algorithms could provide fairness and QoS guarantees for all the services classes introduced in WiMAX [19]. For example EDF fulfills the QoS requirement of rtPS class, however for UGS class it introduces unnecessary complexity and therefore it is not appropriate.

Hybrid Schedulers

Hybrid algorithms were introduced to avoid the problems that occur in homogeneous scheduling. These algorithms use different scheduling schemes for different classes of traffic. For examples EDF can be used for rtPS class, WRR for nrtPS class, and RR for BE class. Usually, these schedulers use priority based schemes [50] [177] [47] [221] [18] for inter-class resource allocation and assign bandwidth to BE connections only when the QoS requirements of higher priority classes are fulfilled. Hybrid schemes provide some degree of fairness of resource allocation. However, these schemes do not consider the variable channel conditions of subscriber stations.

Y. Shang and S. Cheng proposed a hierarchical packet scheduling model [180] for the base station uplink scheduler. The model is an extension of the work [33] done by J. Bennett and H. Zhang. In [33], the authors provided a Hierarchical Worst-case Fair Weighted Fair (HWF^2Q^+) scheduler. The scheduler groups flows with similar QoS requirements into sets and then assigns weight to these sets. The bandwidth is then allocated to these sets according to their weights. Y. Shang and S. Cheng argued that the model, proposed in [33], could not satisfy the QoS requirements of multimedia traffic because it does not take into account the QoS needs of realtime traffic. They proposed to divide the traffic into QoS-enabled and Best-effort classes. The QoS-enabled traffic is further divided into "hard-QoS" and "soft-QoS"

classes. For QoS-enabled traffic, if $MRTR^3 < MSTR^4$ then it is classified as soft-QoS traffic. Otherwise, it is classified as hard-QoS traffic. The proposed scheduler consists of two levels. Firstly, the scheduler distributes the uplink bandwidth among three logical servers i.e. hard-QoS server, soft-QoS server, and best-effort server. Then at the second level the hard-QoS server schedules UGS traffic, the soft-QoS server schedules rtPS and nrtPS traffic, and the best-effort server schedules BE traffic. The soft-QoS server can also steal the resources of the BE server to obtain additional bandwidth. The co-scheduling server and the three logical servers, all implement WF^2Q [228] [32] algorithm. The simulation results provided by the authors show improvement in delay and jitter of soft-QoS traffic as compared to the Bennett and Zhang model. The proposed scheduling model does not distinguish between realtime and non-realtime traffic, as it does not take into account the deadlines of realtime traffic. Therefore realtime packets may miss their deadlines.

K. Wongthavarawat and A. Ganz provided a hierarchical scheduling algorithm [221] and an admission control policy for the IEEE 802.16 broadband wireless access system. The admission control admits a new connection if enough resources are available to guarantee both bandwidth and delay for the connection without compromising the QoS of existing connections. The proposed scheduler uses strict priority to distribute bandwidth among various classes of traffic. The distribution is done in the order UGS, rtPS, nrtPS and BE. This implies that UGS flows have the highest priority, while BE class has the lowest priority. Then at the next level, intra-class scheduling algorithms are used to distribute bandwidth among service flows of the same class. For UGS connections, fixed bandwidth allocations are done as specified by the IEEE 802.16 standard. For rtPS intra-class scheduling, EDF [80] is used and for nrtPS connections Weighted Fair Queue (WFQ) [60] is used. After allocating bandwidth to all QoS classes, the remaining bandwidth is equally distributed among BE connections. The proposed scheme uses the concept of arrival-service curve [58] to estimate the sizes and deadlines of the packets arrived in the previous time frame for each connection. Due to strict priority, lower priority flows may starve under high load. For rtPS scheduling, if the available bandwidth is not sufficient to fulfill the demand, then the available bandwidth is distributed among connections according to their average data rates. This distribution may result in some wastage of bandwidth due to over-allocation.

Cicconetti et al. argued that the minimum reserved traffic rate is a basic QoS parameter for each class of service and therefore the class of latency-rate algorithms [195] is suitable for scheduling in WiMAX. They proposed [56] to use DRR [183] for the base station download scheduler. DRR requires that the size of head-of-line packet of each queue is known in advance. This is possible for the

3. Minimum reserved traffic rate

4. Maximum sustained traffic rate

downlink queues but not for the uplink queues. Therefore, for the base station uplink scheduler, they proposed to use WRR [104] algorithm. The proposed model is validated with simulations. For analysis, they did not consider throughput as a performance indicator. Instead, they used delay and jitter as performance metrics. The simulation results show that BE traffic could starve under high network load.

L. Chan et al. proposed a two-tier scheduling strategy [45] for the base station uplink scheduler. The first tier is category-based, while the second tier is weight-based. The main objective of the strategy was to assure QoS and at the same time avoid starvation of lower priority classes. Firstly, each connection is classified in one of the following three categories:

Unsatisfied A connection is classified as Unsatisfied, if the bandwidth allocated to it is less than its specified minimum reserved rate.

Satisfied A connection is Satisfied, if the bandwidth allocated to it is between its minimum reserved traffic rate and the maximum sustained traffic rate.

Over-satisfied A connection is categorized as Over-satisfied, if the bandwidth allocated to it is more than its specified maximum sustained traffic rate.

Then in the first-tier, the uplink bandwidth is distributed among these categories in the order: unsatisfied, satisfied, and over-satisfied. In the second-tier, for each category the allocated bandwidth is distributed among connections according to the degree of satisfaction. The lower the degree of satisfaction of a connection, the higher the allocation priority it has. The actual amount of bandwidth to be allocated to each connection is not specified by the standard. The scheduling strategy does not make distinction between realtime and non-realtime traffic. As a result, some rtPS packets may miss their deadlines.

V. Rangel et al. provided an uplink scheduling algorithm [155], called EDF BWA Scheduling Algorithm (EBSA). It is based on the properties of 802.16 MAC protocol. The aim was to minimize delays for realtime applications especially during network congestion. The algorithm combines prioritization, EDF [24], WFQ [117], RR [116], and first-in first-out (FIFO). For UGS, rtPS, and nrtPS classes tolerated poll jitter is taken as ordering parameter. For these classes grants are allocated using the EDF principle. While FIFO scheme is used to order grants for BE flows. Finally, the grants are dispatched using a WFQ scheme, where the weights for service classes are determined by the operator.

S. Sengupta and M. Chatterjee studied the capabilities of WiMAX MAC layer to stream audio and video traffic. They proposed [176] to exploit the flexibility of WiMAX MAC to dynamically construct PDUs according to channel conditions. MAC SDUs are converted into MAC PDUs using packing and fragmentation. The optimal PDU size is determined after analyzing channel state information. Large PDUs are preferred under good channel conditions, while small PDUs otherwise.

However, how to differentiate various classes of traffic and schedule accordingly was not discussed by the authors. Similar schemes of determining optimal MAC PDU size were proposed by H. Martikainen et al [130] and S. Sengupta et al. [175].

J. Chen et al. proposed an integrated QoS architecture [47] for IEEE 802.16 standard. They created a mapping rule [48] for providing DiffServ [35] between the network layer and the MAC layer. The aim was to provide higher layer QoS for services traversing WiMAX MAC and physical layer. For scheduling, they proposed a priority based hierarchical scheme. The service class priority is maintained in the order: UGS, rtPS, nrtPS and BE. Two queues are defined for each service class at the base station. One queue for the uplink and one for the downlink transmission. Downlink traffic is given priority over uplink traffic. In the first phase, Deficit Fair Priority Queuing (DFPQ) [48] is used as inter-class scheduling algorithm. Then in the second phase, EDF [80] is used for scheduling rtPS traffic, WFQ [60] for nrtPS traffic, and RR for BE traffic. To schedule nrtPS traffic, the MRTR is used as connection weight for the WFQ algorithm.

S. Maheshwari et al. suggested the use of a distributed QoS architecture [127] for WiMAX. In this architecture, the base station allocates bandwidth to each subscriber station according to its fixed and variable bandwidth requirements using Min-Max Fair allocation scheme [106]. The bandwidth allocated to a subscriber station is then distributed among its connections using a combination of strict priority and WFQ. The uplink scheduling comprises of two stages. In the first stage, bandwidth is distributed among service classes according to their normalized weights. In the second stage, bandwidth allocated to each service class is distributed equally among all subscriber stations. Then, excess bandwidth allocated to any subscriber station is evenly distributed among unsatisfied subscriber stations. The order of transmission among subscriber stations is determined according to the deadlines of UGS connections hosted by the subscriber stations. The proposed scheme is complex, and it does not specify the procedure to determine connection weights. Moreover, the simulations performed to validate the architecture were done using the physical layer of IEEE 802.11 and not that of WiMAX.

A. Belghith and L. Nuaymi provided a comparison [30] of various scheduling algorithms proposed for rtPS class. The comparison was done using NS-2 [139] simulator. They also provided an improved version of Maximum Signal-to-Interference Ratio (mSIR) scheduling algorithm, called Modified mSIR (mmSIR). The authors developed a new NS-2 WiMAX module, which is based on WiMAX NIST module [13]. They implemented RR, mSIR, Temporary Removal Scheduler (TRS) [28], and DRR [156] algorithms in NS-2 and analyzed the performance of these algorithms using their WiMAX module. The RR scheduler proved to be the most inefficient as it delivered the minimum number of packets. The channel aware algorithms, mSIR and combination of TRS+mSIR, performed well and delivered the

highest number of packets. These algorithms allow subscriber stations with good channel conditions to transmit data and then use the most efficient burst profile. Therefore the overall efficiency is high. However, these algorithms cause huge transmission delays. This is due to the starving of subscriber stations with low SIR. The proposed algorithm (mmSIR) improved average delay by giving transmission opportunities to subscriber stations with low SIR. However, the delay observed by rtPS traffic was still too large.

H. Safa et al. proposed a scheduling scheme [166], which is based on the work [60] of A. Demers et al. In [60], DFPQ is used as scheduling algorithm for distributing bandwidth among various service classes. The authors proposed to replace DFPQ with preemptive DFPQ (PDFPQ). Similar to [60], EDF is used for scheduling rtPS traffic, WFQ for nrtPS traffic, and RR for BE traffic. An uplink and a downlink queue is set for each service flow. The queues of rtPS class are defined as non-preemptive, while the queues of nrtPS and BE classes are set to be preemptive. An rtPS packet whose deadline may expire before the next frame is treated as a critical packet. If a critical packet arrives in one of the rtPS queues, the scheduler takes bandwidth from preemptive queues to schedule the critical packet.

T-C. Tsai. et al. provided a connection admission control and scheduling mechanism [100] based on Token Bucket model [199]. The admission control is based on the estimates of bandwidth requirements of each service class and the delay requirements of rtPS connections. To avoid starvation of low priority flows, a threshold of maximum allocatable bandwidth is set for each class. If a service class uses more bandwidth than its threshold, then its priority is decreased. Bandwidth distribution is done according to class priorities. The scheme specified the details of bandwidth allocation to classes. However, how bandwidth is distributed among service flows was not specified.

A. Sayenko et al. proposed a scheme [169] similar to WRR. The scheme treats each connection as a separate session. The QoS requirements are used to determine the required number of frame slots, which then become the weights for WRR. The scheduling scheme comprises of three stages: (i) allocation of minimum number of slots (ii) allocation of unused slots (iii) ordering of slots to improve the provisioning of QoS. The first stage is mandatory, while the other two are optimization steps. The calculation of the number of slots for rtPS and nrtPS is identical, and the algorithm does not take into account the deadlines of rtPS packets.

J. Sun et al. proposed a hierarchical and distributed QoS scheduling strategy [196]. Each connection is assigned a weight according to its service type. The strategy defines two types of queues at the base station, called Type I and Type II. Type I queues are used to schedule grants for UGS connections and to assign dedicated request opportunities to rtPS and nrtPS connections. The scheduling of Type I queues are done in FIFO order. Type II queues are always processed after

Type I queues. These queues are used to allocate bandwidth to rtPS, nrtPS and BE connections according to the bandwidth requests received at the base station. These queues are processed by a fair queuing (FQ)[179] algorithm that consists of two phases. In the first phase, the algorithm tries to satisfy the minimum reserved traffic rate for each connection. In the second phase, any residual excess bandwidth is distributed among connections according to the pre-assigned weights. The method of determining connection weights was not specified by the authors.

2.5 Conclusion

In this chapter, we presented an overview of the QoS architecture provided by the IEEE 802.16 standard. The architecture specifies bandwidth request mechanisms, service classes and associated application types, connection admission control, and functions of packet schedulers. However, the standard does not specify the algorithms to be implemented at the CAC module and packet schedulers. We focused on CAC and uplink packet scheduling, and provided a literature survey of the algorithms proposed for these components.

The CAC algorithms proposed for 802.16 standard can be classified as either flexible or conservative. To accommodate a new connection, a flexible policy may degrade the QoS of existing connections to obtain more resources. While, a conservative policy simply rejects a new connection request, if enough resources are not available for the new connection.

The algorithms proposed for uplink scheduling can be broadly categorized into two categories: channel-aware and channel-unaware. The channel-aware algorithms make scheduling decisions according to channel-state information. Preference is given to subscriber stations with good channel conditions. While, the channel-unaware algorithms make scheduling decisions according to QoS requirements and bandwidth requests of individual connections. Both schemes have associated advantages and drawbacks. Channel-aware schedulers maximize overall system performance but they do not guarantee QoS for various classes of traffic. Channel-unaware algorithms can furnish QoS to various service classes, however they ignore the variable nature of wireless link. Despite several proposals, the IEEE 802.16 working group has not approved any algorithm, and therefore the area is still open for research.



Two-Level Scheduling Algorithm (TLSA)

Packet scheduling and connection admission control are two of the most important functions of a QoS framework. In the uplink direction packet scheduling is done by a cooperation of the base station uplink scheduler and the subscriber station scheduler. Connection admission control facilitates the task of packet schedulers by selectively admitting new connections to the network. The IEEE 802.16 standard does not specify algorithms for connection admission control and packet scheduling. Moreover, none of the algorithms proposed so far have been accepted by the standard.

In this part, we provide the details of our proposed packet scheduling and admission control algorithms. Moreover, we provide the details of an H.264 compatible video transmission framework that is based on the proposed scheduling algorithm. The detailed theoretical and experimental analysis of the proposed algorithms are also presented in this part.

The part is organized as follows. Chapter 3 provides the specifications of the connection admission control and Two-level scheduling algorithms. The chapter also provides a queuing model that can be used for the theoretical analysis of the proposed algorithm. Then, Chapter 4 presents the simulation analysis of Two-level scheduling algorithm. In the end, Chapter 5 furnishes the details of a video transmission framework based on TLSA that is used to assess the performance of the proposed algorithm in transmitting video streams.



3

CAC and Two-Level Scheduling Algorithm

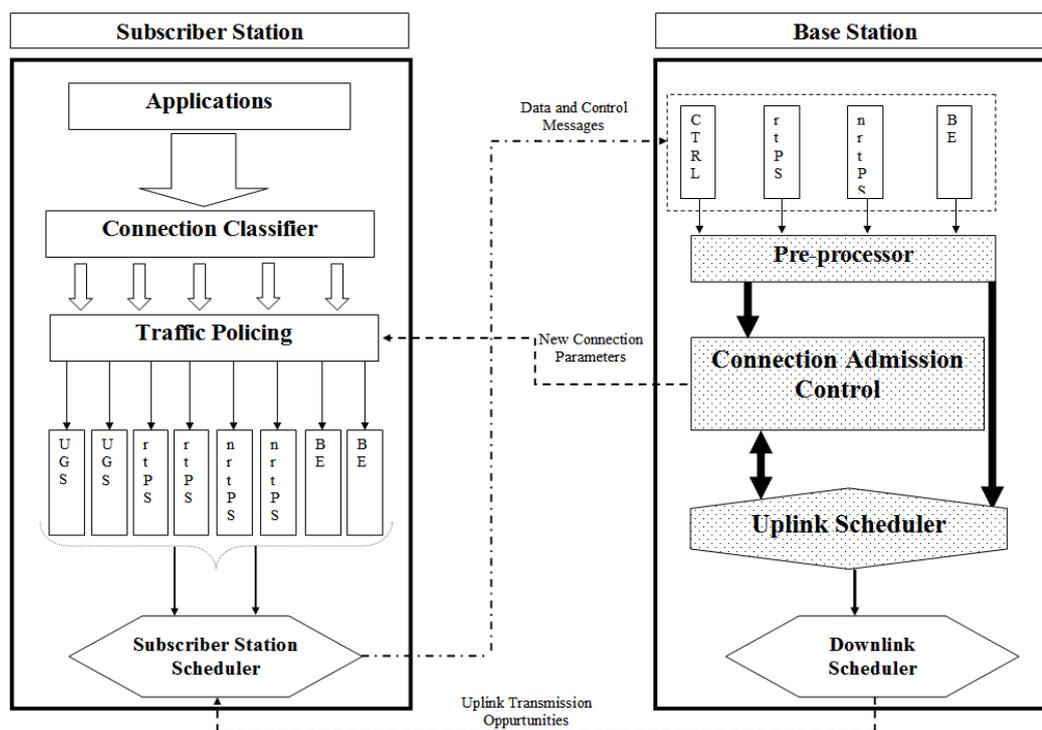


Figure 3.1: The proposed uplink QoS components

In this chapter, we provide the details of Two-Level Scheduling Algorithm (TLSA) for the base station uplink scheduler. We also provide a connection admission control algorithm that works in conjunction with TLSA to ensure QoS for various classes of traffic. Moreover, we propose several components that are required for efficient working of the uplink QoS architecture. The proposed architecture is shown in Figure 3.1. It consists of the following components:

- Pre-processor
- Data and control queues
- Traffic policing module
- CAC
- Base station uplink scheduler
- Subscriber station scheduler

The chapter is organized as follows. Firstly, we give the details of the components and information infrastructures that are proposed to facilitate the working of the proposed CAC and packet scheduling algorithms. Then in Section 3.4, the details of the CAC algorithm are provided. In the next section, we furnish the specifications of TLSA. Section 3.6 presents a detailed analytical queuing model based on Markov chains that could be used to determine the performance metrics of TLSA. Section 3.7 concludes the chapter.

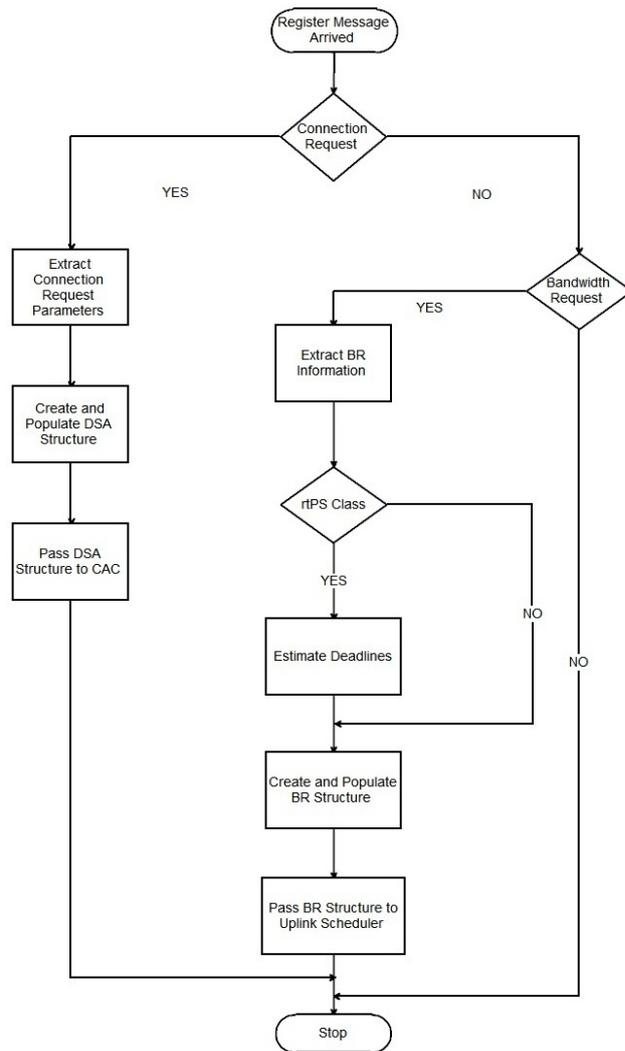


Figure 3.2: Operation of pre-processor

3.1 Pre-processor

Pre-processor is the first QoS component at the base station. It is responsible for processing the control and bandwidth request messages received from the subscriber stations. It processes these messages to extract bandwidth requests from the established connections, and connection setup requests for establishing new connections. It then passes the extracted information to the concerned modules for further processing.

We introduce an information structure, called Dynamic Service Addition (*DSA*) structure, that is used by subscriber stations to report the QoS requirements of an incoming connection to the base station. *DSA* structure is shown in Table 3.1. *DSA* structure is designed to facilitate the exchange of QoS information during connection setup. It is attached with a new connection request at the subscriber station. The pre-processing module extracts the structure from the *DSA* message and transmits this information to the CAC module. *DSA* structure contains a field called “Class”

to identify the service class requested for the new connection. Only those fields are included in the structure that are required for a particular service class. *DSA* structure can be represented by an n-tuple as shown below.

$$S_{DSA} = (SID, SFID, Class, MRTR, MSTR, Latency, Jitter, Timestamp)$$

$$\text{where, } Class \in \{0 \rightarrow UGS, 1 \rightarrow ertPS, 2 \rightarrow rtPS, 3 \rightarrow nrtPS, 4 \rightarrow BE\}$$

Pre-processor is also responsible for extracting bandwidth request information from subscriber station messages and to estimate the sizes and deadlines of the packets that arrived during the previous MAC frame. It then creates a data structure, called Bandwidth Request (*BR*) structure, and populates it with the estimated values of packet sizes and deadlines. The design of *BR* structure is shown in Table 3.2. The structure is then passed to the uplink scheduling module for further processing. A *BR* structure can be represented by an n-tuple as shown below.

$$S_{BR} = (SID, CID, Type, BRQ, Timestamp)$$

As discussed in Section 3.5, for a bandwidth request from an rtPS connection, TLSA decides the allocation scheme for the entire request and there is no need to resend the size of these packets again as normally done in aggregate bandwidth requests. Therefore, it is preferable that a subscriber station only sends the total size of data that actually arrived during the previous time frame as bandwidth request rather than the size of entire queue. This simplifies and accelerates the functioning of the pre-processor module. However over time, the perception of the base station about the bandwidth requirements of a connection may become incorrect. Therefore, the connections regularly send aggregate bandwidth requests. In this case, arrival-service curve [58] can be used to estimate the size of data traffic arrived during the previous time frame. Let $\varrho_i[f]$ be the total queue size of connection i at the start of frame f , and $\varrho_i[f - 1]$ be the total queue size at the start of frame $f - 1$. If $\Upsilon_i[f - 1, f]$ is the service received by connection i in frame $f - 1$. Then the size of the traffic ($\zeta_i[f - 1, f]$) arrived during $f - 1$ can be determined as follow.

$$\zeta_i[f - 1, f] = (\varrho_i[f] - \varrho_i[f - 1]) + \Upsilon_i[f - 1, f] \quad (3.1)$$

For realtime connections, the subscriber station scheduler drops packets that miss their deadlines. Therefore, Equation 3.1 must be modified to take into account the dropped packets. If $d_i[f - 1, f]$ is the size of packets of connection i dropped during $f - 1$ and f , then the size of traffic arrived during $f - 1$ can be determined by Equation 3.2.

Field	Description
SID	Subscriber station identifier
SFID	Service flow identifier
Class	Class of service requested for the service flow
MRTR	Requested minimum traffic rate
MSTR	Specified maximum traffic rate
Latency	Maximum tolerable delay
Jitter	Maximum tolerable variation in delay
Timestamp	Timestamp of the request

Table 3.1: Fields of *DSA* Structure

Field	Description
SID	Subscriber station identifier
CID	Connection identifier
Type	0=aggregate request, 1=incremental request
BRQ	Size of bandwidth requested
Timestamp	Timestamp of the request

Table 3.2: Fields of *BR* Structure

$$\zeta_i[f-1, f] = (\varrho_i[f] - \varrho_i[f-1]) + \Upsilon_i[f-1, f] + d_i[f-1, f] \quad (3.2)$$

The pre-processor module estimates the deadlines of the packets by adding the maximum tolerable latency to the arrival time. If δ_i is the maximum tolerable latency for connection i , then the packets arrived during frames $f-1$ and f must be scheduled before frame $f-1 + \delta_i$ to avoid expiry of deadline. The flowchart depicting the working of pre-processor is shown in Figure 3.2.

3.2 Queue Management

At the subscriber station, a separate queue is used to store data packets of each connection. The data generated by the user applications is passed through a packet classifier. It classifies packets into connections and then associates appropriate connection identifiers to facilitate scheduling. Packet classification is done according to the transmission parameters and QoS needs of the packets. Then, these packets are stored in appropriate queues such that the packets belonging to the same connection are stored in a separate queue dedicated for that connection.

At the base station, each service class has an associated queue to hold bandwidth requests until they are processed by the uplink scheduler. The bandwidth requests from all connections belonging to a particular class are grouped in a single queue. Thus, each intra-class scheduling algorithm has only one queue to process. This simplifies the design and operation of intra-class scheduling algorithms. Each element of a bandwidth request queue is a *BR* structure, which is extracted by pre-

processor. A new bandwidth request is always added at the rear of the queue. Bandwidth requests are processed in FIFO order. However, the order of uplink transmission opportunities provided to subscriber stations depends upon the scheduling algorithm used for a given service class. rtPS queue is processed first, then the nrtPS queue, and the BE queue in the end. In this way, prioritized services are provided to QoS enabled connections.

3.3 Traffic Policing

The traffic policing module at subscriber station is responsible for monitoring application traffic and to take appropriate actions to ensure that the traffic of each connection is in compliance with the traffic contract. The module discards traffic if the application data generation rate exceeds the maximum sustained traffic rate specified by the application during connection setup. The module then signals the action to the application layer so that the concerned applications can apply traffic shaping to ensure their traffic stays within the limits and thus not discarded.

3.4 Connection Admission Control (CAC)

The CAC module decides whether or not to accept a new connection request. The decision is based on the QoS requirements of both the incoming and existing connections, and the available network resources. The proposed CAC algorithm admits a new connection only if the following two conditions are satisfied: (i) The requested guarantees on delay and the minimum traffic rate could be provided to the incoming connection. (ii) Delay and throughput guarantees of the existing connections are not violated.

In case the entire uplink bandwidth is in use by the established connections, we propose to use the technique of bandwidth stealing to accommodate a new connection. However, the technique must respect the guarantees on minimum service levels of existing connections. That is, if a new connection arrives when all the network resources are in use, then the CAC module can take the resources from the established connections to admit the new connection such that at least the minimum service level could be guaranteed to both the new and all existing connections. If the minimum service level could not be guaranteed, the connection is rejected. Degrading the service levels of existing connections is undesirable as it reduces user-satisfaction [39]. Therefore, the admission of a new connection is not allowed to cause degradation of the minimum service guarantees of the existing connections. The bandwidth stealing is always done in the order BE, nrtPS, and rtPS. That is bandwidth is first taken from BE class until the total bandwidth allocated to

BE class is reduced to β_{BE} , where β_{BE} is the size of bandwidth reserved for BE class. Then, bandwidth is taken from nrtPS class until the total bandwidth allocated to nrtPS class is reduced to $\sum_{i \in \Delta_{nrtPS}} \alpha_i^{min}$, where Δ_c is the set of all connections of class c admitted by the base station, $c \in \{UGS, ertPS, rtPS, nrtPS, BE\}$, and α_i^{min} is the minimum traffic rate of connection i . Finally, if there is still some need of bandwidth for the new connections, then it is taken from rtPS class as long as conditions (i) and (ii) are satisfied. No bandwidth stealing can be done from UGS and ertPS classes. Now, we provide the necessary condition that could be used to determine if it is possible fulfill the delay requirements of an rtPS connection.

Theorem 1

Statement:

The guarantee on the maximum tolerable latency of an rtPS connection i can be provided if the following condition is satisfied

$$\alpha_i^{max} \leq \left(\frac{\delta_i}{\gamma} - 1 \right) \left((\beta - \beta^u) - \Psi' \sum_{j \in \Delta_{rtPS} - \{i\}} \alpha_j^{avg} \right) \quad (3.3)$$

where, α_i^{max} is the maximum sustained traffic rate of connection i , γ is the duration of MAC frame in seconds, β is the total uplink transmission capacity, and

$$\beta^u = \sum_{k \in \Delta_{UGS}} \alpha_k^{min} + \sum_{m \in \Delta_{ertPS}} \alpha_m^{min} + \sum_{n \in \Delta_{nrtPS}} \alpha_n^{min} + \beta_{BE}$$

Proof:

The average data generation rate for all rtPS connections, excluding connection i , is equal to $\sum_{j \in \Delta_{rtPS} - \{i\}} \alpha_j^{avg}$. Therefore, on average these connections demand

$\left(\gamma \sum_{j \in \Delta_{rtPS} - \{i\}} \alpha_j^{avg} \right)$ units of bandwidth in a frame. If Ψ' is the mean service ratio (equation 3.11), then the average of total bandwidth allocated to these connections per frame is equal to $\left(\Psi' \gamma \sum_{j \in \Delta_{rtPS} - \{i\}} \alpha_j^{avg} \right)$. This implies that the bandwidth

available for connection i per frame is equal to $(\beta - \beta^u) - \left(\Psi' \gamma \sum_{j \in \Delta_{rtPS} - \{i\}} \alpha_j^{avg} \right)$

The maximum data generation rate of connection i is α_i^{max} , therefore the maximum amount of data that could arrive during one frame is $\alpha_i^{max} \gamma$ units. Since δ_i is the maximum tolerable latency of connection i , therefore this data must be transmitted in next $\left(\frac{\delta_i}{\gamma} - 1 \right)$ frames to meet the deadline. To fulfill the delay requirements, the available bandwidth for connection i in $\left(\frac{\delta_i}{\gamma} - 1 \right)$ frames should be greater than the data generation rate of connection i . Mathematically,

$$\alpha_i^{max} \gamma \leq \left(\frac{\delta_i}{\gamma} - 1 \right) \left((\beta - \beta^u) \gamma - \Psi'_i \gamma \sum_{j \in \Delta_{rtPS} - \{i\}} \alpha_j^{avg} \right)$$

-or-

$$\alpha_i^{max} \leq \left(\frac{\delta_i}{\gamma} - 1 \right) \left((\beta - \beta^u) - \Psi' \sum_{j \in \Delta_{rtPS} - \{i\}} \alpha_j^{avg} \right)$$

which completes the proof.

Now we provide the details of connection admission procedure for each service class.

3.4.1 BE Class

Since a BE connection does not require any guarantee on throughput and delay, therefore it can always be admitted without degrading the service levels of established connections. Therefore, the proposed CAC module always admit an incoming BE connection.

3.4.2 nrtPS Class

The nrtPS connections demand assurance on the minimum throughput level. The CAC module admits an nrtPS connection i , if the condition of Equation 3.3 remains valid for the established rtPS connections and the condition given by Equation 3.4 is satisfied. Otherwise, the request is rejected.

$$\alpha_i^{min} < \beta - \left(\sum_{j \in \Delta_{rtPS}} \alpha_j^{min} + \sum_{k \in \Delta_{nrtPS}} \alpha_k^{min} + \sum_{l \in \Delta_{UGS} \cup \Delta_{ertPS}} \alpha_l^{min} + \beta_{BE} \right) \quad (3.4)$$

3.4.3 rtPS Class

An rtPS connection requires guarantees on both the minimum traffic rate and maximum delay. If a new connection (m) arrives, then the CAC module proceeds as follows:

1. If the requested MRTR is less than the available bandwidth then goto step 2, else reject the connection. Mathematically,

$$\alpha_m^{min} < \beta - \left(\sum_{j \in \Delta_{rtPS}} \alpha_j^{min} + \sum_{k \in \Delta_{nrtPS}} \alpha_k^{min} + \sum_{l \in \Delta_{UGS} \cup \Delta_{ertPS}} \alpha_l^{min} + \beta_{BE} \right)$$

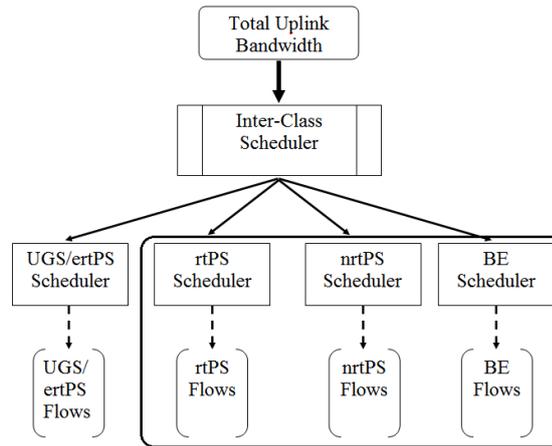


Figure 3.4: Hierarchical representation of TLSA

2. If Equation 3.3 is valid for the incoming connection then goto step 3 else reject the connection.
3. If Equation 3.3 remains valid for the existing rtPS connections then admit the connection else reject the request.

3.5 Two-Level Scheduling Algorithm

To schedule traffic with various QoS requirements and priorities, we provide a two-level scheduling scheme for the base station uplink scheduler. In the first level, an inter-class scheduling algorithm distributes available uplink bandwidth among various service classes according to their QoS requirements and available resources. Then in the second level, class-specific algorithms distribute bandwidth among the connections of a given class. A class-specific algorithm of class c takes the bandwidth allocated to class c by the inter-class scheduling algorithm and distributes it among the active connections belonging to the class. The proposed hierarchical scheme is shown in Figure 3.4. The details of the inter-class and intra-class scheduling algorithms are provided in Sections 3.5.1 and 3.5.2, respectively.

3.5.1 Inter-Class Scheduling

The inter-class scheduling algorithm follows the service class priority order as implicitly suggested by the IEEE 802.16 standard. It ensures that enough resources are provided to each service class so that the intra-class scheduling algorithms could ensure the service level guaranteed to each connection. Bandwidth distribution among service classes is done in the order: UGS/ertPS, rtPS, nrtPS, and BE. This implies that the UGS and ertPS classes have the highest priority, while the BE class has the lowest priority. The main objectives of the inter-class scheduling algorithm are as follows:

- The QoS level guaranteed by the CAC module is ensured for all service classes.
- Higher priority flows are not affected by the lower priority flows.
- No service class starves.
- Efficient bandwidth utilization

The scheduling of UGS and ertPS classes are similar and well-defined by the IEEE 802.16 standard. The standard specifies fixed-bandwidth allocation on periodic basis for these classes. The fixed-bandwidth allocation to be done to each connection is equal to its MRTR. Let α_k^{min} be the MRTR of connection k , where $k \in \Delta_{UGS} \cup \Delta_{ertPS}$. Then the algorithm allocates $\sum_{j \in \Delta_{UGS}} \alpha_j^{min}$ units of bandwidth to the UGS class. Let Δ_{ertPS}^a be the set of ertPS connections in active state, then the algorithm allocates $\sum_{l \in \Delta_{ertPS}^a} \alpha_l^{min}$ units of bandwidth to the ertPS class. Note that Δ_{ertPS}^a is an improper subset of Δ_{ertPS} , i.e. $\Delta_{ertPS}^a \subseteq \Delta_{ertPS}$.

For the nrtPS class, the algorithm ensures that enough bandwidth is allocated to the class so that the intra-class scheduling algorithm could provide at least the MRTR to each connection. The minimum bandwidth allocated to the nrtPS class is equal to the sum of the minimum traffic rates of all admitted nrtPS connections. At the start of frame f , the algorithm begins by allocating $min(\sum_{i \in \Delta_{nrtPS}} \alpha_i^{min}, \sum_{i \in \Delta_{nrtPS}} \varrho_i[f])$ units of bandwidth to the nrtPS class. Henceforth, this allocation of bandwidth is represented by β_{nrtPS} . β_{nrtPS} is the minimum amount of bandwidth allocated to the nrtPS class. Mathematically,

$$\beta_{nrtPS} = min \left(\sum_{i \in \Delta_{nrtPS}} \alpha_i^{min}, \sum_{i \in \Delta_{nrtPS}} \varrho_i[f] \right) \quad (3.5)$$

Since BE connections do not have an associated minimum traffic rate, therefore no bandwidth allocation could be done for BE class in this manner. Nevertheless to prevent starvation of the BE connections, the algorithm reserves a small part of the uplink bandwidth for the BE class. This bandwidth is denoted by β_{BE} . The value of β_{BE} is not fixed and may vary for each MAC frame. However, β_{BE} is always less than or equal to $\sum_{i \in \Delta_{BE}} \varrho_i[f]$. Mathematically, the following condition always holds

$$\beta_{BE} \leq \sum_{j \in \Delta_{BE}} \varrho_j[f] \text{ and } \beta_{BE} \leq \iota \quad (3.6)$$

where ι is the maximum possible value of β_{BE} . The value of ι can be defined by the service providers to best suit their business model.

Since in each MAC frame, β_{nrtPS} and β_{BE} units of bandwidth are reserved for nrtPS and BE classes, respectively. Therefore after scheduling UGS and ertPS

classes, $\beta - \beta_{UGS} - \beta_{ertPS} - \beta_{nrtPS} - \beta_{BE}$ bandwidth is available for the rtPS class. This is the maximum amount of bandwidth that could be allocated to the rtPS class. However, the actual bandwidth allocation depends upon the current bandwidth requirements of the rtPS class. Mathematically, the bandwidth allocated to the rtPS class (Θ_{rtPS}) in frame f is given by Equation 3.7. Henceforth, we use $\Theta_c[f]$ to denote the size of bandwidth allocated to class c in frame f , where $c \in \{UGS, ertPS, rtPS, nrtPS, BE\}$

$$\Theta_{rtPS}[f] = \min \left(\sum_{k \in \Delta_{rtPS}} \varrho_k[f], \beta - \sum_{j \in \Delta_{UGS} \cup \Delta_{ertPS}} \alpha_j^{avg} - \beta_{nrtPS} - \beta_{BE} \right) \quad (3.7)$$

If the bandwidth required by the rtPS class is less than the available bandwidth, then the residual bandwidth could be utilized by the nrtPS and BE classes. Thus, the bandwidth available to nrtPS class in frame f is equal to $\beta_{nrtPS} + (\beta - \beta_{UGS} - \beta_{ertPS} - \beta_{BE} - \Theta_{rtPS}[f])$. However, the actual size of bandwidth allocated to the class depends upon the current bandwidth requests and is given by Equation 3.8. After scheduling of rtPS and nrtPS classes, the residual bandwidth is allocated to the BE class. Obviously, at least β_{BE} bandwidth could be utilized by the BE class per frame. The maximum amount of bandwidth (Θ_{BE}^{max}) available to the BE class is given by Equation 3.9.

$$\Theta_{nrtPS}[f] = \beta_{nrtPS} + \min \left(\sum_{i \in \Delta_{nrtPS}} \varrho_i[f] - \beta_{nrtPS}, \beta - \Theta_{rtPS}[f] - \sum_{j \in \Delta_{UGS} \cup \Delta_{ertPS}} \alpha_j^{min} \right) \quad (3.8)$$

$$\Theta_{BE}^{max}[f] = \left(\beta - \Theta_{nrtPS} - \Theta_{rtPS} - \sum_{j \in \Delta_{UGS} \cup \Delta_{ertPS}} \alpha_j^{min} \right) \quad (3.9)$$

3.5.2 Intra-Class Scheduling

rtPS Scheduling

In this subsection, we provide the details of rtPS intra-class scheduling algorithm. We begin by introducing a parameter that is used by the intra-class scheduling algorithm for ensuring fair bandwidth distribution among rtPS connections. Then we provide the details of the algorithm. At the end, an illustrative example is provided to demonstrate how the algorithm works.

Fairness In order to guarantee fairness of resource allocation among rtPS flows, we introduce a parameter called *Service Ratio*. The parameter is computed sep-

arately for each connection at the start of each scheduling round by using Equation 3.10. The value of *Service Ratio* for connection i is denoted by Ψ_i . We also introduce a parameter, called *Mean Service Ratio*, that is calculated at the start of each scheduling round by using Equation 3.11. *Mean Service Ratio* is the ratio of total service availed by all rtPS connections to total service requested by these connections. It is denoted by Ψ' . At the start of frame f , an rtPS connection i is eligible to get uplink bandwidth if $\Psi_i[f] \leq \Psi'[f]$ is true. If $\Psi_i[f] > \Psi'[f]$, this implies that there are some connections which received less service than connection i , and therefore these connections must be given priority over i . The mechanisms to ensure QoS must take into account the leading and lagging flows [72]. Ψ_i and Ψ' effectively takes bandwidth from leading flows and distribute it among lagging flows. The usage of these parameters along with the CAC algorithm ensures MRTR for each rtPS connection, and fair distribution of resources within the rtPS class.

$$\Psi_i[f] = \frac{\sum_{t=1}^{f-1} \Upsilon_i[t]}{\sum_{t=1}^{f-1} \Gamma_i[t]} \quad (3.10)$$

where, $\Gamma_i[f]$ is the bandwidth requested by connection i at the start of frame f

$$\Psi'[f] = \frac{\sum_{t=1}^{f-1} \sum_{i \in \Delta_{rtPS}} \Upsilon_i[t]}{\sum_{t=1}^{f-1} \sum_{i \in \Delta_{rtPS}} \Gamma_i[t]} \quad (3.11)$$

Definition A bandwidth allocation is called *completely fair*, if the value of *Service Ratio* of each connection is equal to *Mean Service Ratio*. Mathematically,

$$\Psi_i = \Psi_j = \Psi', \text{ for all } i, j \in \Delta_{rtPS}.$$

Scheduling To meet the deadlines of rtPS packets and to ensure deterministic service to rtPS connections, the base station provides periodic dedicated bandwidth request opportunities to these connections. It is important to determine an appropriate order of polling to guarantee deadlines, especially for connections with tight delay limits. We propose to use the maximum tolerable latency of connections as the criteria to determine the polling order of rtPS connections. The subscriber station hosting rtPS connections are polled in increasing order of the maximum latency of rtPS connections. In this way, the connection with the lowest value of maximum tolerable latency is polled first, then the connection with the second lowest value of this parameter, and so on. Thus for each uplink subframe, the base station receives

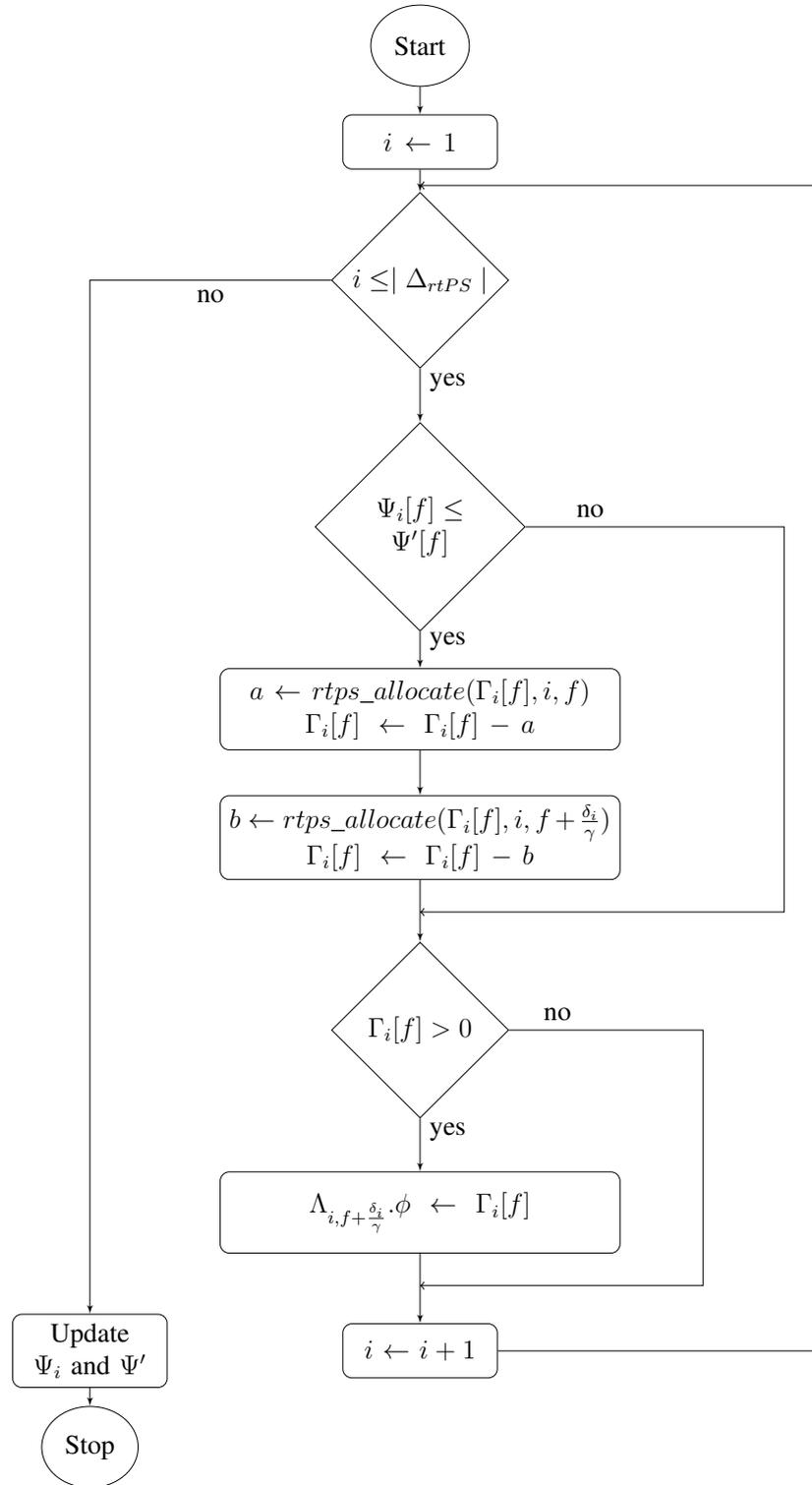
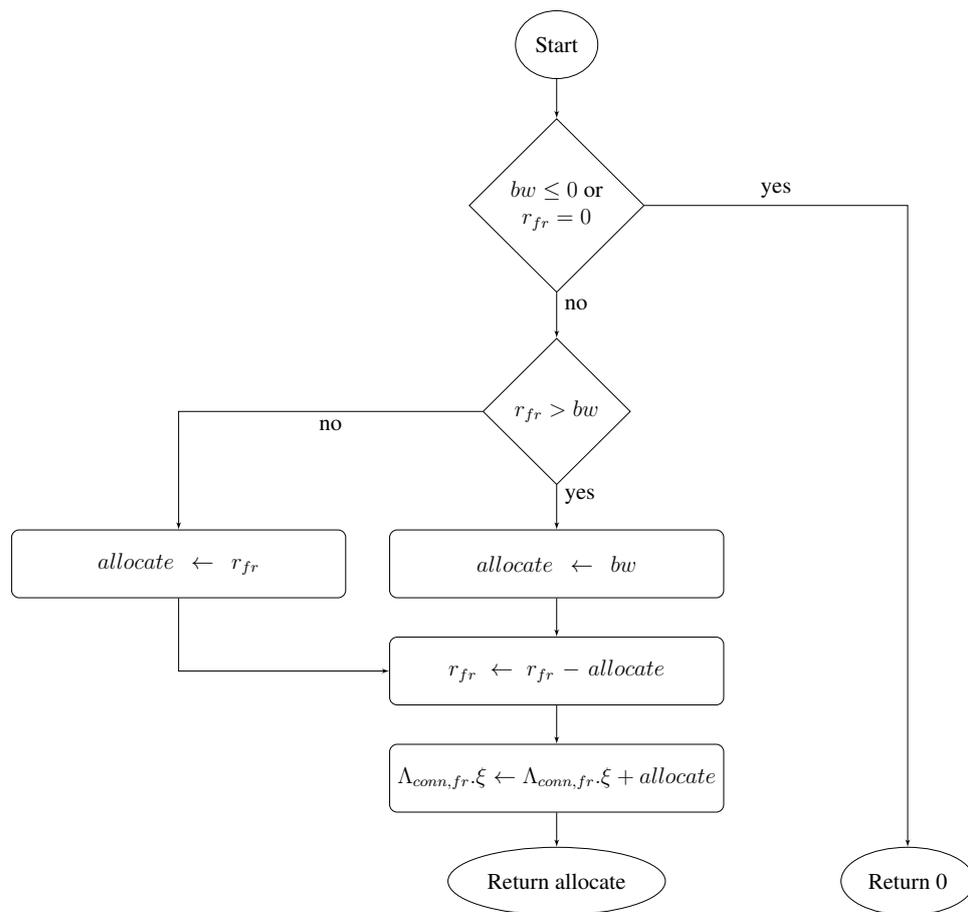


Figure 3.5: Flowchart of rtps intra-class scheduling algorithm

Figure 3.6: Flowchart of `rtPS_allocate` function

	1	2	3
i	5	0	15
ii	5	0	10 5
iii	0	10	15 0
iv	0	0	0

Figure 3.7: Bandwidth allocation table

and processes the bandwidth requests in increasing order of tolerable latency. This processing order prioritizes connections with tight delay constraints.

At the start of frame f , an rtPS connection i is eligible to receive bandwidth allocation if $\Psi_i[f] \leq \Psi'[f]$. When a bandwidth request $\Gamma_i[f]$ arrives for connection i , the base station uplink scheduler tries to allocate the requested amount of bandwidth to i in f . However if the available bandwidth in f is not sufficient to fulfill the request, then the scheduler utilizes the available bandwidth in f to schedule a part of $\Gamma_i[f]$. The remaining part of $\Gamma_i[f]$ is scheduled in frame $f + \frac{\delta_i}{\gamma}$ as discussed in the subsequent paragraphs.

To facilitate the process of bandwidth allocation, the base station uplink scheduler maintains an $|\Delta_{rtPS}| \times \delta^{max}$ table, called bandwidth allocation table. Here $|\Delta_c|$ is the cardinality of Δ_c i.e. the total number of elements in Δ_c , and δ^{max} is equal to $max(\frac{\delta_j}{\gamma})$, where $j \in \Delta_{rtPS}$. Henceforth, the bandwidth allocation table is denoted by Λ . An entry $\Lambda_{r,s}$ in Λ is an ordered pair (ξ, ϕ) , where ξ and ϕ are bandwidth allocations to connection r in frame $f + s$. ξ is called *confirmed allocation*, while ϕ is called *tentative allocation*. The allocation ξ is the guaranteed amount of bandwidth for r in frame $f + s$, while ϕ is the amount of bandwidth that could be possibly allocated to r between frames f and $f + s$. However, there is no guarantee that the algorithm would be able to make this allocation. A sample bandwidth allocation table is shown in Figure 3.7. The connections are labeled as i, ii, iii, and iv, while the frames are labeled as 1,2, and 3. The entries with gray background in the table represent *tentative* allocations. Thus, the entry $\Lambda_{ii,3}$ is (10,5), which signifies 10 units of *confirmed* allocation and 5 units of *tentative* allocation has been done for connection *ii* in frame $f + 3$. The bandwidth allocation table is also used by the base station to generate UL_MAP. At the end of each scheduling round, the first column of the allocation table corresponds to UL-MAP for the next uplink subframe. The proposed intra-class scheduling algorithm for rtPS class is shown in Algorithm 1 and the associated flowchart is shown in Figure 3.5. The step-by-step explanation of the algorithm is provided in the following paragraphs.

In Algorithm 1, the procedure `rtps_schedule` (lines 1-13) is invoked in every scheduling round to distribute bandwidth among rtPS connections. The code in *for* loop (lines 2-10), processes and schedules the bandwidth request of each connec-

tion. Function `rtps_allocate` is invoked at line 4. The function tries to allocate $\Gamma_i[f]$ units of bandwidth to connection i in frame f . If the available bandwidth in f is less than $\Gamma_i[f]$, then the available bandwidth in f is used to schedule a part of $\Gamma_i[f]$. The function then returns the amount of bandwidth successfully allocated to the connection in f . This value is subtracted from $\Gamma_i[f]$ to obtain the amount of bandwidth still needed to be scheduled for i . The remaining bandwidth could be scheduled between frames $f + 1$ and $f + \frac{\delta_i}{\gamma}$. Instead of determining the exact frame in which the allocation could be done, the algorithm makes the maximum possible allocation in frame $f + \frac{\delta_i}{\gamma}$ (line 5). In fact, if some bandwidth becomes available between frames $f + 1$ and $f + \frac{\delta_i}{\gamma}$, this allocated bandwidth could be scheduled earlier. If a part of $\Gamma_i[f]$ still remains unallocated, then it is registered as a *tentative* allocation at $\Lambda_{i, f + \frac{\delta_i}{\gamma}}$ (line 8). The condition at line 7 can be true either because $\Psi_i > \Psi'$ and therefore i is not eligible for bandwidth allocation in f , or there is some bandwidth which could not be allocated in steps 4 and 5. Regardless of the case, a *tentative* allocation is done at $\Lambda_{i, \frac{\delta_i}{\gamma}}$. Later on, if some bandwidth become available between $f + 1$ and $f + \frac{\delta_i}{\gamma}$, this entry could be converted to a *confirmed* allocation. After the bandwidth requests of all rtPS connections are processed, the values of $\Psi_1, \Psi_2, \dots, \Psi_{|\Delta_{rtPS}|}$ and Ψ' are updated according to Equations 3.10 and 3.11.

The function `rtps_allocate` is invoked by the procedure `rtps_schedule` to make an entry in the bandwidth allocation table. The definition of `rtps_allocate` (lines 17-29) is self-explanatory and it is provided for the sake of completeness. On completion, the function returns the amount of bandwidth that it is able to allocate for the connection in the specified frame.

At the end of procedure `rtps_schedule`, the first column of the bandwidth allocation table corresponds to UL_MAP for the next uplink subframe. If some bandwidth is available in the next uplink subframe, then the *confirmed* allocations with the earliest deadlines from subsequent frames are scheduled in the next frame. If some bandwidth is still available, then *tentative* allocations are scheduled in order of their deadlines. The rtPS intra-class scheduling algorithm is illustrated with the help of an example presented at the end of this subsection.

The run-time complexity of the proposed algorithm is easy to calculate. Lines 3 to 9 are executed for each bandwidth request. Lines 4 and 5 call the function `rtps_allocate`. It can be seen that all steps in the function are done in constant time. Therefore the complexity of `rtps_allocate` is $O(1)$. Similarly, statements 7 to 9 are executed in constant time. Hence, for each bandwidth request, the run-time complexity of the proposed algorithm is $O(1)$.

Algorithm 1 rtPS intra-class scheduling algorithm

```

1: procedure rtps_schedule()
2: for  $i = 1$  to  $|\Delta_{rtPS}|$  do
3:   if  $\Psi_i[f] \leq \Psi'[f]$  then
4:     set  $\Gamma_i[f] := rtps\_allocate(\Gamma_i[f], i, f)$ 
5:     set  $\Gamma_i[f] := rtps\_allocate(\Gamma_i[f], i, f + \frac{\delta_i}{\gamma})$ 
6:   end if
7:   if  $\Gamma_i[f] > 0$  then
8:     set  $\Lambda_{i, f + \frac{\delta_i}{\gamma}} \cdot \phi += \Gamma_i[f]$ 
9:   end if
10: end for
11: generate UL-MAP
12: Update  $\Psi'$  and  $\Psi_j$ , where  $j \in \Delta_{rtPS}$ 
13: end procedure
14:
15: {Function rtps_allocate attempts to reserve an amount bw of bandwidth for
    the connection conn in frame fr. It takes three parameters: (i) bw– size of
    bandwidth to allocate. (ii) conn– connection for which the allocation is to be
    done. (iii) fr– frame in which the bandwidth is to be allocated.
    It returns the amount of bandwidth successfully allocated}
16:
17: function rtps_allocate(bw, conn, fr) { $r_{fr}$  is the size of unallocated bandwidth
    in fr}
18: if  $bw \leq 0$  or  $r_{fr} = 0$  then
19:   return 0
20: end if
21: if  $r_{fr} \geq bw$  then
22:   set allocate = bw
23: else
24:   set allocate =  $r_{fr}$ 
25: end if
26: set  $r_{fr} -= allocate$ 
27: set  $\Lambda_{conn, fr} \cdot \xi += allocate$ 
28: return allocate
29: end function

```

Theorem 2

Statement: Under full network utilization, the ratio of the dropped packets χ_i to the total packets generated for connection i is independent of its data generation rate and it is given by

$$\chi_i = 1 - \left(\frac{\beta - \beta^u}{\sum_{k \in \Delta_{rtPS}} \alpha_k^{avg}} \right) \quad (3.12)$$

Proof:

Let us suppose that the average data generation rate of connection i is α_i^{avg} . Therefore, on average connection i requests $(\alpha_i^{avg} \gamma)$ units of data per MAC frame. However, the amount of bandwidth actually allocated to i , under completely fair allocation, is regulated by the mean service ratio and is equal to $(\Psi' \alpha_i^{avg} \gamma)$.

By using Equation 3.11, we have

$$\Psi' \alpha_i^{avg} \gamma = \left(\frac{\sum_{t=1}^{f-1} \sum_{i \in \Delta_{rtPS}} \Upsilon_i[t]}{\sum_{t=1}^{f-1} \sum_{i \in \Delta_{rtPS}} \Gamma_i[t]} \right) (\alpha_i^{avg} \gamma) \quad (3.13)$$

Under full network utilization, the average value of $\sum_{t=1}^{f-1} \sum_{i \in \Delta_{rtPS}} \Upsilon_i[t]$ is equal to available bandwidth for $rtPS$ class. Similarly, the average value of $\sum_{t=1}^{f-1} \sum_{i \in \Delta_{rtPS}} \Gamma_i[t]$ is equal to average combined bandwidth request of the $rtPS$ class. Mathematically,

$$Avg \left(\sum_{t=1}^{f-1} \sum_{i \in \Delta_{rtPS}} \Upsilon_i[t] \right) = \beta - \beta^u$$

and

$$Avg \left(\sum_{t=1}^{f-1} \sum_{i \in \Delta_{rtPS}} \Gamma_i[t] \right) = \sum_{i \in \Delta_{rtPS}} \alpha_i^{avg}$$

Putting these values in Equation 3.13, we have

$$\Psi'(\alpha_i^{avg} \gamma) = \left(\frac{\beta - \beta^u}{\sum_{k \in \Delta_{rtPS}} \alpha_k^{avg}} \right) (\alpha_i^{avg} \gamma)$$

This implies that average traffic drop rate is equal to

$$\alpha_i^{avg} \gamma - \left(\frac{\beta - \beta^u}{\sum_{k \in \Delta_{rtPS}} \alpha_k^{avg}} \right) (\alpha_i^{avg} \gamma)$$

and thus the ratio of drop traffic to the total traffic generated is equal to

$$\chi_i = \frac{\alpha_i^{avg} \gamma - \left(\frac{\beta - \beta^u}{\sum_{k \in \Delta_{rtPS}} \alpha_k^{avg}} \right) (\alpha_i^{avg} \gamma)}{\alpha_i^{avg} \gamma}$$

simplifying,

$$\chi_i = 1 - \left(\frac{\beta - \beta^u}{\sum_{k \in \Delta_{rtPS}} \alpha_k^{avg}} \right)$$

which completes the proof.

Illustrative Example Consider the scenario shown in Figure 3.8. In the scenario, there are three rtPS connections: A , B and C with delay limits of 30ms, 40ms and 60ms respectively. The total uplink capacity per frame is 10 units and a frame duration is 20ms. This implies that, to meet deadline, the packets generated by connections A, B and C between frames $f - 1$ and f must be scheduled within next 1, 2 and 3 frames respectively. The bandwidth requests generated by the three connections are shown in column 2. For example, the first entry in the first row of column 2, is the amount of traffic that arrived in the input queue of connection A for uplink transmission between frame 0 and frame 1. The bandwidth request for this traffic will be treated at the start of frame 1 by the base station uplink scheduler. The third column shows the values of Ψ' and Ψ_i at the start of frame f . An entry in the fourth column is the bandwidth allocation table (Λ) that is obtained at the end of scheduling frame f . The entries with gray background in Λ are *tentative allocations*. The underlined entries in Λ are the allocations done during frame f . UL-MAP corresponding to next uplink subframe is shown in the fifth column.

The scheduling in the example is done as follows. The algorithm is able to schedule the requested bandwidths in $[0, 1]$. Note specially the allocations done for connections B and C . Since only 10 units can be allocated in a frame, therefore we cannot do *confirm allocation* of more than 10 units in a column. For the duration $[1, 2]$, $\Psi_A \leq \Psi'$ but no bandwidth is available in the current frame. Furthermore, due to delay limits this request cannot be fulfilled in the subsequent frames. Therefore, it is entered as a *tentative allocation* in the column corresponding to the next frame.

Duration [f-1,f]	Bandwidth Requests			Service Ratios				Bandwidth Allocation Table	UL-MAP																		
	A	B	C	Ψ_A	Ψ_B	Ψ_C	Ψ'		A	B	C																
[0,1]	8	12	5	1.00	1.00	1.00	1.00	<table border="1"> <tr><td></td><td>1</td><td>2</td><td>3</td></tr> <tr><td>A</td><td>8</td><td></td><td></td></tr> <tr><td>B</td><td>2</td><td>10</td><td></td></tr> <tr><td>C</td><td></td><td></td><td>5</td></tr> </table>		1	2	3	A	8			B	2	10		C			5	8	2	
	1	2	3																								
A	8																										
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C			5																								
[1,2]	5	0	15	1.00	1.00	1.00	1.00	<table border="1"> <tr><td></td><td>2</td><td>3</td><td>4</td></tr> <tr><td>A</td><td></td><td>5</td><td></td></tr> <tr><td>B</td><td>10</td><td></td><td></td></tr> <tr><td>C</td><td></td><td>5</td><td>10</td></tr> </table>		2	3	4	A		5		B	10			C		5	10		10	
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C		5	10																								
[2,3]	0	12	0	0.62	1.00	0.75	0.78	<table border="1"> <tr><td></td><td>3</td><td>4</td><td>5</td></tr> <tr><td>A</td><td></td><td></td><td></td></tr> <tr><td>B</td><td></td><td></td><td>12</td></tr> <tr><td>C</td><td>5</td><td>10</td><td>5</td></tr> </table>		3	4	5	A				B			12	C	5	10	5			10
	3	4	5																								
A																											
B			12																								
C	5	10	5																								
[3,4]	0	0	0	0.62	0.50	0.75	0.61	<table border="1"> <tr><td></td><td>5</td><td>6</td><td>7</td></tr> <tr><td>A</td><td></td><td></td><td></td></tr> <tr><td>B</td><td></td><td>12</td><td></td></tr> <tr><td>C</td><td>5</td><td>5</td><td></td></tr> </table>		5	6	7	A				B		12		C	5	5			5	5
	5	6	7																								
A																											
B		12																									
C	5	5																									
				0.62	0.71	0.75	0.70																				

Figure 3.8: An example of scheduling several frames to demonstrate the working of rtPS intra-class scheduling algorithm

As there is no provision in the current frame, therefore this request is not scheduled in UL-MAP of frame 2. In the duration [2,3], B requests 12 units of bandwidth. Since $\Psi_B > \Psi'$, therefore the algorithm allocates it as a *tentative allocation* in the frame $f + \frac{\delta_B}{\gamma}$ i.e. in frame 4. The unused 5 units of bandwidth in frame 3 are used to schedule 5 units of bandwidth from the next frame. For the duration [3,4], there is no bandwidth request. There is a *confirm allocation* of 5 units and 2 *tentative allocation* of 17 units. Therefore, 5 units can be allocated to first *tentative allocation* for B. The remaining bandwidth demand cannot be fulfilled. The final values of Ψ_i and Ψ' are shown in the last row.

It is important to understand that all unexpired packets belonging to a connection are always scheduled in the order of their deadlines by the subscriber station scheduler. The important thing is the amount of bandwidth allocated to the connection and not the actual packets against which the allocations are done. This is due to the fact that the subscriber station scheduler transmits packets in first-in first-out (FIFO) order. Consider the example given in Figure 3.9. We assume two connections A and B, with $\delta_A = \delta_B = 2$ frames. Note that the base station grants 5 units to A against demand of 10 units. However, the subscriber station scheduler schedules the packets which are at the front of A's queue. Note, however, the 5 units were granted against the second packet in the queue and not the packet at the front of the queue.

nrtPS Scheduling

The main aim of the intra-class scheduling algorithm for nrtPS class is to guarantee the minimum traffic rate for each connection, while giving prioritized service to needy connections. The algorithm first makes sure that the MRTR is satisfied for each connection. Next, it allocates more bandwidth to connections with greater

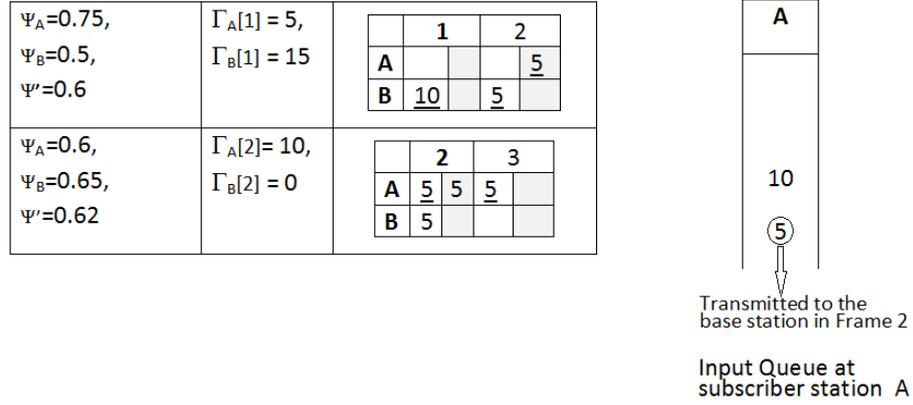


Figure 3.9: FIFO scheduling at subscriber station

queue size (backlog). Let us assume that for connection $v \in \Delta_{nrtPS}$, $\varrho_v[f]$ be the current bandwidth demand. Then for all nrtPS connections, $\forall v$, the algorithm first allocates $\min(\varrho_v[f], \alpha_v^{min})$ amount of bandwidth to v . At least α_v^{min} units of bandwidth are available for connection v , due to the policies of the CAC module. Then, the algorithm allocates the residual bandwidth among nrtPS connections in proportion of their queue sizes. After the initial allocation, the bandwidth requirements of connection u is equal to $\eta_u = \varrho_u[f] - \min(\varrho_u[f], \alpha_u^{min})$. Let r_{nrtPS} be the bandwidth still available for nrtPS class in frame f . The algorithm distribute r_{nrtPS} among nrtPS connections in proportion of η_u . Mathematically, the total bandwidth allocated to connection u is given by Equation 3.14.

$$\Upsilon_u = \min(\varrho_u[f], \alpha_u^{min}) + \min\left(r_{nrtPS}, \sum_{v \in \Delta_{nrtPS}} \eta_v\right) \left(\frac{\eta_u}{\sum_{v \in \Delta_{nrtPS}} \eta_v}\right) \quad (3.14)$$

The algorithm guarantees the minimum traffic rate to each connection. Using connection's backlog as weight enables the algorithm to expedite data transfer for more demanding connections. We illustrate the working of the algorithm with the help of an example. Consider the scenario of three nrtPS connections to be scheduled with traffic parameters as shown in Table 3.3. The total bandwidth available for nrtPS class is 12 units per frame, and the duration of each frame is one second. The MRTR is 3 units for each connection, therefore each connection sends minimum of 3 units of data in a second. The average data generation rate is 3, 6, and 9 units for connections n1, n2, and n3 respectively. The frame-by-frame bandwidth allocation by the intra-class scheduling algorithm and the backlog at the end of each frame is shown in Table 3.4. The bandwidth requested for a connection is equal to the size of its data queue at the time of the request, i.e. the backlog at the end of previous frame plus the bandwidth required for newly arrived packets. The algorithm first allocates

Connection	MRTR	MSTR	Actual Rate
n1	3	5	3
n2	3	10	6
n3	3	15	9

Table 3.3: Parameters of connections for nrtPS scheduling example

Connection		Frames			
		f_1	f_2	f_3	f_4
n1	<i>allocation</i>	$\frac{3}{3}$	$\frac{3}{3}$	$\frac{3}{3}$	$\frac{3}{3}$
	<i>request</i>	3	3	3	3
	backlog	0	0	0	0
n2	<i>allocation</i>	$\frac{4}{6}$	$\frac{4}{8}$	$\frac{4}{10}$	$\frac{4}{12}$
	<i>request</i>	6	8	10	12
	backlog	2	4	6	8
n3	<i>allocation</i>	$\frac{5}{9}$	$\frac{5}{13}$	$\frac{5}{17}$	$\frac{5}{21}$
	<i>request</i>	9	13	17	21
	backlog	4	8	12	16

Table 3.4: Bandwidth allocation for nrtPS scheduling example

3 units of bandwidth to each connection, then the remaining 3 units of bandwidth are distributed according to their backlogs as dictated by Equation 3.14.

BE Scheduling

The allocation of bandwidth at the physical layer is done in terms of time-slots. A subscriber station with bad channel conditions consumes more time-slots for transmitting relatively small amount of data. On the other hand, a subscriber station with good channel conditions can send relatively more data in the same number of time-slots. Therefore to enhance the efficiency of radio resource usage, we propose to distribute the available time-slots equally among BE connections. Let C be the number of available time-slots for BE traffic, and $|\Delta_{BE}|$ be the number of BE connections. Then the number of slots available per connection can be given as $C / |\Delta_{BE}|$. For a BE connection w , let $q_w[f]$ be the current bandwidth request and C_w time-slots are required to fulfill the request. Then the algorithm allocates $\min(C_w, C / |\Delta_{BE}|)$ time-slots to w . In case the number of available time-slots is less than the number of BE connections, the slots can be allocated to the connections in turns. A subscriber station with good channel conditions will be able to send more data within the same number of time-slots, than a subscriber station with poor channel conditions, and thus automatically get prioritized. This scheme thus prevents subscriber stations with poor channel conditions to affect the entire network, while at the same time avoids starvation of such subscriber stations.

The difference between equal bandwidth allocation and equal time-slot allocation can be illustrated with the help of an example. Suppose there are four subscriber stations: S_1, S_2, S_3 and S_4 with one BE connection each. Let the first three subscriber stations have good channel conditions and in each time slot they can send 5

(a)

	S1	S2	S3	S4	Total
Bandwidth Allocated	10	10	10	10	40
Time-slots Required	2	2	2	10	16

(b)

	S1	S2	S3	S4	Total
Time-slots Allocated	4	4	4	4	16
Bandwidth Available	20	20	20	4	64

Figure 3.10: (a) Equal bandwidth distribution (b) Equal time-slot distribution

units of data, while $S4$ has poor channel conditions and it can send only 1 unit of data per slot. We also assume that 16 time slots are available for BE traffic. Then the bandwidth allocation under the two schemes is shown in Figure 3.10. Under equal bandwidth distribution $S4$ is able to reduce the bandwidth of other connections by 50%. There is no QoS to guarantee and $S1$, $S2$ and $S3$ have good channel conditions but still they are paying the penalty of poor channel conditions of $S4$. Clearly, equal slot allocation makes use of bandwidth much more efficiently.

3.6 Theoretical Analysis of TLSA

In this section, the theoretical analysis of the proposed scheduling scheme is provided for each class of traffic. Since UGS and ertPS classes are scheduled as specified by the standard, therefore the analysis is focused on the transmission characteristics and service levels of rtPS, nrtPS, and BE classes. The analysis of the proposed scheduling algorithm is not trivial due to the following reasons:

- The proposed scheme comprises of two levels. In the second level, each service class is processed by a different scheduling algorithm.
- Each packet in an rtPS data queue has an associated deadline. A packet is dropped if it misses the deadline.
- The available bandwidth for each intra-class scheduling algorithm depends upon the bandwidth allocations by the inter-class scheduling algorithm.

Therefore we developed a queuing model specific to our scheme, so that the behavior of each intra-class scheduling algorithm can be analyzed in detail. However, it is practically not possible to capture every aspect of the proposed scheme and the transmission characteristics of various classes of traffic. Therefore to make the analysis tractable, following assumptions are made:

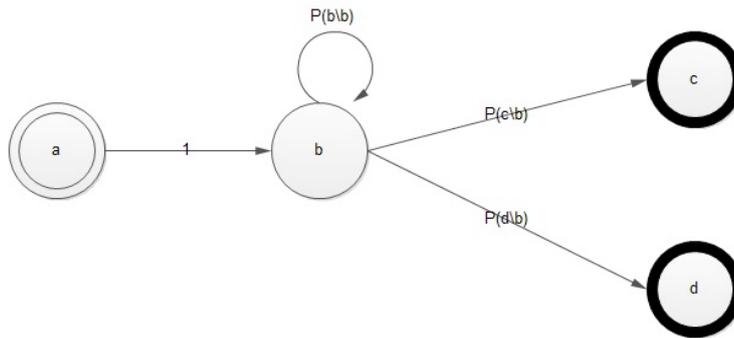


Figure 3.11: Life cycle of a data packet

- The packet arrival rate is independent of the number of packets already waiting in the queue or served by the system.
- All queues are of infinite size. Therefore, a queue can grow without bounds.
- For a given data queue, packets are served in First-in First-out order.
- Packets arrive randomly and independent of one another. Such an arrival pattern obeys Poisson distribution. While, the service time follows negative exponential distribution. Thus a single queue can be modeled as an M/M/1 system. While the overall scheme can be modeled as an M/M/c system.

3.6.1 Queuing Model

In the IEEE 802.16 standard, data arrive in the form of packets and these packets are stored in queues and scheduled on per frame basis. To analyze the system, it is important to understand the life cycle of a data packet. A packet can be in one of the four states as shown in Figure 3.11. State ‘a’ corresponds to the creation of a packet. After the creation, it is stored in a data queue. State ‘b’ corresponds to waiting state of the packet in the queue. We assume queues of infinite size, therefore a packet is never dropped due to queue overflow. A directed link signifies a state transition with an associated probability $P(y|x)$. $P(y|x)$ denotes the probability of moving to state y , given that the packet is currently in state x . Note that in the state diagram, $P(b|a)=1$. A packet, which is in state ‘b’ has three possible transitions: (i) It remains in the same state waiting for the service with probability $P(b|b)$ (ii) It gets service (state ‘c’) with probability $P(c|b)$ (iii) It gets dropped due to the expiry of deadline (state ‘d’) with a probability $P(d|b)$. For non-realtime packets, there is no associated deadline and therefore $P(d|b)=0$. Several parameters are of interest to us.

1. The average time a packet stays in state ‘b’, i.e. the average waiting time in queue
2. The probability that a packet would stay in state ‘b’ and eventually the average number of packets in state ‘b’, i.e. the average queue size
3. The probability that a packet gets service and therefore it is transmitted

4. The probability that the deadline expires and therefore the packet is dropped

The state of a queue at time 't' is defined by the total number of packets stored in the queue at time 't'. A queue is said to be in state $W(x)$, if x packets are stored in the queue. Let N_g be the maximum queue size of queue g , then the state space of queue g is the set $\{0, 1, 2, \dots, N_g\}$. If several packets move from state 'a' to 'b', then the queue is moved to a higher state. The effect of packets switching to state 'b' on queue's state can be visualized in the queue state diagram shown in Figure 3.12(a). A probability $P(z)$ associated with a state transition in the queue state diagram represents the probability of 'z' packet arrivals during the previous frame. Since the packets arrival follows Poisson distribution, therefore the probability of 'z' arrivals during a frame can be determined by Equation 3.15.

$$P(z) = \left(\frac{\lambda\gamma}{z!} \right) e^{-\lambda z} \quad (3.15)$$

Similarly, the effect of packets moving from state 'b' to states 'c' and 'd' can be represented by the state diagram as shown in Figure 3.12(b). In the figure, $P^d(x)$ represents the probability of total of x packets being dropped or transmitted during the previous frame. Let $R(e)$ be the probability that e packets missed their deadlines during previous frame, and $S(h)$ be the probability that h packets get transmitted during previous frame, then $P^d(x) = R(e) + S(h)$, where $x = e + h$.

Obviously, the actual state of queue at any time is function of $P(z)$, $R(e)$ and $S(h)$. $P(z)$ is easy to determine. However, the determination of $R(e)$ and $S(h)$ is not straightforward and requires some exercise. We would determine the mathematical relations of these probabilities and other parameters of interest by in-depth queuing analysis.

The distribution of the uplink bandwidth among service classes can be logically represented by the network diagram shown in Figure 3.14. We use the semantics proposed by VL Wallace and R. Rosenberg [211]. The semantics of the symbols are presented in Figure 3.13. μ is the average service rate of the uplink scheduler, which is divided into μ_r , μ_n , and μ_b by the inter-class scheduling algorithm. Here μ_r , μ_n , and μ_b represents the service rates of rtPS, nrtPS, and BE intra-class scheduling algorithms, respectively. In the second level, the intra-class scheduling algorithms distribute bandwidth among the connections. As an approximation, the overall uplink scheduling can be logically represented by the network diagram shown in Figure 3.15. In the figure, μ_{ij} corresponds to the service rate of queue ij . The value of μ_{ij} depends upon the value of μ_i and the scheduling scheme used. In general, the following relation always holds true

$$\mu_{i1} + \mu_{i2} + \dots + \mu_{in} = \mu_i, \text{ where } i \in \{r, n, b\}.$$

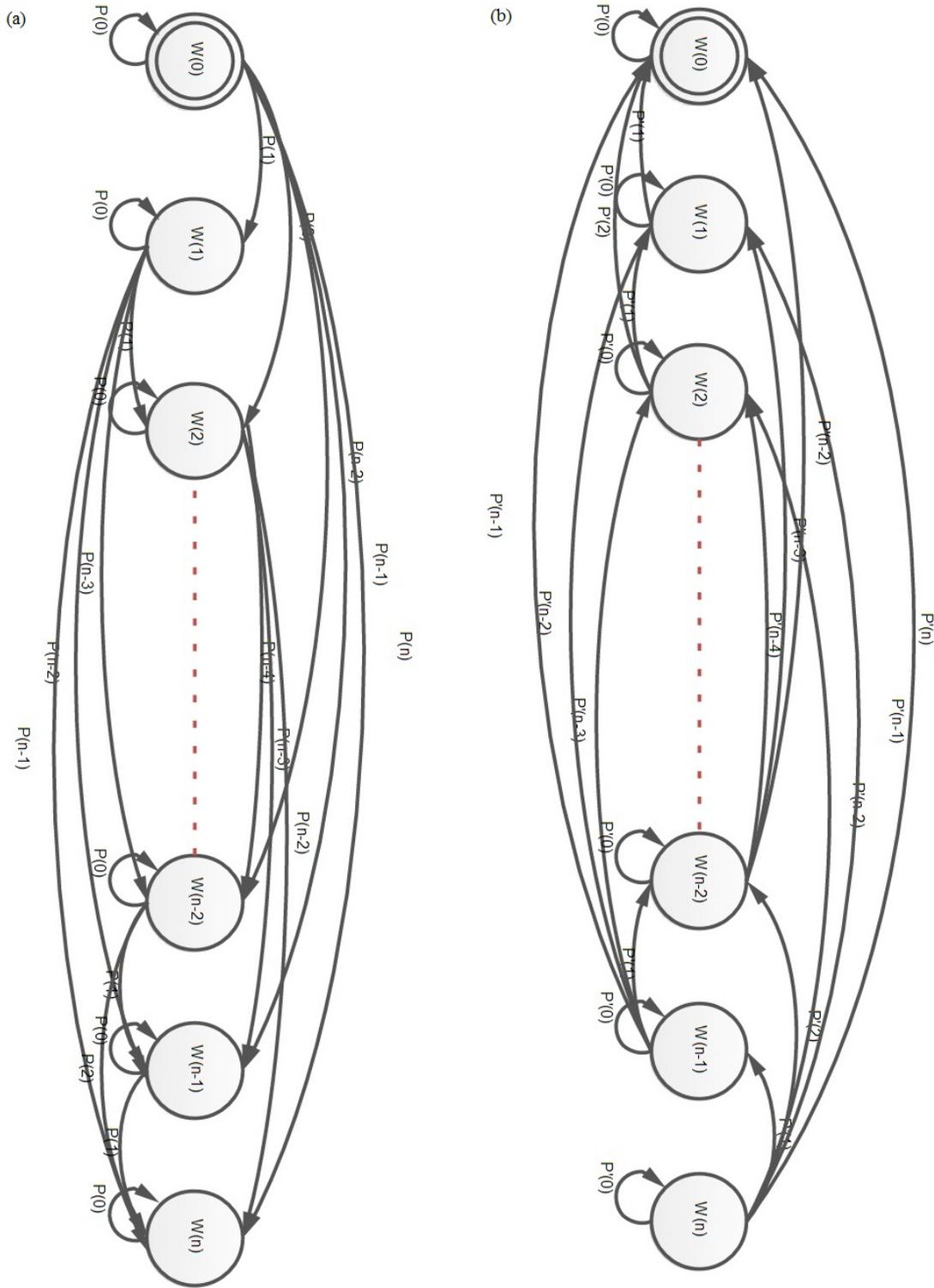


Figure 3.12: (a) State diagram of a queue representing the transition of packets from state 'a' to 'b' (b) State diagram of a queue representing the transition of packets from state 'b' to states 'c' and 'd'

Symbol	Name of Symbol	Parameter	State Set
	Queue	N : the maximum allowed number of waiting tasks	$\{0, 1, \dots, N\}$
	Exponential Server	μ : the mean rate of service for the occupied server	$\{\mathfrak{R}\}$
	Poisson Arrival Element	λ : the mean rate of packet arrival	$\{\}$
	Merge	None	$\{\}$
	Blocker	None	$\{\}$

Figure 3.13: Semantics of state variables and parameters

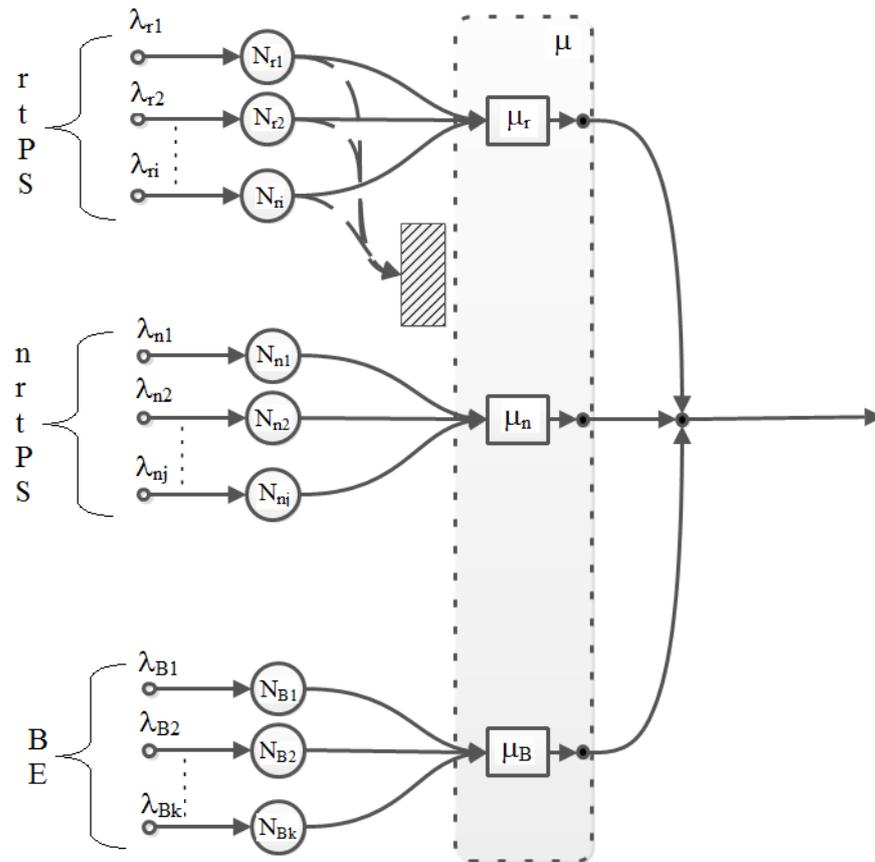


Figure 3.14: Queueing model of inter-class scheduling algorithm

3.6.2 Analysis of BE Intra-class Scheduling Algorithm

Let us assume that there are $|\Delta_{BE}|$ BE connections competing for uplink bandwidth. Connection i has an average packet arrival rate of λ_i . The MAC frame has a length of γ seconds, and ϑ_i^f is the average number of packets of connection i served per frame. This implies that the total number of frames per second is equal to $\frac{1}{\gamma}$ and thus the average number of packets served per second (ϑ_i) for connection i can be given as

$$\vartheta_i = \frac{\vartheta_i^f}{\gamma} \quad (3.16)$$

Let T_i^s be the average time to serve a packet of connection i . Then by using Equation 3.16, T_i^s can be determined as

$$T_i^s = \frac{1}{\vartheta_i} = \frac{\gamma}{\vartheta_i^f} \quad (3.17)$$

Based on equations 3.16 and 3.17, various parameters of the M/M/1 model can be determined as follow.

Scheduler utilization (ρ_i) by connection $i = \lambda_i T_i^s$

$$\rho_i = \frac{\lambda_i \gamma}{\vartheta_i^f} \quad (3.18)$$

The average response time (T_i^r) by the scheduler is equal to $\frac{T_i^s}{1-\rho_i}$

$$T_i^r = \frac{\gamma}{\vartheta_i^f - \lambda_i \gamma} \quad (3.19)$$

The average waiting time (T_i^w) for a packet of connection i in queue is equal to $\frac{\rho_i T_i^s}{1-\rho_i}$

$$T_i^w = \frac{\lambda_i \gamma^2}{\vartheta_i^f (\vartheta_i^f - \lambda_i \gamma)} \quad (3.20)$$

By using Little's formula [82], the average number of packets waiting (w_i) in data queue of connection i is equal to $\frac{\rho_i^2}{1-\rho_i}$

$$w_i = \frac{\lambda_i^2 \gamma^2}{\vartheta_i^f (\vartheta_i^f - \lambda_i \gamma)} \quad (3.21)$$

The average number of packets in the system (r_i) is given by $\frac{\rho_i}{1-\rho_i}$

$$r_i = \frac{\lambda_i \gamma}{\vartheta_i^f - \lambda_i \gamma} \quad (3.22)$$

Standard deviation of r_i (σ_{r_i}) is equal to $\frac{\sqrt{\rho_i}}{1-\rho_i}$

$$\sigma_{r_i} = \frac{\sqrt{\lambda_i \gamma \vartheta_i^f}}{\vartheta_i^f - \lambda_i \gamma} \quad (3.23)$$

Another important parameter to determine is the probability ($P(\kappa)$) of κ packets waiting in the queue. The parameter is useful in determining the queue size for which the probability of overflow is below a given threshold $P(\kappa)$. That is the size of queue should be κ to avoid overflow with a probability of $P(\kappa)$.

$$P(\kappa) = 1 - \rho_i^{1+\kappa}$$

Taking log on both sides and rearranging, we get

$$\kappa = \frac{\ln(1 - P(\kappa))}{\ln\left(\frac{\lambda_i \gamma}{\vartheta_i^f}\right)} - 1 \quad (3.24)$$

Special Case:

In the special case of all subscriber stations having identical channel conditions, equal slot distribution results in equal bandwidth distribution among BE connections. Let us assume that the total bandwidth available for BE class (Θ_{BE}) is equally distributed among $|\Delta_{BE}|$ connections, and l_i is the average packet size of connection i . Then, the average service time of connection i is given by

$$T_i^s = \frac{l_i}{\Theta_{BE}/|\Delta_{BE}|}$$

$$T_i^s = \frac{l_i |\Delta_{BE}|}{\Theta_{BE}} \quad (3.25)$$

By using Equation 3.25, equations 3.18 to 3.24 can be rewritten as follows:

$$\rho_i = \lambda_i l_i \left(\frac{|\Delta_{BE}|}{\Theta_{BE}} \right) \quad (3.26)$$

$$w_i = \frac{\lambda_i^2 l_i^2 |\Delta_{BE}|^2}{\Theta_{BE}(\Theta_{BE} - \lambda_i l_i |\Delta_{BE}|)} \quad (3.27)$$

$$T_i^w = \frac{\lambda_i l_i^2 |\Delta_{BE}|^2}{\Theta_{BE}(\Theta_{BE} - \lambda_i l_i |\Delta_{BE}|)} \quad (3.28)$$

$$r_i = \frac{\lambda_i l_i |\Delta_{BE}|}{(\Theta_{BE} - \lambda_i l_i |\Delta_{BE}|)} \quad (3.29)$$

$$T_i^r = \frac{l_i |\Delta_{BE}|}{(\Theta_{BE} - \lambda_i l_i |\Delta_{BE}|)} \quad (3.30)$$

$$\sigma_{r_i} = \frac{\sqrt{\lambda_i l_i | \Delta_{BE} | \Theta_{BE}}}{(\Theta_{BE} - \lambda_i l_i | \Delta_{BE} |)} \quad (3.31)$$

$$\kappa = \frac{\ln(1 - P(\kappa))}{\ln(\lambda_i l_i | \Delta_{BE} |) - \ln(\Theta_{BE})} - 1 \quad (3.32)$$

The average throughput τ_i of connection i is given by equation 3.33.

$$\tau_i = \frac{\vartheta_i^f l_i}{\gamma} \quad (3.33)$$

All of the above equations (3.16- 3.33) are valid for a single BE queue. Since there are $| \Delta_{BE} |$ queues and the arrival rate of each queue follows Poisson distribution, therefore the parameters calculated above for a single queue can be generalized to the entire BE class by using simple arithmetic.

$$\lambda = \sum_{j \in \Delta_{BE}} \lambda_j \quad (3.34)$$

$$\vartheta = \sum_{j \in \Delta_{BE}} \vartheta_j \quad (3.35)$$

$$w = \sum_{j \in \Delta_{BE}} w_j \quad (3.36)$$

$$\rho = \sum_{j \in \Delta_{BE}} \rho_j \quad (3.37)$$

$$r = \sum_{j \in \Delta_{BE}} r_j \quad (3.38)$$

$$T^w = \frac{\sum_{j \in \Delta_{BE}} T_j^w}{| \Delta_{BE} |} \quad (3.39)$$

$$T^r = \frac{\sum_{j \in \Delta_{BE}} T_j^r}{| \Delta_{BE} |} \quad (3.40)$$

3.6.3 Analysis of nrtPS Intra-class Scheduling Algorithm

Let Θ_{nrtPS} be the bandwidth allocated by the inter-class scheduling algorithm to the nrtPS class. Let ϱ_u be the backlog of connection u at the start of current frame. The bandwidth allocated to connection u is given by Equation 3.14. If ϱ_u is used as basis of bandwidth distribution, then the resulting model is very complex and demands an iterative solution. Therefore to make the analysis simple, we assume

that $\rho_u \propto \lambda_u$. With this assumption, equation 3.14 can be rewritten as

$$\Upsilon_u = \min(\rho_u, \alpha_u^{min}) + \min\left(r_f, \sum_{v \in \Delta_{nrtps}} \lambda_v\right) \left(\frac{\lambda_u}{\sum_{v \in \Delta_{nrtps}} \lambda_v}\right) \quad (3.41)$$

To simplify the presentation of subsequent equations we assume that $\pi_u = \min(\rho_u[f], \alpha_u^{min})$ and $\nu_u = \min\left(r_f, \sum_{v \in \Delta_{nrtps}} \lambda_v\right) \left(\frac{\lambda_u}{\sum_{v \in \Delta_{nrtps}} \lambda_v}\right)$. In the simplified form, equation 3.41 can be written as

$$\Upsilon_u = \pi_u + \nu_u \quad (3.42)$$

Let l_u be the average packet size of connection u , then we have

$$T_u^s = \frac{l_u}{\pi_u + \nu_u} \quad (3.43)$$

$$\rho_u = \frac{\lambda_u l_u}{\pi_u + \nu_u} \quad (3.44)$$

$$w_u = \frac{(\lambda_u l_u)^2}{(\pi_u + \nu_u)(\pi_u + \nu_u - \lambda_u l_u)} \quad (3.45)$$

$$T_i^w = \frac{\lambda_u l_u^2}{(\pi_u + \nu_u)(\pi_u + \nu_u - \lambda_u l_u)} \quad (3.46)$$

$$r_u = \frac{\lambda_u l_u}{\pi_u + \nu_u - \lambda_u l_u} \quad (3.47)$$

$$T_i^r = \frac{l_u}{\pi_u + \nu_u - \lambda_u l_u} \quad (3.48)$$

$$\sigma_{r_u} = \frac{\sqrt{(\pi_u + \nu_u)\lambda_u l_u}}{\pi_u + \nu_u - \lambda_u l_u} \quad (3.49)$$

and

$$\kappa = \left(\frac{\ln(1 - P(\kappa))}{\ln(\lambda_u l_u) - \ln(\pi_u + \nu_u)}\right) - 1 \quad (3.50)$$

3.6.4 Analysis of rtPS Intra-class Scheduling Algorithm

For analyzing the rtPS intra-class scheduling algorithm, the same set of parameters is used as used for analyzing the BE algorithm. However, a new parameter δ

is introduced to account for the deadlines of rtPS packets. We define δ_i be the maximum tolerable latency for connection i . That is, if a packet k of i arrives at time a_k , then it must be scheduled between a_k and $a_k + \delta_i$ to meet the latency constraint. If the packet is not transmitted before $a_k + \delta_i$, then it is considered to be expired and therefore it is dropped from the queue. The constraint of deadline makes the analysis considerably difficult than the classes analyzed earlier.

Let $|\Delta_{rtPS}|$ be the number of rtPS connections and Θ_{rtPS} be the average bandwidth available for the rtPS class, i.e. $\Theta_{rtPS} = \beta - \sum_{i \in \Delta_{UGS} \cup \Delta_{ertPS}} \alpha_i^{min} - \sum_{j \in \Delta_{nrtPS}} \alpha_j^{min} - \beta_{BE}$. The algorithm assures fairness of resource allocation by using the Equations 3.10 and 3.11. This allocation scheme implies that the average service rate of connection i can be given by Equation 3.51.

Following the reasoning of Theorem 2, μ_i can be defined as

$$\mu_i = \frac{\lambda_i(\beta - \beta^u)}{\sum_{j \in \Delta_{rtPS}} \alpha_j^{avg}} \quad (3.51)$$

To make the analysis tractable, we assume that each queue can be analyzed separately with an average arrival rate of λ_i and average service rate of μ_i . Using equation 3.51, the values of T_i^s and ρ_i can be calculated as follows.

$$T_i^s = \frac{1}{\mu_i}$$

$$T_i^s = \frac{\sum_{j \in \Delta_{rtPS}} \alpha_j^{avg}}{\lambda_i(\beta - \beta^u)} \quad (3.52)$$

Similarly,

$$\rho_i = \lambda_i T_i^s$$

$$\rho_i = \frac{\sum_{j \in \Delta_{rtPS}} \alpha_j^{avg}}{\beta - \beta^u} \quad (3.53)$$

The arrival rate of each connection obeys the Poisson distribution, the average service time follows exponential distribution, and the deadline is deterministic. Therefore, each rtPS queue can be analyzed as an M/M/1+D system. This class of queues were analyzed by D.Y Barrer [29]. The most important parameter for the rtPS class is maximum tolerable latency. Specifically, we are interested in knowing the number of packets dropped due to expiry of deadline. According to D.Y. Barrer, the packet loss probability (Q) under statistical equilibrium can be computed by Equation 3.54.

$$Q_i = \frac{(1 - \rho_i)e^{\mu_i \delta_i (\rho_i - 1)}}{1 - \rho_i e^{\mu_i \delta_i (\rho_i - 1)}} \quad (3.54)$$

Substituting the values of ρ_i in Equation 3.54

$$Q_i = \frac{\left(\beta - \beta^u - \sum_{j \in \Delta_{rtPS}} \alpha_j^{avg} \right) e^{\left(\frac{\lambda_i \delta_i \left(\sum_{j \in \Delta_{rtPS}} \alpha_j^{avg} - \beta + \beta^u \right)}{\sum_{j \in \Delta_{rtPS}} \alpha_j^{avg}} \right)}}{\beta - \beta^u - \sum_{j \in \Delta_{rtPS}} \alpha_j^{avg} e^{\left(\frac{\lambda_i \delta_i \left(\sum_{j \in \Delta_{rtPS}} \alpha_j^{avg} - \beta + \beta^u \right)}{\sum_{j \in \Delta_{rtPS}} \alpha_j^{avg}} \right)}} \quad (3.55)$$

Since a packet is dropped if its waiting time exceeds δ_i , therefore T_i^r is always between 0 and δ_i , Mathematically, $0 < T_i^r \leq \delta_i$. Since λ_i is the arrival rate and Q_i is the loss probability, therefore the average packet loss rate (ϖ_i) is the product of λ_i and Q_i and the throughput is equal to $\lambda_i l_i (1 - Q_i)$

$$\varpi_i = \lambda_i Q_i \quad (3.56)$$

and the total packet drop rate for the entire rtPS class can be given as

$$\varpi = \sum_{j \in \Delta_{rtPS}} (\lambda_j Q_j) \quad (3.57)$$

The average throughput (τ_i) of connection i is given by equation 3.58

$$\tau_i = \lambda_i l_i (1 - Q_i) \quad (3.58)$$

and the average throughput for the entire rtPS class can be determined by Equation 3.59.

$$\Phi = \sum_{j \in \Delta_{rtPS}} \lambda_j l_j (1 - Q_j) \quad (3.59)$$

The probability that the average queue size is equal to κ packets at statistical equilibrium can be determined by Equation 3.60.

$$P_i(\kappa) = \lambda_i P_i(0) \prod_{j=1}^{\kappa} (\mu_i + C_i(j))^{-1} \quad (3.60)$$

Where,

$$P_i(0) = \frac{1}{\mu_i T_i^s + 1}$$

and

$$C_i(j) = \frac{\mu_i (\mu_i \delta_i)^{j-1} e^{\mu_i \delta_i}}{\int_0^{\mu_i \delta_i} t^{j-1} e^{-t} dt}$$

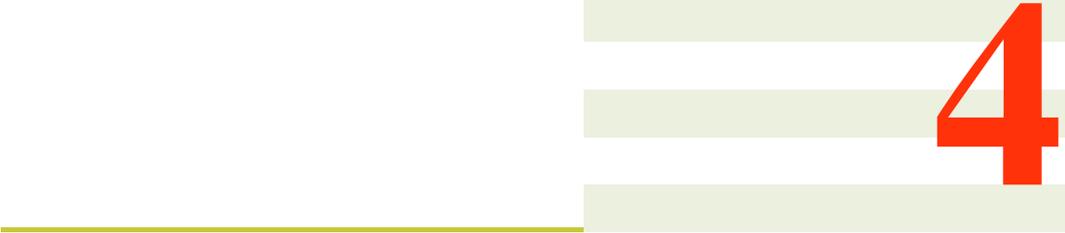
3.7 Conclusion

In this chapter, we proposed connection admission control and packet scheduling algorithms for the base station uplink scheduler. We also provided the details of complementary modules that are required for proper functioning of connection admission control and packet schedulers.

The proposed connection admission control is based on the idea of bandwidth stealing without violating guarantees on minimum throughput and maximum permissible delay. The algorithm admits a new connection if following conditions are satisfied: (i) Maximum delay and minimum throughput guarantees could be provided to the new connection (ii) Maximum delay and minimum throughput guarantees of the established connections are not violated.

The uplink packet scheduling algorithm consists of two levels. In the first level, an inter-class scheduling algorithm distributes bandwidth among various service classes according to their QoS and bandwidth requirements. It ensures that enough resources are provided to each service class to furnish appropriate service to each connection. Then in the second level, class-specific algorithms are used to distribute bandwidth among the connections of a given class.

We also introduced a parameter, called Service Ratio, to ensure fair distribution of resources among rtPS connections. Maximum tolerable latency of rtPS connections is used as the criteria for determining polling-order of subscriber stations. rtPS class-specific algorithm considers packet deadlines and fairness in making scheduling decisions. The nrtPS intra-class scheduling algorithm ensures the minimum traffic rate for each connection, while giving more transmission opportunities to needy connections. The bandwidth allocation for nrtPS class is based on the queue size of each connection. While, the BE class-specific algorithm distributes the available time-slots equally among BE connections. We also provided a detailed queuing model based on Markov chains that could be used to analyze the two-level scheduling algorithm.



4

Simulation Analysis

In this chapter, we provide the simulation analysis of Two-level scheduling algorithm. In the broadest sense, simulation is a tool to evaluate the performance of a system, existing or proposed, under different configurations of interest and over long periods of realtime [128]. It is very useful as a design validation tool for a new system. It helps designers in avoiding unforeseen problems and thus designing a system that better provides the desired performance levels. Simulators can be classified as either continuous or discrete event. Continuous simulators, such as flight simulators, try to model the changes in a system continuously in response to controls. While, discrete event simulators assume that certain events occur at discrete points in time and the system responds instantaneously to these events, often by changing its state. These simulators are much easier to design and operate than continuous simulators and therefore they are of special interest to network researchers.

The main stages in simulation analysis are model development, experiment design, simulation running, output analysis, conclusion formulation, and decision making. All of these stages except execution of simulation are human dependent. A complete simulation experiment usually consists of a series of tests in which changes are made to the input variables and the system behavior is observed. Network researchers should test new protocols and algorithms under varied conditions to determine their robustness and reliability. These experiments enable researchers to analyze new protocols and algorithms under varying network conditions. Thus, they are critical for the better understanding of behavior and characteristics of these systems.

The most important part of simulation analysis is the development of system model. A model is a mathematical representation of the structure and working of a real-world system [128]. A model can be used to simulate many configurations, which is practically impossible to do by using the real-world system it represents. By studying the operation of the model, the behavior of the real system can be inferred. Mathematical models can be classified as either deterministic or stochastic. In deterministic models, the input and output variables are fixed. While in stochastic models, at least one of the variables is probabilistic. Another way of classification is static vs dynamic. Static models do not take time into account, while dynamic models consider the time-varying interactions among various variables.

A good model is simpler than the system it represents yet it is a close approximation of the real system. Thus, it enables an analyst to observe the expected behavior of the system under given input to a good level of accuracy. A good model is an appropriate compromise between realism and simplicity [41]. In fact, a simulation is as good as the underlying model. Therefore, a faithful model of the IEEE 802.16 standard was the most important criteria in choosing Qualnet to validate our results.

A model must be validated extensively before its application. Model validation

is a strenuous process and involves observing model output under a broad range of input [41]. Due to time-consuming and complicated nature of model development and validation, we decided to use a commercial simulator and focus ourselves on the actual problem instead of developing our own simulation environment. Commercial network simulators provide a good opportunity for efficient experimentation. A well-tested commercial simulator provides many benefits, including [41]:

1. Validated implementation of existing protocols
2. A rich infrastructure for developing new protocols
3. The opportunity to study large-scale protocol interaction
4. Easier comparison due to standardized set of result

We found that Qualnet provides a faithful model of the 802.16 standard. Furthermore, it provides a user-friendly graphical user-interface and analytical tools. Therefore, we chose Qualnet to perform simulation analysis of our proposed algorithms.

Alternatives to simulation analysis are laboratory setups and testbeds. Experiments performed on these setups often capture details that might be missed in simulation. However, they have some major drawbacks. Building these experimental setups is very expensive and later on modifying them is even more difficult. Furthermore, reproducing some real-world phenomena can be difficult which complicates system validation.

Discrete-event Simulators The main tool used to analyze the behavior and characteristics of communication networks is discrete-event simulator. In discrete-event simulation, the operation of a system is modeled as a chronological sequence of discrete events. Each event occurs at a particular instant in time, which cause an immediate change in the state of the system [161]. The system state is assumed to be constant between the occurrence of consecutive events. Therefore, the simulator can directly jump to the next event in time. This allows discrete-event simulations to run much faster than the original process being simulated. Due to increasing maturity and ease of use, the use of discrete-event simulation has grown in recent years [76].

A discrete-event simulator consists of following components:

Clock A simulator keeps track of simulation time, which is usually independent of the system time on which the simulator runs. The clock is incremented in discrete steps.

Event Queue It is the list of events waiting to be processed by the simulator. The list is usually maintained as a priority queue, sorted by event occurrence time [88].

Pseudorandom-Number Generator The simulator needs to generate various random variables according to the system model. These variables often mimic some real world phenomena, such as user mobility and noise.

Statistics The simulator keeps track of various parameters of interest.

Condition It is the condition which specifies when the simulation will end. For e.g. at time t , or after processing n events etc.

The main loop of a discrete-event simulation continually operates until the end of simulation condition becomes true. The loop usually consists of following four steps.

1. Removes the event with the earliest occurrence time from the event queue.
2. Sets the Clock to the event time of the event removed in step 1.
3. Process the event and gather statistics of interest
4. If end condition is false go to step 1.

4.1 Introduction to Qualnet

QualNet is a commercial version of GloMoSim [27] developed by Scalable Network Technologies (SNT). It is a comprehensive suite of tools to model large-scale wired and wireless networks. It uses both simulation and emulation to predict the behavior and characteristics of communication systems. Qualnet enables user to perform following tasks [187]:

- Design new protocol models
- Optimize new and existing models
- Design large wired and wireless networks using pre-configure or user-designed models
- Analyze the performance of networks and perform *what-if* analysis to optimize them

The key features of Qualnet are summarized below [187]:

Speed It supports both real-time speed for network emulation and faster speed for running multiple simulations by varying the configuration.

Scalability It can run on cluster, multi-core, and multi-processor systems to model thousands of nodes with high fidelity.

Model Fidelity It uses detailed standard-based implementation of protocol models for accurate modeling of real networks.

Portability It runs on various platforms, including Microsoft Windows , MAX OS, and Linux operating systems.

Extensibility It can connect to other hardware and software applications to greatly enhance the value of the network model.

4.1.1 Components of Qualnet

Qualnet has three core components [186]. A brief description of these components is provided below.

Simulator Qualnet simulator is an scalable simulation engine for large-scale, heterogeneous networks and distributed applications that run on those networks. It makes efficient use of resources and models large-scale networks with tens of thousands of nodes with high fidelity. The main features of the simulator are listed below [186]:

- Fast model set up with a powerful Graphical User Interface (GUI) for custom code development and reporting options
- Instant playback of simulation results to minimize unnecessary model executions
- Fast simulation results for thorough exploration of model parameters
- Scalable up to tens of thousands of nodes
- Real-time simulation for man-in-the-loop and hardware-in-the-loop models
- Multi-platform support

Architect Qualnet Architect provides a user-friendly GUI to setup the simulation scenario and execute it. It has two modes: Design mode and Visualize mode. In Design mode, the GUI is used to design simulation experiments, including geographical distribution of nodes, physical connections, and the functional parameters of network elements. While in Visualize mode, Architect is used to execute and animate experiments created in Design mode.

Analyzer Qualnet Analyzer is a statistical graphing tool that displays network statistics generated from a simulation. Using Analyzer, a user can view statistics as they are being generated, as well as compare results from different experiments.

4.1.2 Qualnet Architecture

The structure of Qualnet is shown in Figure 4.1. It consists of a kernel, several model libraries, graphical user interface, command-line interface, and a number of external interfaces.

Kernel Qualnet kernel is a proprietary, parallel discrete-event scheduler. It is capable of simulating large-scale networks consisting of thousands of nodes with high-fidelity models on a variety of platforms. Qualnet users do not directly interact with the kernel, but use Qualnet API to access the kernel functionality.

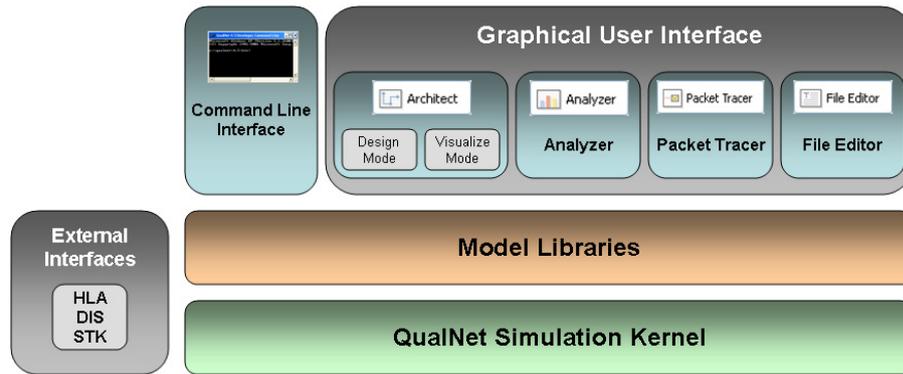


Figure 4.1: Qualnet Architecture [187]

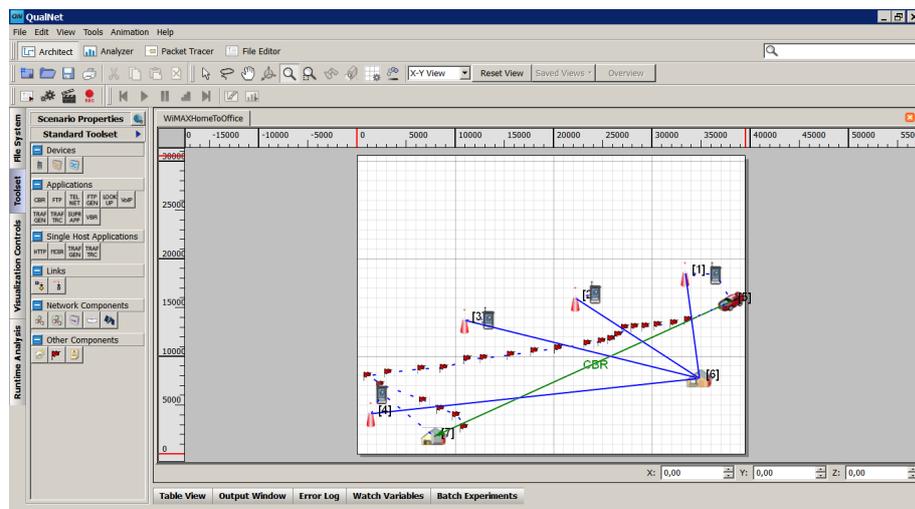


Figure 4.2: Qualnet Graphical User Interface Architect

Model Libraries Qualnet provides several libraries that enable network researchers to design networks using real-world protocol models. Some of the important libraries provided by Qualnet include Developer, Wireless, and Multimedia, Enterprise, Advanced Wireless, network security, sensor network, satellite, and cellular Model libraries.

Graphical User Interface Qualnet GUI consists of four components [187]: Architect, Analyzer, Packet Tracer, and File Editor. These components are directly accessible from the GUI.

Architect Architect is a network design and visualization tool. It operates in two modes: Design mode and Visualize mode. Qualnet GUI Architect is shown in Figure 4.2.

Design Mode In design mode, the users can setup configuration of the experiment, including terrain, network connections, subnets, mobility patterns, and other functional parameters of network nodes. Moreover, the users can customize the protocol stack of any of the nodes, and the ap-

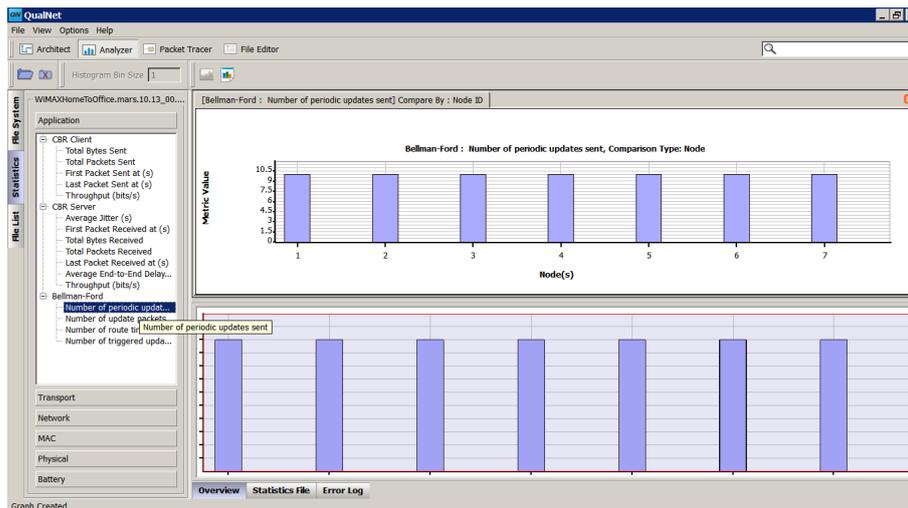


Figure 4.3: Qualnet Graphical User Interface Analyzer

application layer traffic and services that run on it.

Visualize Mode Visualize mode enables in-depth visualization and analysis of an experiment designed in Design mode. Users can watch packets at various layers flow through the network and the dynamic graphs of various performance metrics during the simulation. Users can also perform *what-if* analysis by specifying a range of values of a particular protocol parameter and comparing the performance results for each of them.

Analyzer Analyzer is a statistical graphing tool that displays the metrics collected during the simulation of a network scenario in a graphical format. Users can customize the graph display. All statistics are exportable to spreadsheets in CSV format. Qualnet Analyzer is shown in Figure 4.3.

Packet Tracer Packet Tracer provides a visual representation of packet trace files generated during the simulation of a network scenario. Trace files are text files in XML format that contain information about packets as they move up and down the protocol stack.

File Editor It is a text editing tool that displays the contents of the selected file in text format and allows the user to edit files.

Command-line Interface The command-line interface of Qualnet allows users to execute Qualnet from a command prompt (in Microsoft Windows) or from a command window (in Linux or MAC OS). To execute Qualnet from command-line, input is provided in the form of text files that could be created and modified using any text editor. Building and running scenarios from command-line takes less memory and scenarios typically run faster than with the GUI. Furthermore, the users have the flexibility to interface with visualization and analysis tools of their choice.

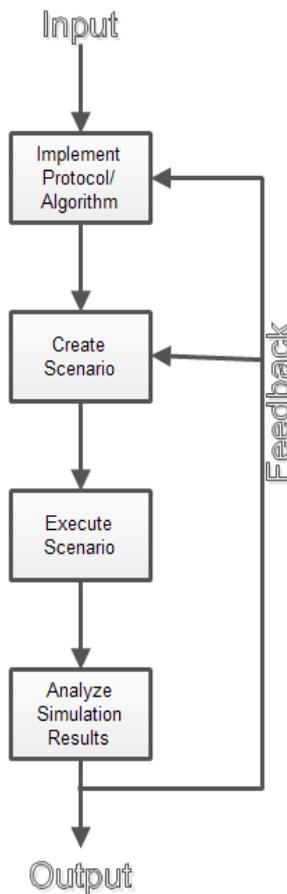


Figure 4.4: Simulation Procedure in Qualnet

External Interfaces Qualnet can also interact with several external tools in real-time. The HLA/DIS module, which is a part of the Standard Interfaces Model Library, allows Qualnet to interact with other HLA/DIS compliant simulators and computer-generated force (CGF) tools like OTB. The Qualnet STK interface, which is a part of the Developer Model Library, provides a way to interface Qualnet with the Satellite Toolkit (STK) developed by Analytical Graphics, Inc. (AGI) and function in a client-server environment.

4.1.3 Simulation Steps

The simulation in Qualnet is scenario based. A scenario refers to a specific network topology and all the network components and conditions under which the network will operate. In general, a simulation experiment comprises the following phases:

- The first phase is to design and implement the protocol or algorithm to be tested at an appropriate layer. This include developing a protocol library and modifying several common files to integrate the new functionality. The step must be performed carefully to get accurate results. Qualnet must be

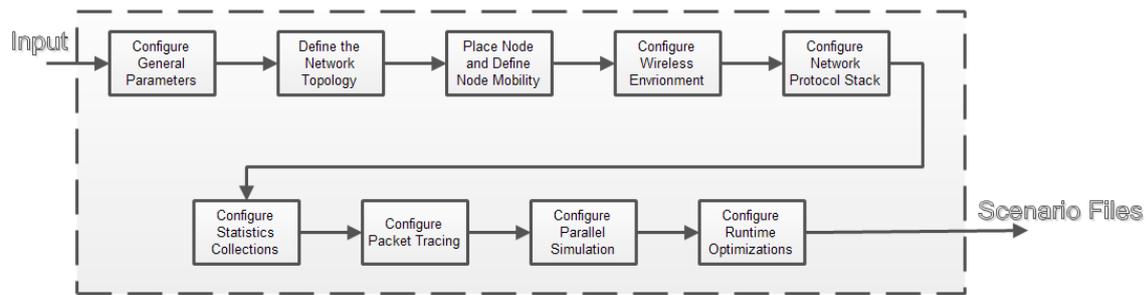


Figure 4.5: Scenario Creation in Qualnet

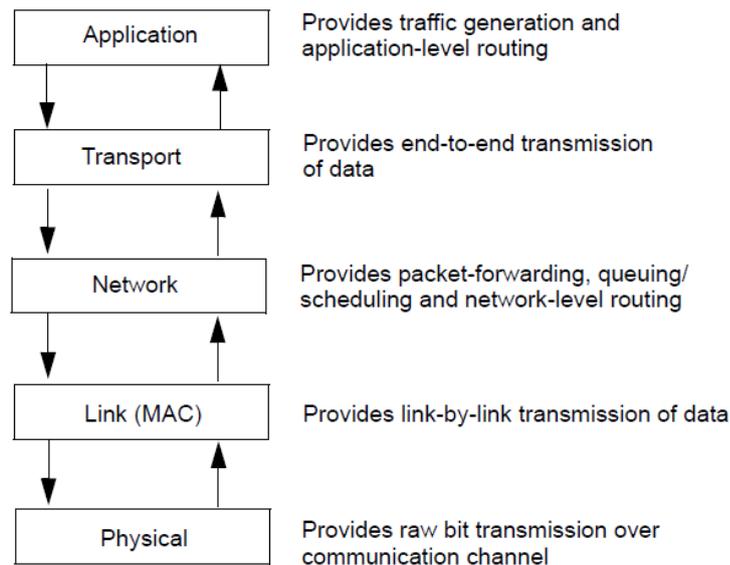


Figure 4.6: Qualnet protocol stack

re-compiled after this step.

- The next phase is to create and prepare the simulation scenario according to the system description and the performance metrics that are to be observed. The steps involved in a scenario creation are shown in Figure 4.5.
- The third phase, consists of executing, visualizing, and analyzing the created scenario and then collecting simulation results. The results include animations, runtime statistics, final statistics, and trace files.
- The last phase is to analyze the results obtained in the third phase. The analysis can be done by Qualnet Analyzer. However, if Qualnet Analyzer does not provide the required analysis, then the user must develop customized tools to parse the output files and present the results in the required format.

4.2 Implementation of TLSA in Qualnet

The protocol stack used by Qualnet is similar to the TCP/IP network protocol stack. It consists of Application, Transport, Network, Link (MAC), and Physical

layers. Adjacent layers in the stack communicate via APIs, and generally layer communication occurs between adjacent layers only. The Qualnet protocol stack is shown in Figure 4.6.

TLSA is a scheduling algorithm for the base station uplink scheduler. In WiMAX, QoS mechanisms, including packet scheduling is implemented at the MAC layer. More specifically, packet scheduling is done in Common-Part sublayer of the MAC layer. In Qualnet, the support of WiMAX networks is provided through Advanced Wireless Model library [188]. The library provides an extensive set of customizable parameters and a faithful implementation of the 802.16 standard. The source code of the library is available and the researchers can modify the code to implement new algorithms and protocols.

The library includes the specifications of IEEE 802.16d-2004 and IEEE 802.16e-2005 standards. The library provides a rich set of MAC features for both the base station and subscriber stations. It implements a strict priority-based scheduling for different service types where UGS > ertPS > rtPS > nrtPS > BE. Within each service class, the bandwidth distribution is done by using WFQ algorithm.

The uplink scheduling is done at the base station. Therefore, TLSA is implemented at Common-Part sublayer of the base station. The strict priority and WFQ based scheduling is implemented in Function *MacDot16ScheduleUlSubframe* in files “mac_dot16_sch.h” and “mac_dot16_sch.cpp”. The function is modified to replace strict priority and WFQ algorithms by the inter-class and intra-class algorithms of TLSA, respectively. The subscriber station scheduler provided by Qualnet does not drop packets that cross the deadlines. Therefore, code is added in “mac_dot16_ss.cpp” to drop the expired packets from data queues.

Simulation Parameters The performance delivered by a WiMAX network is strongly dependent upon the working of the MAC and physical layers. Qualnet provides all the mandatory features as specified by the IEEE 802.16 standard. The physical layer implemented in Advance Wireless Library supports OFDMA for maximum accuracy. It supports various modulation schemes such as QPSK, 16-QAM, and 64-QAM with convolutional coding at various encoding rates. It also provides variable channel bandwidth, variable FFT sizes, and multiple cyclic prefix codes. The data rate of an OFDMA channel is a function of channel bandwidth, FFT sizes, cyclic prefix time, sampling factor, modulation scheme, and encoding rate. The raw data transfer rate of an OFDMA channel can be obtained by Equation 4.1.

$$data\ rate = \frac{N \times b \times c}{T} \quad (4.1)$$

where,

FFT Size	Number of Sub-carriers
2048	1440
1024	720
512	360
128	72

Table 4.1: FFT sizes and correspond number of sub-carriers implemented in Qualnet Advanced Wireless Model library

Modulation Scheme	Number of Bits per Modulation Symbol
QPSK	2
16-QAM	4
64-QAM	6

Table 4.2: Number of bits per symbol as function of modulation scheme

N = Number of sub-carriers used

b = Number of bits per modulation symbol

c = Coding rate

T = OFDMA symbol duration

The number of sub-carriers used (N) depends upon the FFT size being used. Table 4.1 lists the values of N and corresponding FFT sizes implemented in Qualnet.

The number of bits per modulation scheme (b) depends upon the modulation scheme being used. Table 4.2 lists modulation schemes supported by Qualnet Advanced Model library and corresponding values of b .

For the simulation experiments, the total uplink channel capacity was set to 1 Mbps. However, for simulations of rtPS applications the uplink bandwidth was set to 10Mbps. Two-ray ground reflection was used as radio propagation model. The model considers direct and ground reflected paths. With Two-ray ground reflection, the signal strength S at the receiver can be obtained by Equation 4.2.

$$S = \frac{PGG'H^2H'^2}{d^4L} \quad (4.2)$$

where,

S = signal strength at the receiver

P = transmitted power

G = antenna gain of the transmitter

G' = antenna gain of the receiver

H = height of transmit antenna

H' = height of receive antenna

d = distance between transmitter and receiver

L = system losses

The constant shadowing model with a mean of 4.0 dB was used, and the transmission power was set to 20 dBm. For each simulation, data transmission was set to start at 20s. This delay is necessary for proper functioning of routing protocols. Since the actual transmission starts at 20s, so we consider this as the starting point of each simulation i.e. $t = 0s$. For each repetition, a different pseudo-random seed was used to alter the characteristics of simulation such as traffic generation patterns, back-off timers, interference levels, and mobility patterns. The values of important parameters used for simulation are presented in Table 4.3.

Performance Metrics The description of various performance metrics used to analyze the performance of the proposed scheduling scheme is given below:

- **Throughput:** The total units of data transmitted in duration Δt divided by Δt
- **End-to-end Delay:** The average delay observed by packets from source to destination. It includes queuing delay and propagation delay.
- **Packet Loss Ratio:** The number of lost packets to the total number of packets sent by the sender
- **Fairness:** The fairness of resource allocation is determined by Equations 3.10 and 3.11.
- **Scalability:** The change in performance of algorithm as the number of subscriber stations increases

4.2.1 Simulation Analysis of Inter-Class Scheduling Algorithm

Bandwidth Allocation

The purpose of this experiment is to assess the performance of inter-class scheduling algorithm. For the simulation, BE traffic is generated at an average rate of 200 Kbps throughout the experiment. The value of β_{BE} is set to 90 Kbps to prevent starvation of BE class. While for nrtPS class, the MRTR is 375 Kbps and the average traffic rate is 500 Kbps. Simulations are performed with increasing load of rtPS traffic. Initially, the average traffic rate of rtPS class is 300 Kbps, which is gradually increased to 600 Kbps. The MRTR for rtPS traffic is 300 Kbps throughout the experiment, while the maximum permissible delay is set to 160 ms.

The bandwidth distribution by the algorithm is shown in Figure 4.7. As the data generation rate of rtPS class is increased from 300 Kbps to 400 Kbps, the throughput of BE traffic is reduced from 200 Kbps to approximately 90 Kbps. While there is no effect on the throughput of nrtPS class. As the rtPS data rate is further increased, the throughput of nrtPS decreases. Since 90 Kbps is the reserved bandwidth for BE class, therefore the throughput of BE traffic cannot be further reduced and remains

Parameter	Value
Total uplink bandwidth	1 Mbps
Frame duration	20 ms
TDD downlink duration	10 ms
MAC propagation delay	1 μ s
Cyclic prefix	8.0
Input queue size	50000 bytes
Antenna model	omni antenna
Sampling factor	8/7
Propagation model	Two ray ground
Shadowing model	constant
Shadowing mean	4.0 dB
Temperature	290K
Noise factor	10.0
Service flow timeout	15 s
Transmit power	20 dBm
Receive power threshold	205e-12
Carrier sense power threshold	0.9 * Receive power threshold
Handover RSS trigger	-78.0 dBm
Propagation limit	-111.0 dBm
Link adaptation	Enabled
Packing	Enable

Table 4.3: Important simulation parameters

unaffected. When rtPS data generation rate is increased to 520 Kbps, the throughput of nrtPS reaches its MRTR i.e. 375 Kbps. Further increase in rtPS traffic rate has no effect on the throughput of nrtPS and BE classes. So the throughput of rtPS class cannot be further increased by just increasing its traffic generation rate. It can be seen that the algorithm is able to ensure that rtPS and nrtPS classes get at least their MRTR. In case of overload rtPS class gets priority and the algorithm takes away extra bandwidth from nrtPS and BE classes.

The percentage of lost packets is shown in Figure 4.8. The percentage of lost packets is negligible till the total data generation rate is less than the available uplink bandwidth. Packet loss starts, once the combined bandwidth requirements of all classes exceeds the available bandwidth. This is due to the fact that the rtPS packets that miss their deadlines due to overload are dropped by the subscriber station scheduler.

Simulations are also performed to determine if the inter-class scheduling algorithm is able to meet the deadlines of rtPS traffic. The end-to-end delay observed by different service classes is shown in Figure 4.9. It can be seen that rtPS traffic observed the least delay. In fact, the end-to-end delay of rtPS traffic remains around 30 ms throughout the experiment, while the maximum tolerable latency is 160 ms. Increase in rtPS throughput results in reduced bandwidth allocation to nrtPS and BE classes, which in turn results in relatively higher delays for these classes.

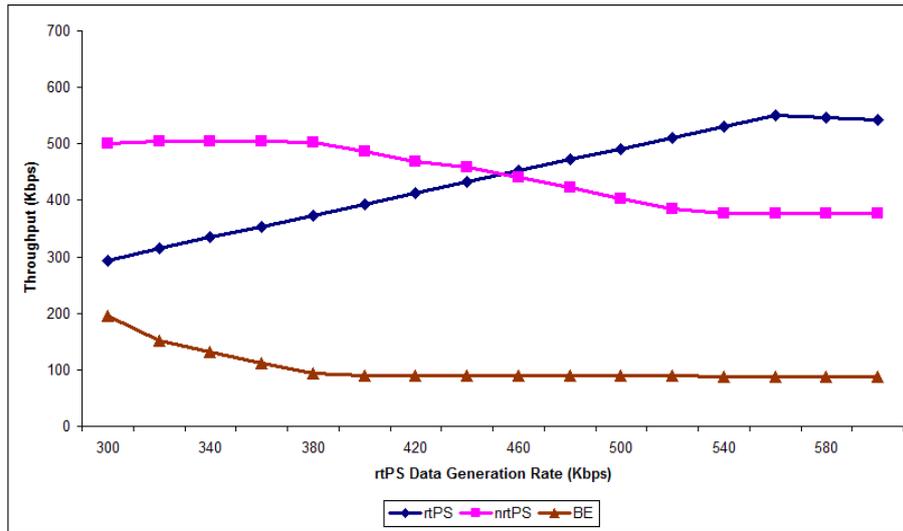


Figure 4.7: Bandwidth distribution by the inter-class scheduling algorithm

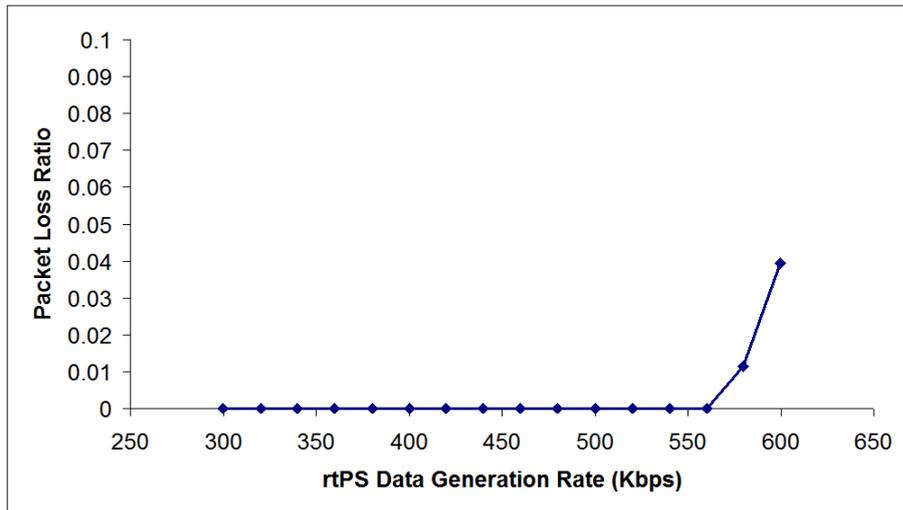


Figure 4.8: Packet loss ratio

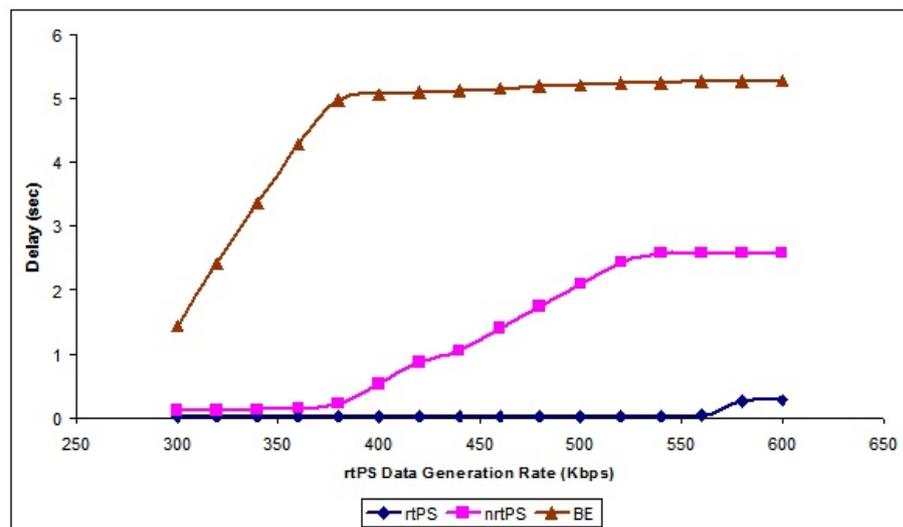


Figure 4.9: End-to-end delay for different service classes

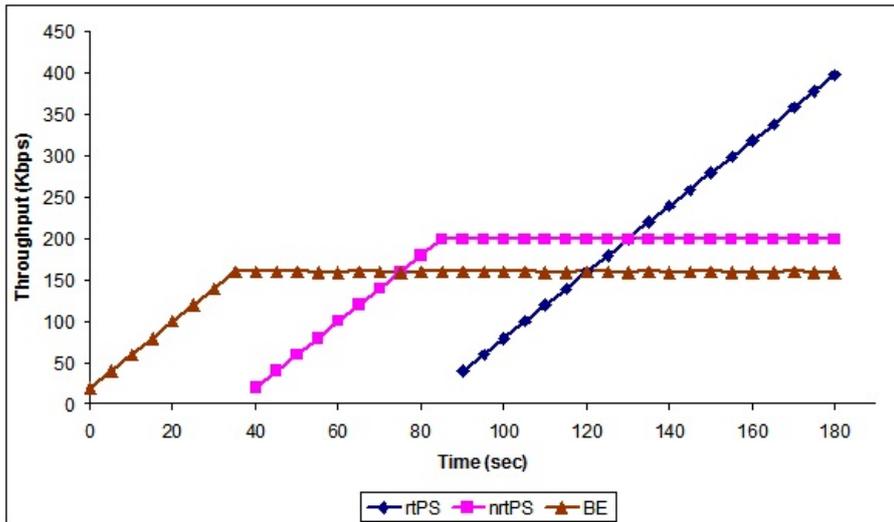


Figure 4.10: Throughput of different classes of traffic for mobile stations

Effect of Mobility

Mobility of subscriber stations can adversely affect the QoS provided by the network. This experiment is designed to determine the effect of mobility on the performance of the inter-class scheduling algorithm. The scenario consists of two base stations and three subscriber stations. Each subscriber station has one connection of each service class (rtPS, nrtPS and BE). The subscriber stations move linearly at a constant speed of 16.67 m/s (60 km/h) and performs one handover during the simulation. At the start of the simulation, only BE traffic is present. The traffic rate of BE traffic is gradually increased from 20 Kbps to 160 Kbps in 40 seconds (0-40 sec). After 40 second, the average rate of BE traffic is kept constant. At 40th second nrtPS traffic is introduced in the network. The rate of nrtPS traffic is gradually increased to 200 Kbps (40-85 sec). Henceforth, the average traffic rate of nrtPS class is kept constant at 200 Kbps. rtPS traffic is introduced at this point and its data generation rate is increased gradually to 400 Kbps (85-180 sec).

The throughput of all service classes at the receiver is shown in Figure 4.10. As the applied load is less than the available bandwidth, therefore the algorithm is able to allocate bandwidth to service classes that exactly matches the input traffic pattern.

The packet loss ratio is shown in Figure 4.11. It can be seen that the ratio remains below 0.0075 for all classes of traffic. Furthermore, the fluctuation is the least in case of rtPS traffic. The packet loss ratio is minimum for rtPS traffic, while maximum for BE traffic. However, the difference is not more than 0.001. It can be seen that under normal load, the introduction of nrtPS and rtPS traffics do not have significant effect on BE traffic.

The end-to-end delay for different classes of traffic is shown in Figure 4.12. The

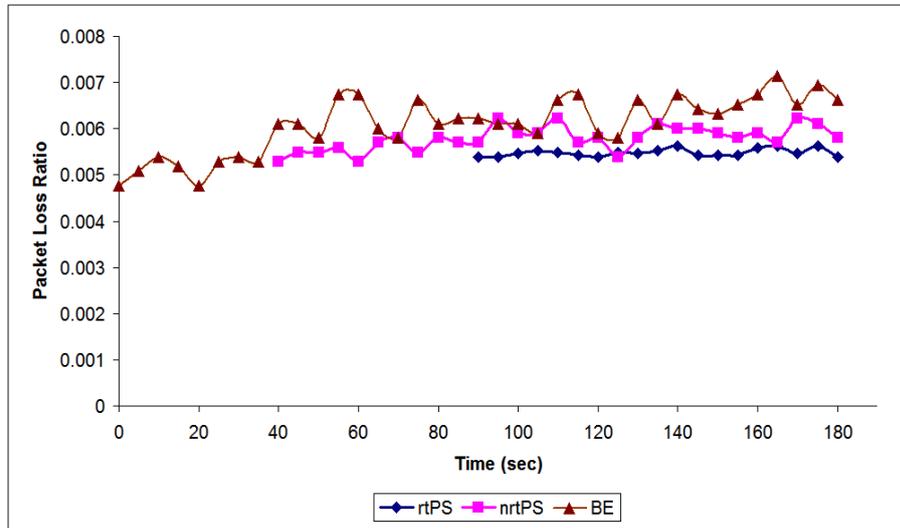


Figure 4.11: Packet loss ratio for mobile stations

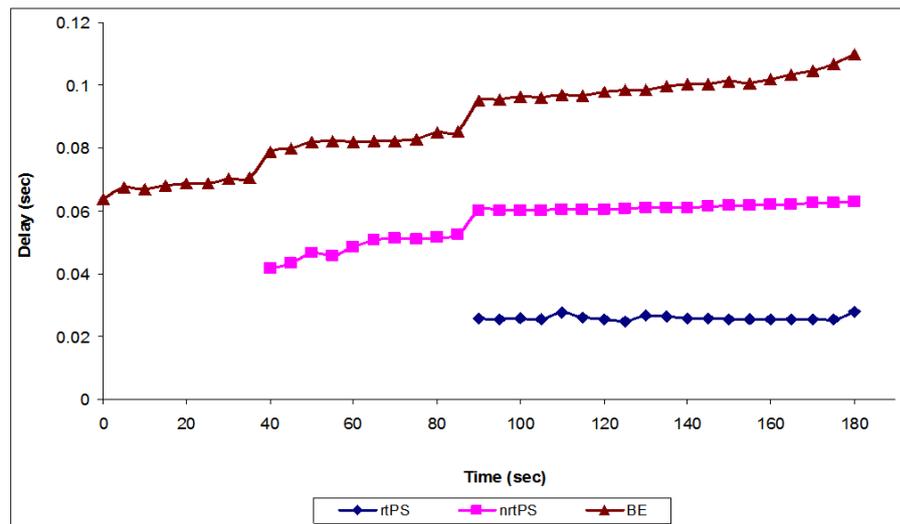


Figure 4.12: Delay in mixed traffic scenario under mobility

Connection	MRTR (Kbps)	Average Traffic Rate (Kbps)
$N1$	140	200
$N2$	200	225
$N3$	225	275
$N4$	250	300
Total	815	1000

Table 4.4: Input Traffic Parameters for nrtPS Connections

introduction of nrtPS traffic increases delay for BE class. Similarly, the introduction of rtPS traffic results in slight increase in delays of nrtPS and BE classes. The delay of rtPS traffic remains constant irrespective of the applied load and is around 25 ms, which is very good for realtime traffic. Also, the delay of nrtPS traffic remains below 70 ms throughout the experiment.

4.2.2 Simulation Analysis of Class Specific Scheduling Algorithms

nrtPS Intra-Class Scheduling Algorithm

The experiment is performed to analyze the performance of nrtPS intra-class scheduling algorithm under high load. For this experiment, the data generation rate is almost equal to the total available uplink bandwidth. Four subscriber stations with one nrtPS connection each are used in the scenario. The parameters of the connections as shown in Table 4.4. Note that the only type of traffic present is nrtPS and the ratio of available bandwidth to the applied load is almost 1.

The corresponding bandwidth allocation is shown in Figure 4.13. It can be seen that throughput of each connection follows the data generation pattern and remains stable throughout the experiment. Furthermore, the MRTR is guaranteed for all nrtPS connections. We also calculated the service ratio (Ψ_i) as defined in Equation 3.10. Ψ_i for connections $N1$, $N2$, $N3$ and $N4$ are approximately 0.99, 0.99, 0.98 and 0.98 respectively.

The end-to-end delays experienced by the four connections are shown in Figure 4.14. Since all the connections get approximately same service ratio, therefore the end-to-end delay is identical for all connections. The applied load is almost equal to the available bandwidth. Therefore, slowly the packets accumulate in the data queues and thus there is a gradual increase in observed delay with the passage of time. The maximum delay is observed by $N4$, which is slightly less than 0.5 seconds.

BE Intra-Class Scheduling Algorithm

In this scenario four subscriber stations with one BE connection each are used to analyze the performance of BE intra-class scheduling algorithm. The average data

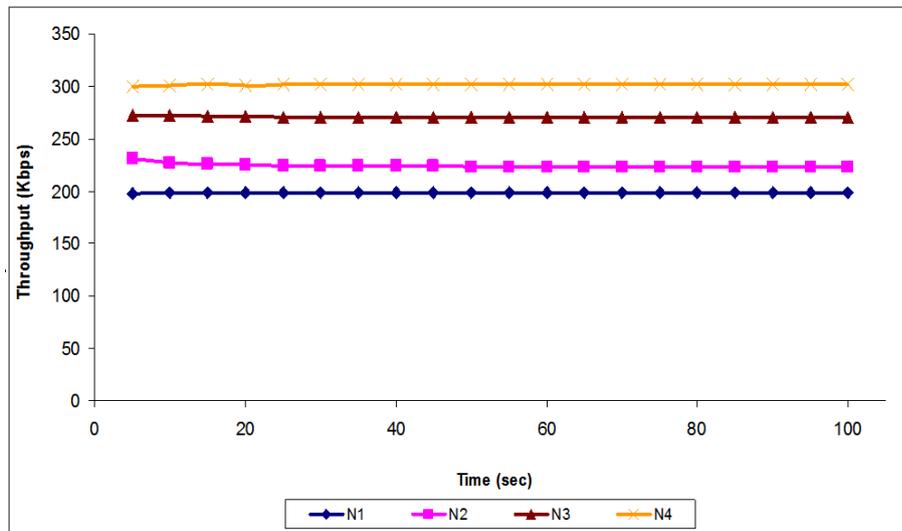


Figure 4.13: Bandwidth allocation by nrtPS class specific algorithm

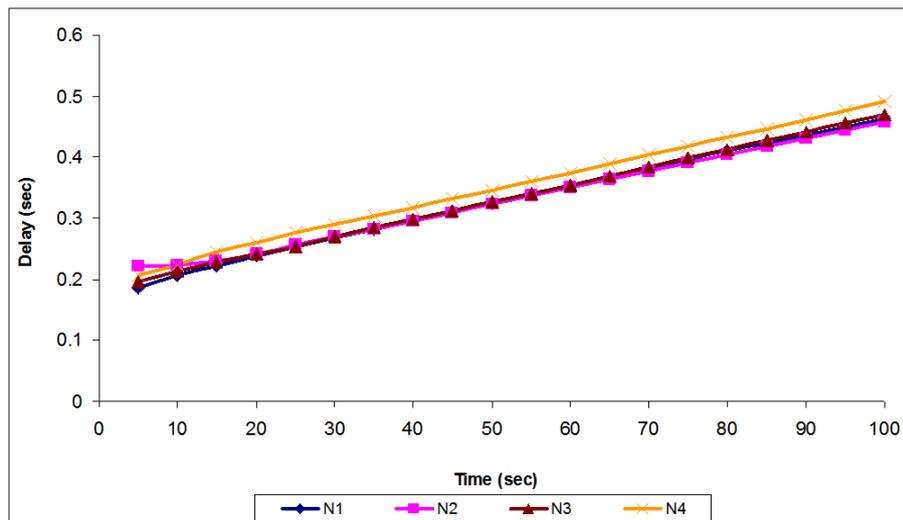


Figure 4.14: End-to-end delay for nrtPS connections

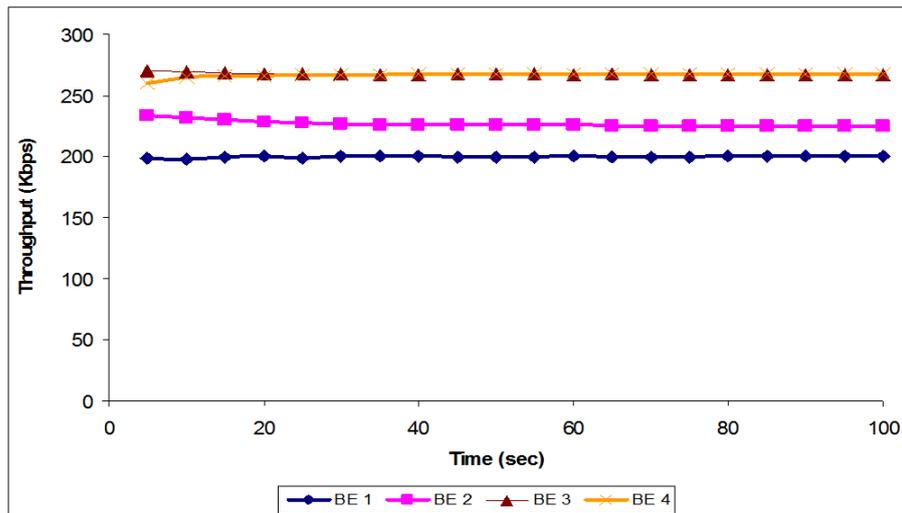


Figure 4.15: Bandwidth allocation by BE class specific algorithm

generation rate is 200 Kbps, 225 Kbps, 275 Kbps, and 300 Kbps for connections *BE1*, *BE2*, *BE3* and *BE4* respectively. Again, the ratio of available bandwidth to applied load is almost 1 and only BE traffic is used for the analysis.

The throughput of the connections is shown in Figure 4.15. The algorithm equally divides the available time-slots among the BE connections. However, the data generation rate of *BE1* and *BE2* is less than the available bandwidth per connection. Therefore, the throughput of *BE1* and *BE2* are equal to their data generation rates. The remaining time-slots are distributed among other two connections.

The end-to-end delay experienced by the connections is shown in Figure 4.16. The delay is almost negligible for *BE1* and *BE2*, while it has the greatest value for *BE4*. Since for *BE1* and *BE2*, the allocated bandwidth is equal to their data generation rate, therefore the input queues remain almost empty and thus the waiting time in the queue is negligible. While, the difference of throughput and data generation rate is maximum for *BE4*. Therefore, more and more packets wait in the input queue with the passage of time and thus the connection has relatively large delays.

rtPS Intra-Class Scheduling Algorithm

Fairness The objective of the experiment is to determine the degree of fairness of resource allocation by rtPS intra-class scheduling algorithm. The same experiment was repeated with EDF algorithm and the comparative results are provided. For the experiment, four subscriber stations hosting one rtPS connection each is used. The parameters of the connections are shown in Table 4.5. These parameters imply a very heavy load on system as the ratio of available bandwidth to data generation rate is less than 0.5.

The service ratio for each connection Ψ_i and the mean service ratio (Ψ') ob-

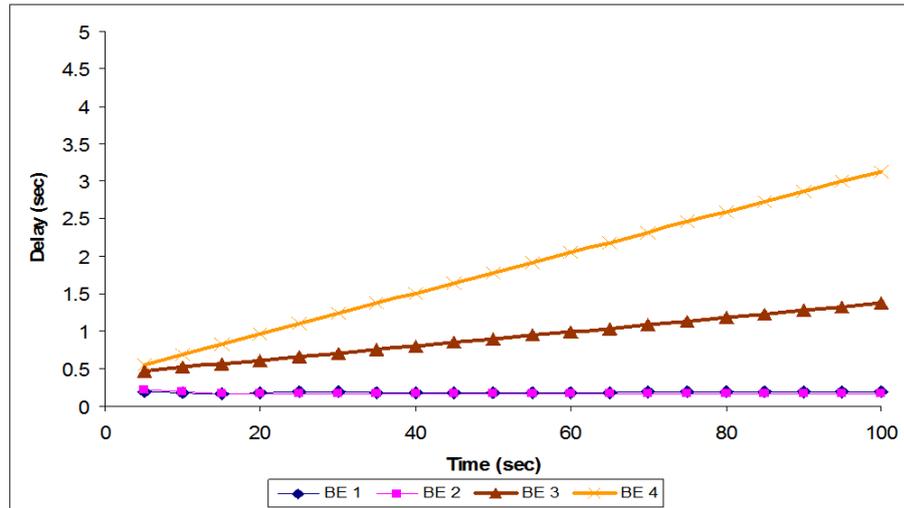


Figure 4.16: End-to-end delay for BE traffic

Connection	MRTR (Kbps)	MSTR (Kbps)	Maximum Tolerable Delay (frames)
A	4000	9000	2
B	1000	3000	3
C	2000	4000	3
D	3000	5000	4

Table 4.5: Input traffic parameters for rtPS connections

tained during the simulation are shown in Figure 4.17. It can be seen that service ratios of all rtPS connections adapt and follow Ψ' . Even though the available bandwidth could only provide minimum guaranteed service to each connection, the proposed algorithm performs very well and dynamically allocates bandwidth to ensure fairness. In fact, Ψ' is the best a connection can get and all the connections seem to closely follow Ψ' . Thus, it shows that the algorithm is able to fairly allocate maximum possible bandwidth to each rtPS connection.

Figure 4.18 shows the service ratios obtained by using EDF on the same set of connections. There is not much difference between Ψ' provided by EDF to that of provided by our proposed algorithm. However, obviously there is greater difference among the Ψ_i of individual connections. In this case, EDF allocates maximum bandwidth to *A*, while least bandwidth is allocated to *B*. This dispersion in *service ratios* is due to the fact that EDF tries to minimize the average delay but does not take fairness into account.

Bandwidth Allocation The aim of this experiment is to determine the behavior of the rtPS intra-class scheduling algorithm under increasing load. The experiment begins with a data generation rate of 5 Mbps. The data generation rate is gradually increased to 20 Mbps and values of various performance metrics are recorded. The

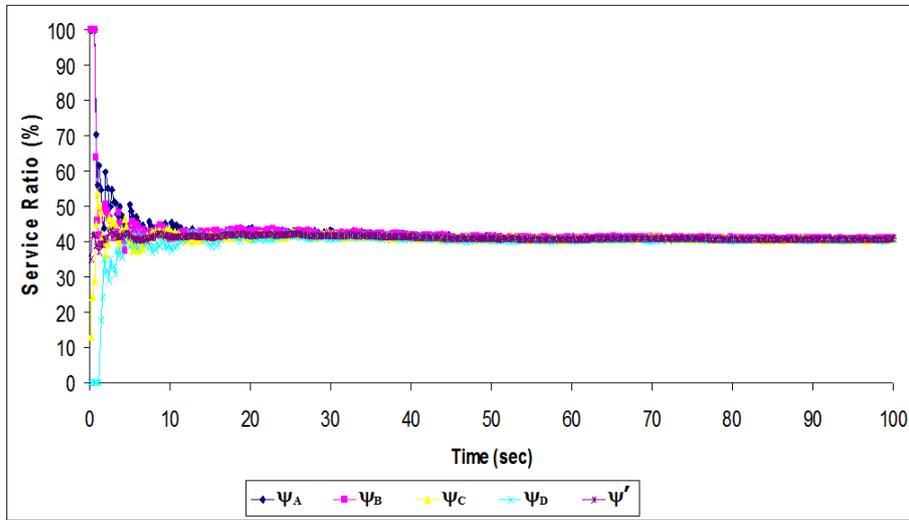


Figure 4.17: Service ratio of rtPS connections by applying the proposed algorithm

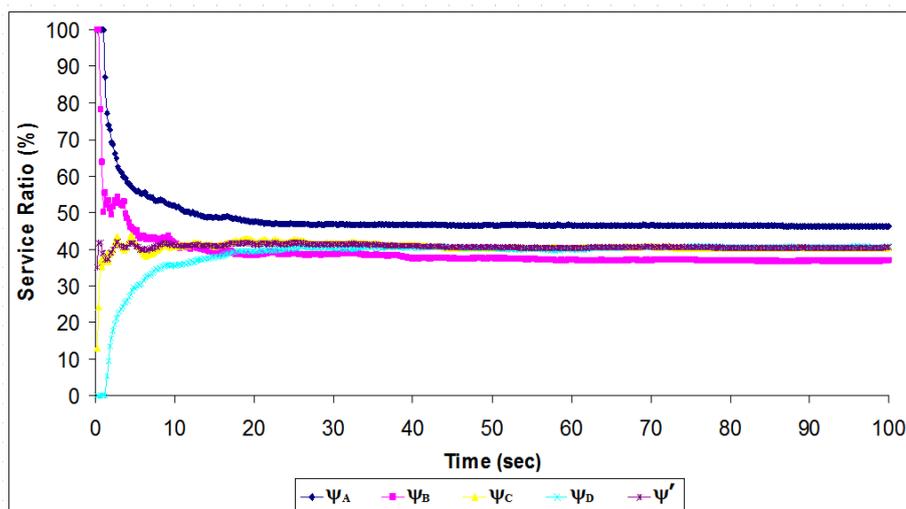


Figure 4.18: Service ratio of rtPS connections by applying EDF

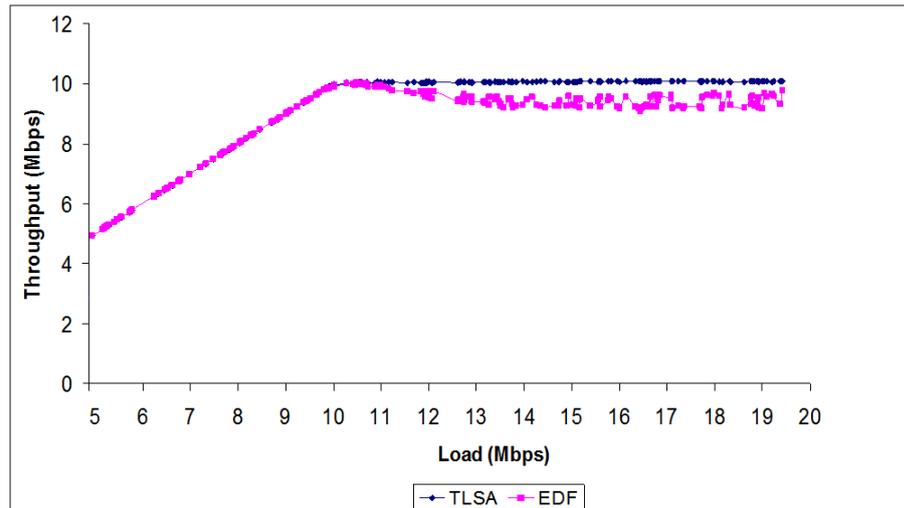


Figure 4.19: Throughput vs applied load for rtPS connections

experiment is performed with both TLSA and EDF, and comparative analyses are provided.

Figure 4.19 shows the throughput as function of load. Clearly both algorithms are able to schedule all traffic until the applied load surpasses the available bandwidth of 10 Mbps. After this point, no matter how much load is applied, the algorithms cannot allocate more bandwidth. Under high load the behavior of EDF is uncertain and it tends to drop some packets and throughput is slightly less than 10 Mbps.

Figure 4.20 represents average delay experienced by packets as function of applied load. Under light and medium load conditions, the packets are scheduled almost immediately by both algorithms. However, under high load the packets have to wait more than the average waiting time. Note that the expired packets are automatically dropped by subscriber station scheduler, which bounds the maximum value of observed delay. This effect is evident in the graph.

Figure 4.21 shows the *packet loss ratio* as function of load. When the applied load is less than 10 Mbps, both algorithms are able to schedule almost all input packets and therefore the *packet loss ratio* is very small. Any traffic above 10 Mbps threshold cannot be scheduled and therefore the ratio increases sharply after this point. It can be seen that at a load of 20 Mbps, half of the traffic is dropped and so the *packet loss ratio* is around 0.5.

Packet loss ratio as function of load under mobility The objective of this experiment is to determine the percentage of lost packets for rtPS class as function of load with mobile stations. The results of the experiment are presented in Figure 4.22. For this simulation, the speed of mobile station is set to 60 Km/h (16.67 m/s) and it performs one handover. Two base stations and one mobile station are

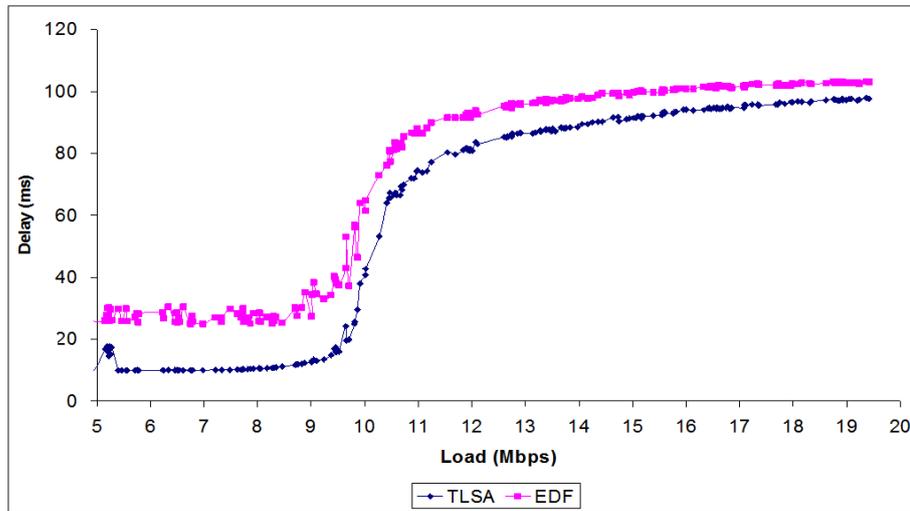


Figure 4.20: Average Delay vs Load

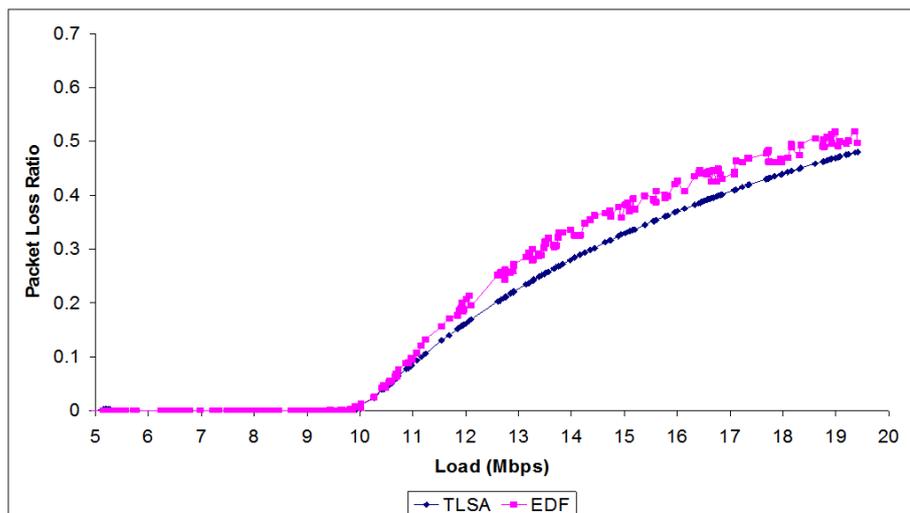


Figure 4.21: Packet loss ratio vs load

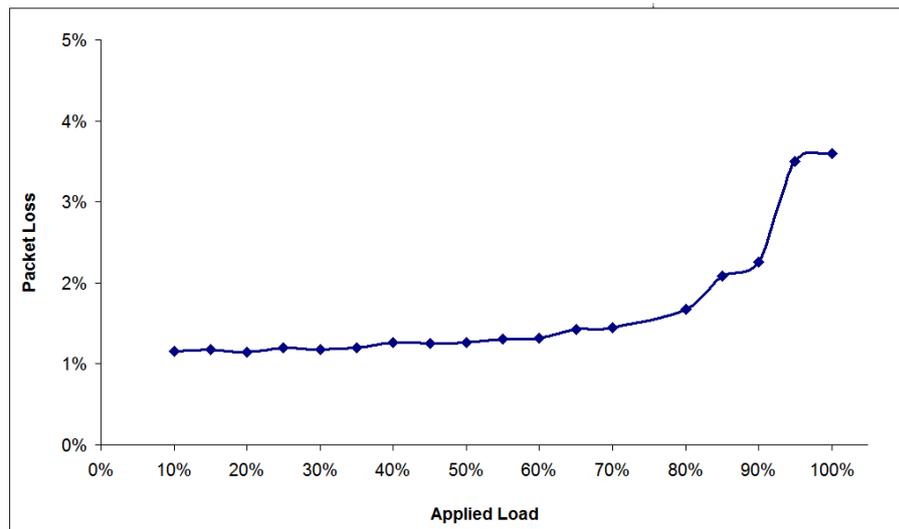


Figure 4.22: Lost packets as function of traffic load under mobility

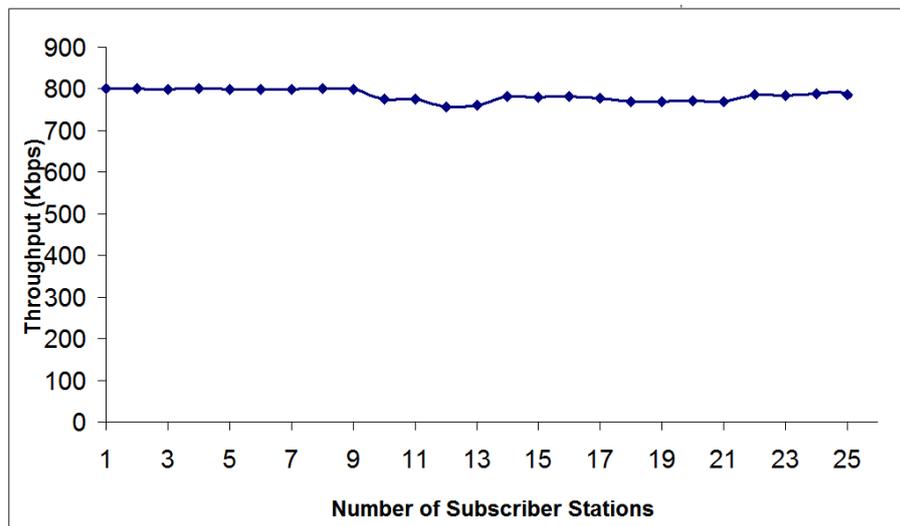


Figure 4.23: Scalability of rtPS class specific algorithm

used in the scenario. Simulations are performed with increasingly more load till the rtPS data generation rate is equal to total available uplink bandwidth. It can be seen that there is little increase in lost packets till the applied load is 80% of the available bandwidth. Further increase in load results in greater packet loss ratio. However, the ratio always remains below 0.04.

Scalability Scalability of a scheduling mechanism is a highly desirable property, especially for the scheduling of delay sensitive traffic such as rtPS. Therefore, this experiment is performed to determine the effect of number of subscriber stations on the performance of rtPS intra-class scheduling algorithm. For this experiment, rtPS traffic is generated at an average rate of 800 Kbps. The experiment is performed with increasing number of subscriber stations. The average throughput achieved versus the number of subscriber stations is shown in Figure 4.23. The result suggests

that the proposed rtPS scheduler remains quite stable with increasing number of subscriber station and hence it is scalable.

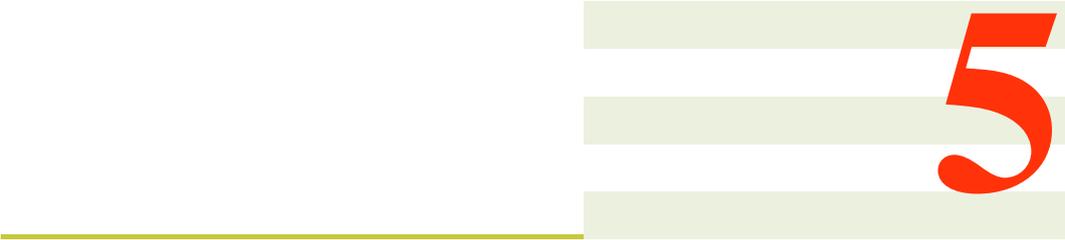
4.3 Conclusion

In this chapter, we provided the simulation analysis of Two-level scheduling algorithm. A good simulation model of the real system being simulated is the most important prerequisite to get accurate results. A well-developed and faithful model of the IEEE 802.16 standard is provided in Qualnet 5.0. Therefore, we chose Qualnet 5.0 as the simulation platform for analyzing the performance of the proposed scheduling algorithm. Since in WiMAX, packet scheduling is done at the MAC layer. Therefore, we implemented TLSA in the MAC layer of the base station.

Several simulation experiments were performed to observe the performance of the inter-class and intra-class scheduling algorithms. The results showed that the inter-class scheduling algorithm effectively ensures QoS and maintain the priority order $rtPS > nrtPS > BE$. In case of insufficient resources, it takes bandwidth from lower priority flows and allocate it to higher priority flows. Moreover, it prevents starvation of lower priority flows.

The analyses of the intra-class scheduling algorithms were performed for each service class. The simulations were performed for these algorithms to observe their behavior under high network load. The results showed that the nrtPS intra-class scheduling algorithm ensures the minimum throughput level for each connection. Furthermore, under statistical equilibrium, the throughput of all connections remain stable and the end-to-end delays are similar.

The simulation experiments of the rtPS intra-class scheduling were repeated with EDF algorithm as well to obtain comparative results. The experiments were performed to observe various performance metrics, including fairness, packet loss ratio, latency, and scalability. The results showed that the proposed algorithm ensures fairness of resource distribution, while EDF algorithm does not take fairness into account and therefore the bandwidth allocation is unfair. Furthermore, the proposed algorithm provided better results for other performance metrics, as compared to the EDF algorithm.



5

Video Transmission Framework



Figure 5.1: An image with high spatial redundancy

Video streaming is the most important application of rtPS scheduling class. With advances in wireless networks, there is a huge increase in wireless video transmission. The usability and flexibility of these systems have enabled several new video services. Some important services include: (1) Video telephony using portable devices (2) Video streaming to mobile phones (3) Interaction of various multimedia entertainment systems in a home (4) Realtime supervision of paramedical staff providing life-support services in disaster recovery operations. Undoubtedly, wireless video services are now well integrated in the society and will continue to pervade.

In this chapter, we provide details of a video transmission framework that is based on the proposed intra-class scheduling algorithm. The framework is compatible with H.264 video coding standard. The objective is to determine the performance of the rtPS intra-class scheduling algorithm in transmitting video streams. The chapter is organized as follows. Section 5.1 gives the basics of video encoding and decoding process. The next section provides a brief overview of the H.264 video coding standard. Then, Section 5.3 discusses the main issues and challenges in H.264 video streaming. The design of video transmission framework is provided in Section 5.6 and the performance analysis of the rtPS intra-class scheduling algorithm is provided in Section 5.6.1. Section 5.7 concludes the chapter.

5.1 Video Coding

Video coding is the process of removing redundant data from a video sequence. Most compression algorithms and codec target temporal and spatial redundancy contained in a video sequence. The majority of algorithms perform lossy compression, which provide a compression of up to factor 200 [205].

Video compression is very important for reducing the size of video sequence so that the sequence could be transmitted over a network link. For example, the throughput provided by most networks is not sufficient to provide realtime transmission of uncompressed videos even at very low frame rates. It is more plausible to transmit several compressed high-resolution videos than a single low-resolution



Figure 5.2: An example of consecutive frames with high temporal redundancy

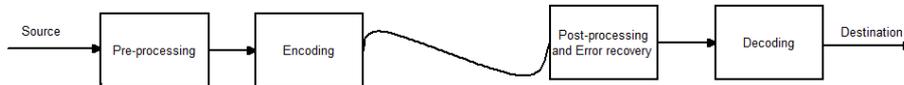


Figure 5.3: Conceptual diagram of an encoder/decoder pair

raw video. Thus, the transmission link is used much more efficiently.

A video sequence consists of a series of still images. The series of images contains two types of redundancies: spatial and temporal. Spatial redundancy refers to the duplication of pixels in an image frame. For example, the image shown in Figure 5.1 contains areas where the variation in pixels is very low and consequently the spatial redundancy is high. While, temporal redundancy refers to the case where two video frames have same pixel values at the same location. Consider the two consecutive image frames from *galleon* video sequence as shown in Figure 5.2. The two images have almost identical pixel values at corresponding locations, thus the frames have high temporal redundancy.

The video compression is performed by an encoder at the sender. The encoded video is then transmitted over the network. A decoder at the receiver reconstruct the video so that it could be viewed by the end-users. Figure 5.3 shows a conceptual presentation of an encoder/decoder system.

5.1.1 Digitization

A video sequence contains a series of images sampled at regular intervals in time. Each image is a rectangular grid of sampled points from the original scene. Therefore to obtain an image of width X and height Y sample points, a total of $X \times Y$ values must be read by the sensor. Each sample point is represented by a set of numbers that describes its brightness and color values [160], called pixel. An image sampled with more sampling points is of higher quality than the same image sampled with fewer sampling points. This effect can be seen in Figure 5.4.



Figure 5.4: (a) Sampled with more sample points (b) Sampled with fewer sample points

Format	Video Resolution
SQCIF	128 × 96
QCIF	176 × 144
SCIF	256 × 192
SIF(525)	352 × 240
CIF/SIF(625)	352 × 288
4SIF(525)	704 × 480
4CIF/4SIF(625)	704 × 576
16CIF	1408 × 1152
DCIF	528 × 384

Table 5.1: Common video formats

The sampling must be done at regular intervals in time to obtain a series of images. These images are then played back at an adequate rate to create the sensation of motion. Higher sampling frequency provides a smoother video sequence at the cost of more processing and storage resources. The most common sampling rates are between 15 and 30 frames per second (fps).

5.1.2 Video Formats

To facilitate the coding and decoding process, videos are captured at pre-defined resolutions. Common Intermediate Format (CIF), also known as Full Common Intermediate Format (FCIF) is a format used as a reference for video resolutions. The commonly used formats are shown in Table 5.1. The choice of the format depends upon the application requirements and available resources. When the available resources could not support a high bitrate, QCIF and SQCIF formats are used. Otherwise CIF and 4CIF formats are usually preferred.

5.1.3 Video Encoding Standards

H.120 was the first digital video encoding standard, which was developed by International Telegraph and Telephone Consultative Committee in 1984. The introduction of H.120 began a new era of research, and several video encoding standards were proposed afterwards. Most of these standards were provided by ITU-T Video Coding Expert Group (VCEPG) and ISO/IEC Moving Picture Expert Group (MPEG).

H.261 is the most influential of these standards and it is considered as the first practical coding standard in video industry. H.261 provided many technological innovations which influenced the development of subsequent standards, such as MPEG-1 and MPEG-2. MPEG-1 is a widely used lossy audio/video coding standard, while MPEG-2 is used for broadcasting digital TV contents. The most advanced encoding standards that are already incorporated in many products are

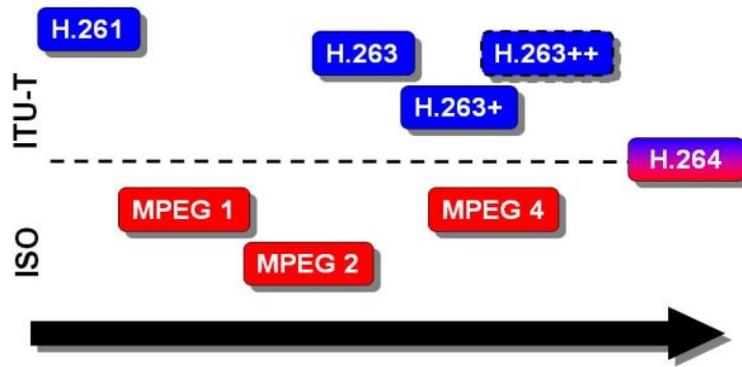


Figure 5.5: Common video coding standards

MPEG-4 and H.264 standards.

5.2 H.264 Standard

5.2.1 H.264 Audio Video Coding (AVC)

The H.264/AVC standard, which is also known as MPEG-4 part 10, was introduced in 2003 by ITU-T VCEG and ISO/IEC MPEG. The standard provides a good compression and network friendliness for both conventional and non-conventional video applications [218], such as VoIP, VOD, IPTv, video conferences, HDTV, MMS, and storage on digital devices. Initially, 3G mobile video services were based on H.263 and MPEG-4 standards, but recent services are already almost exclusively based on H.264/AVC [170]. 3GPP recommends the H.264/AVC baseline profile for all services, such as conversations, packet-switched streaming, messaging, and multimedia broadcast. The main advantages of H.264/AVC can be summarized as follows.

- High coding efficiency
- High video quality
- Error robustness to operate on the Internet and wireless networks
- Support for both realtime and non-realtime applications
- Network friendliness

To make the standard compatible with different classes of applications and various network standards, H.264/AVC divides its operations into two layers: Video Coding Layer (VCL) and Network Abstraction Layer (NAL). The structure of H.264 AVC encoder is shown in Figure 5.6. VCL is responsible for encoding video contents, while NAL converts the VCL representation of video into a format that is compatible with various transport layer protocols and storage media. A brief description of these layers is presented below.

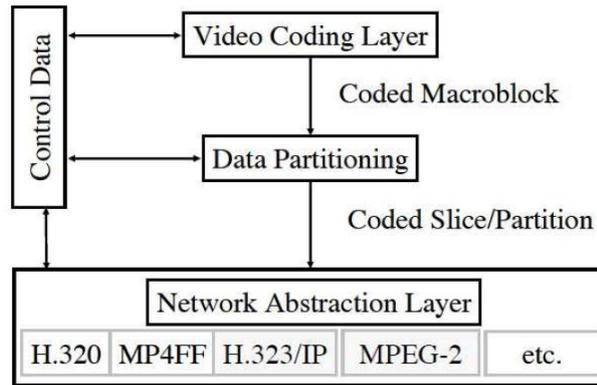


Figure 5.6: Structure of H.264/AVC Encoder

5.2.2 Video Coding Layer

Video encoding is performed by the video coding layer. For efficient encoding, the layer uses a variety of algorithms such as entropy coding, motion compensation, and discrete cosine transform (DCT). VCL first transforms the original video into a bitstream and then divides it into up to three partitions. Since some coded information is more important than other, therefore this separation enables prioritized service for important information.

The coding layer first decides whether to code an image independently or predict it from other images, and then represents the image as a group of blocks of associated luma and chroma values, called macroblocks.

5.2.3 Network Abstraction Layer

The network abstraction layer provides network-friendliness to VCL. It enables simple customization of VCL to map encoded data structures to variety of transport layer protocols and storage media, such as RTP/IP, MPEG-2, ISO MP4, etc. NAL transforms VCL data into discrete blocks, called NAL units. Each NAL unit represents a packet with an associated header and payload. The length of the header is one byte and it signifies the type of payload encapsulated in the unit. The payload is interleaved. The series of NAL units generated by the layer is called a NAL unit stream.

The layer prepares a VCL bitstream to be transmitted in a general way, so that general adaptability could be achieved. However, NAL is not fully standardized and therefore various implementations of H.264/AVC may have different specifications of NAL.

The interface between VCL and NAL layers is called slice. A slice is group of coded macroblocks arranged in the scanning order. The dimension of a slice is not fixed, and therefore an image frame may contain one or more slices. Slices are mapped to the packets that could be transmitted over the physical link. Slices

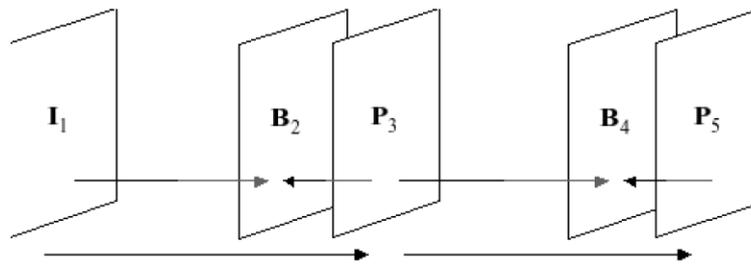


Figure 5.7: Relationship among various categories of image frames

are self-contained and therefore they could be decoded independently of each other. H.264/AVC utilizes the technique of Flexible Macroblock Ordering (FMO) [159] to enhance robustness. The interface provides H.264/AVC the flexibility for efficient encapsulation of the video data.

5.2.4 Types of Image Frames

For efficient encoding, the H.264/AVC standard makes use of four different types of image frames. The relationship among various categories of image frames is shown in Figure 5.7. The details of these frame types are given below.

1. **Intra-frame (I):** An intra-frame is independent of other frames. The regions of an intra-frame are only predicted from other regions within the same frame. Thus I -frames exploits only the spatial correlation of picture elements within the same image. The first image of a video sequence is always coded as an intra-frame due to the absence of other frames that could have been used for prediction. I -frames serve as the basis of the decoding process.
2. **Predictive-frame (P):** A P -frame is coded by using information from previous I -frames or P -frames. This is accomplished with the help of a motion compensation algorithm. The algorithm first searches the most similar region in the previous image and then determines the change in location of the region. Finally, the region is coded as a motion vector specifying the total displacement.
3. **Bidirectional-frame (B):** The regions of a B -frame can be predicted either from the previous images or from the following images or from both. In case a region is predicted from two images, interpolation is used. A B -frame can be combined with a P -frame to form a single PB frame. This reduces the overhead associated with each frame and provides lower bit-rates. B -frames are not used as reference frames, and therefore there is no need to store these frames in the coder or the decoder for future predictions.
4. **Switching P-frame (SP) and switching I-frame (SI):** H.264/AVC is the first standard that introduced the concept of SP and SI frames. The main property

of *SP*-frames is that identical frames can be reconstructed even when different frames are used for their prediction. This allows *SP*-frames to replace *I*-frames in applications such as splicing, random access and error recovery [103]. *SI*-frames are used in conjunction with *SP*-frames for enhanced efficiency.

5.2.5 H.264 Scalable Video Coding

The Joint Video Team (JVT) formed by the MPEG and VSEG groups proposed a scalable extension of H.264/AVC, called H.264 Scalable Video Coding (SVC) [157], [158]. SVC can be classified as a layer video encoder [157]. It employs a layered structure to provide spatial, temporal, and quality scalability. An SVC bitstream can be easily adapted by the server according to the network state and receiver capabilities. The structure is organized such that the data of different importance are kept in different layers. Thus, Unequal Erasure Protection (UEP) can be used to provide higher level of protection to the important data. A scalable bitstream enables unequal error protection, hierarchical modulation, and selective retransmission. The idea is to provide more protection to the base layer to enhance robustness. Thus, an SVC bitstream is less prone to errors when the video is transmitted over an unreliable link.

SVC encodes a video sequence at the highest possible resolution, and then allows the bitstream to be adapted to provide lower resolutions. The standard provides three types of scalability: temporal, spatial, and quality scalabilities. Spatial scalability means the bitstream can provide various spatial resolutions. Temporal scalability implies that the bitstream can provide several frame rates. While, the quality scalability provides various levels of visual qualities. The scalability is enabled by encoding the video into several layers. The lower layers contain data for lower resolutions and vice versa. Lower layers are more important because they provide the basic video quality at the lowest bit-rate. The higher layers have the refinement data, which could be dropped when the available bandwidth is not sufficient.

SVC is very useful for video streaming in digital cellular networks. For a user, the bandwidth and channel conditions are time-varying due to mobility and other physical layer phenomena. Similarly, various users have different amount of available bandwidth due to different channel conditions. Therefore it is not a good idea to support all users with a non-scalable bitstream. An SVC bitstream is very useful in this situation, as it can be adapted according to the channel characteristics of individual users.

To provide spatial scalability, a video sequence is decomposed into multiple spatial layers. Each of these layers can be coded separately or in conjunction with lower spatial layers to eliminate redundancy. Within a spatial layer, the data can be

separated into multiple quality layers.

Scalability of a bitstream enables adaptability both for bit-rate and receiver capabilities without the need of re-coding the original signal. It is a highly desirable feature, especially when a video is to be transmitted over networks with different characteristics. Even within a network, user devices often have different screen resolutions and processing powers. In a multimedia session, usually the video is the most resource consuming service. Therefore, the capability to adapt video bitrates according to network conditions is a highly desirable property. This greatly reduces the computational requirements of video servers as there is no need of transcoding or re-coding. [170]

5.3 Video Transmission over Networks

On packet-switched networks, a video is typically streamed by either an end-to-end protocol or a broadcast mechanism. Therefore signaling messages within the bitstream are required to enable media-aware network elements (MANEs). MANE can provide bit-rate adaptation and enhanced protection to the important data.

The identification of data belonging to different layers is done by the network abstraction layer (NAL). NAL hides the syntactical details of VCL structures and enables application layer readability of NAL units. The NAL header provides identification of temporal, spatial, and quality scalability information encapsulated within a NAL unit.

Modern networks support a variety of network applications, including video conferencing, IPTV, progressive download etc. These services have different QoS requirements in terms of throughput and delays. Mobile and portable devices are characterized by a rapid evolution. These devices offer significantly different characteristics, such as screen resolutions, memory and processing power. In such a heterogenous environment, a non-scalable bitstream cannot efficiently fulfill the requirements of the users. Scalable video streams are an important enabler to fully exploit the power of these environments. Many existing mobile devices and services only support single layer codecs, therefore backward compatibility of H.264/SVC with these services is an important issue.

Integration of SVC in Content Delivery Protocols

For successful deployment of SVC, it must be integrated in existing content delivery protocols, such as Realtime protocol (RTP) [172] and 3G/ISO file format [170]. These protocols are important means of delivering multimedia services, such as video streaming and downloading, respectively.

Many multimedia protocols use RTP for transmission, including DVB-H and

MBMS. [170]. The methodology of integrating SVC data in RTP protocol is provided by Wenger et al. in [216]. An NAL unit is directly encapsulated within an RTP packet, which provides signaling information about various scalabilities. The integration of the signaling information directly into the RTP payload header enables the functionality of MANE nodes.

Another important video transmission protocol is MPEG-2 Transport Stream (TS) [125], which is widely used in digital broadcasting. Currently, there exists specifications to integrate only H.264/AVC into MPEG-2 TS [173]. The current research is focused on enabling layered transmission of SVC in multiple TS streams. This would allow the transmission of scalable bitstreams via MPEG-2 TS protocol. [170]

5.4 Challenges

Video services pose several challenges to wireless multimedia transmission systems. These challenges have two important dimensions [122]. Firstly, video data have stringent QoS requirements that must be fulfilled by the network. Secondly, the unreliable nature of wireless link makes furnishing QoS a challenging task. The main challenges in wireless transmission of video services are discussed below.

Strict Timing Constraints

Video services require guarantees on both throughput and delay. For these applications even an end-to-end delay of several hundred milliseconds is unacceptable. The packets that cross their delay limits are considered expired and are of little use to the receiver. Transmitting these packets results in wastage of network resources and can actually delay the transmission of subsequent valid packets. Meeting video packet deadlines is even more challenging on best-effort networks such as the Internet [122].

High Bandwidth Demand

Video applications are bandwidth hungry. The streaming of multiple video sequences over a wireless network with limited bandwidth is a challenging problem. Efficient scheduling mechanisms are required to ensure satisfactory viewing experience to end-users.

Need for Unequal Error Protection

H.264/AVC and H.264/SVC encode a video sequence by using one or more reference images from the sequence to compress an image frame. This coding tech-

Start-Pos.	Length	LId	TId	QId	Packet-Type	Discardable	Truncatable
0x00000000	262	0	0	0	StreamHeader	No	No
0x00000106	14	0	0	0	ParameterSet	No	No
0x00000114	16	0	0	0	ParameterSet	No	No
0x00006fd4	238	1	0	0	SliceData	No	No
0x000070c2	9	0	1	0	SliceData	Yes	No
0x000070cb	979	0	0	0	SliceData	No	No
0x0000749e	9	0	1	0	SliceData	Yes	No
0x000074a7	121	0	0	0	SliceData	No	No
0x00007520	976	1	1	0	SliceData	Yes	No
0x000078f0	598	1	1	0	SliceData	Yes	No
0x00007b46	9	0	2	0	SliceData	Yes	No
0x00007b4f	679	0	0	0	SliceData	No	No
0x00007dfe	979	1	2	0	SliceData	Yes	No
0x000081c9	9	0	2	0	SliceData	Yes	No
0x000081d2	768	0	0	0	SliceData	No	No
0x000084d2	983	1	2	0	SliceData	Yes	No
0x000088a9	97	1	2	0	SliceData	Yes	No

Figure 5.8: A sample trace file

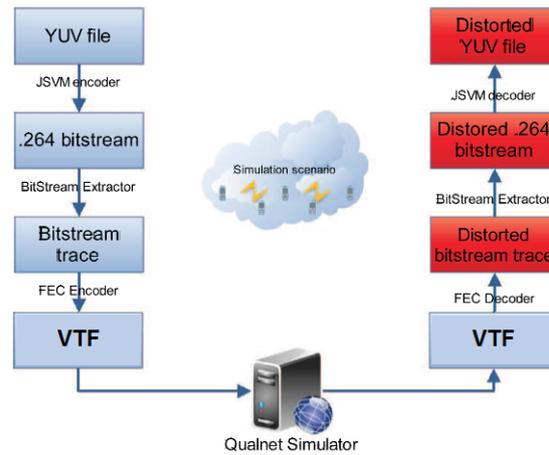


Figure 5.9: Framework for realtime video transmission

nique makes data vulnerable to greater losses. The loss of a reference frame can cause error propagation across all the frames that are derived from that reference frame. Similarly, losses in the base layer of SVC can make the decoding impossible. Therefore, the important data should be transmitted over a reliable channel to avoid these problems. However, unequal protection of data is not supported by most wireless networks [122].

Time-Varying Channel Conditions

Wireless networks have various characteristics that negatively affect the transmission of video streams over these networks. Some of these problems include multipath fading and shadowing, limited bandwidth, interference, and mobility. Wireless channel is a challenging medium in itself, yet the QoS requirements and characteristics of video applications make the problem even more challenging.

5.5 Simulation of Video Transmission over Networks

There are three possible approaches that could be used to simulate the transmission of video streams over networks [227]:

Video bitstream In this approach, the complete bitstream is transmitted over the network. A bitstream is generated by the encoder and it contains the actual video contents. Video bitstreams can be used in both testbeds and network simulation. The advantage of using actual bitstream is that it is the most accurate representation of the original video scene. However, video bitstreams are large in size and therefore difficult to manage in a simulation.

Traffic Models Video traffic models are based on the statistical properties of real video bitstreams. They have received significant attention in the research [89]. The advantage of traffic models is that only a small number of customizable parameters can define a video bitstream in an efficient way. However, currently Qualnet does not support QoS-enabled connections on WiMAX that are based on traffic models.

Packet Trace In this approach important information from a video bitstream is extracted that is later on used for simulation. The information include packet sizes and types, and quality related information. There is no need to transmit the actual bitstream. However, reconstruction of distorted video sequence at the receiver requires writing custom codes.

5.6 Video Transmission Framework

The most important application of the rtPS class is the transmission of video bitstreams. Therefore, we designed and implemented a video transmission framework (VTF) in Qualnet. The objective is to evaluate the performance of the rtPS intra-class scheduling algorithm in transmitting realtime video streams. The framework can transmit videos encoded in H.264/AVC and H.264/SVC formats.

The different components of the framework are shown in Figure 5.9. The framework makes use of JSVM reference library [2] and is similar to a framework proposed in [227]. The two framework utilizes similar tools. However, the earlier framework makes use of packet traces only, while our framework makes use of both video traces and bitstreams. Unlike the earlier framework, VTF provides QoS support to video streams.

To perform the experiment we used the video sequences obtained from the video trace library [1] of Arizona State University. The video sequences are in raw *.yuv* format. Firstly, the framework encodes a raw video sequence into H.264/SVC bitstream by using the JSVM video reference library. Then *BitStreamExtractor*, which is a bitstream generation tool in the JSVM library, is used to generate a packet trace. A packet trace specifies the order, size, and parameters of each packet of the bitstream. A sample packet trace is shown in Figure 5.8. The first and second columns indicate the starting position and size of each packet, respectively.

Parameter	Value
JSVM version	9.8
Number of layers	2
Base layer	0
Enhancement layer	1
GOP	16
Slice mode	2
Slice argument	1000

Table 5.2: The parameters of the JSVM encoder used for encoding the video stream

Columns LId, TId, and QId specify values of dependency identifier, temporal identifier, and quality level identifier, respectively. Then two flags specify whether the packet can be discarded or truncated during transmission.

To enable the transmission of video stream, we modified the variable bitrate traffic generator (VBR) of Qualnet. The current implementation of the Advanced Wireless Library of Qualnet only support QoS-enabled connections for VBR and CBR traffic generators. For an end-to-end transmission, the video transmission framework is implemented at both the sender and the receiver. At the sender, the framework transmits the encoded H.264 bitstream according to the parameters specified in the packet trace. At the receiver, the framework constructs distorted bitstream with the help of packet trace. The distorted H.264 bitstream is then decoded to raw .yuv format with the help of JSVM video reference library.

5.6.1 Performance Analysis

The objective of the simulation is to assess the performance of the proposed rtPS intra-class scheduler in scheduling video traffic. The simulations are performed by using the video transmission framework. For the simulation, we used *Carphone* video sequence obtained from the video trace library of the Arizona State university. The parameters of the JSVM encoder used to generate H.264 bitstream are shown in Table 5.2.

The bitstream used in the experiment was encoded to provide two spatial resolutions, i.e. CIF (352×288) and QCIF (176×144). The CIF layer has four quality sublayers, while the QCIF layer has one quality sublayer. In each spatial-quality layer, the group of picture (GOP) size is set to 16 pictures. An intra-picture is inserted every 4 GOPs (64 image frames) to provide error recovery point to the decoder. The five spatial-quality layers and the four temporal layers provides up to 20 layers bitstream adaptation. The *Carphone* test sequence was used which consists of 382 image frames.

The simulation was performed in Qualnet with a total of 20 subscriber stations and one base station. The parameters of the physical and MAC layers are shown in

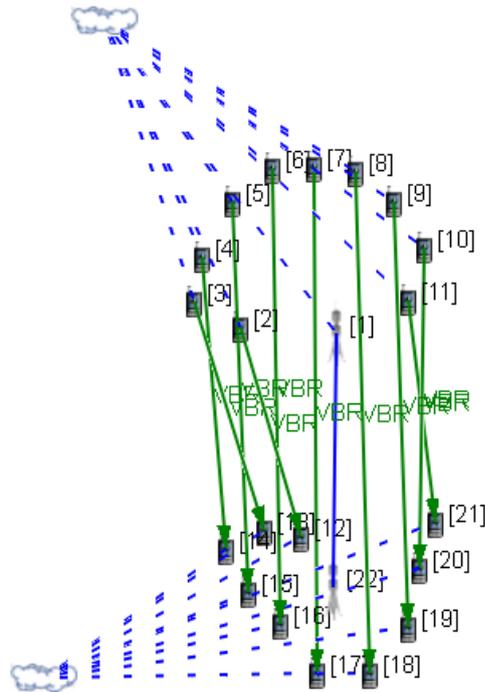


Figure 5.10: Simulation scenario for VTF

Table 4.3. 10 of the subscriber stations are used as source, while other 10 are used as the receiver. The scenario is shown in Figure 5.10. The experiment was performed by gradually increasing the number of videos transmitted simultaneously. Peak Service to Noise Ratio (PSNR) is used to compare the quality of the received video to the transmitted video. Figure 5.11 shows the average PSNR as function of number of concurrent video streams. The average PSNR is simply the arithmetic mean of PSNR of all video streams. High values of PSNR implies that the quality of received videos is good.

The above experiment was repeated with another video sequence, namely *News*. The video sequence was taken from the video sequence library of Arizona State University. It consists of 300 frames. The parameters of the JSVM encoder used for encoding are shown in Table 5.2. Figure 5.12 shows the average PSNR as function of number of simultaneous video streams for *News* video sequence. The value of PSNR remains above 90, which implies that the quality of received video is good.

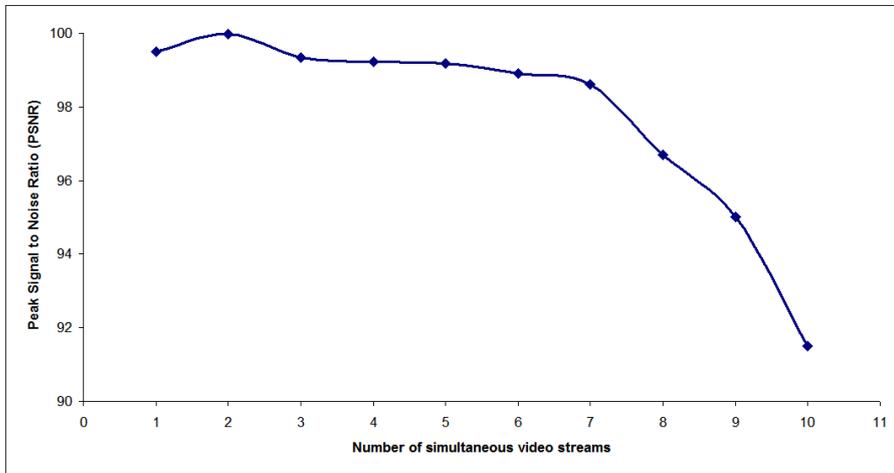


Figure 5.11: Video quality at the receiver for *Carphone* video sequence

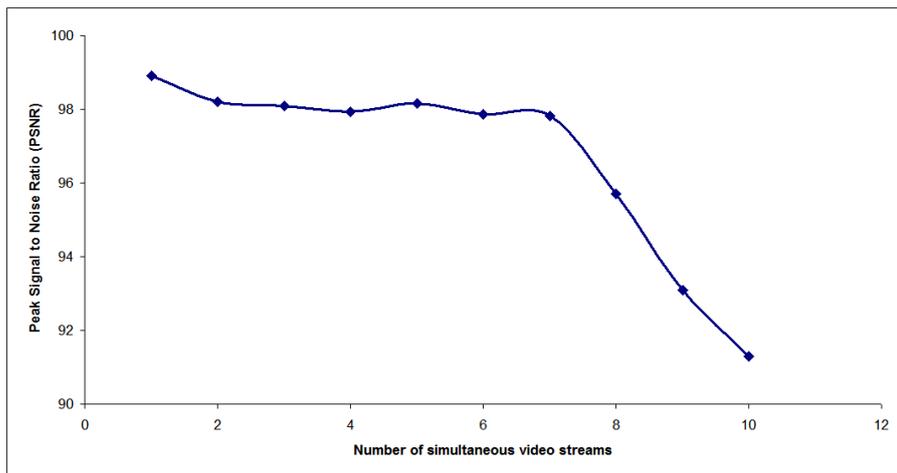


Figure 5.12: Video quality at the receiver for *News* video sequence

5.7 Conclusion

In this chapter, we provided the architecture of a video transmission framework. The framework is based on TLSA and it can be used to transmit H.264/AVC and H.264/SVC video streams. The framework was implemented in Qualnet and the simulation analysis were performed to evaluate the transmission of H.264 video streams using TLSA. The results indicated that TLSA could be used for efficient transmission of video streams.



6

Conclusion and Future Perspectives

6.1 Conclusion

The IEEE 802.16 broadband wireless access standard is an important technology that supports variety of realtime and non-realtime applications. The support of these applications and services is enabled by its QoS framework that provides differentiated connection-based packet scheduling. Without appropriate packet scheduling mechanisms, these applications may not provide acceptable quality of experience to the end-users. In the 802.16 standard, the complex task of scheduling is distributed among three schedulers, i.e. base station uplink scheduler, base station downlink scheduler, and subscriber station scheduler. The packet scheduling in the uplink direction is done by the base station uplink scheduler and the subscriber station scheduler, while the packet scheduling in the downlink direction is the responsibility of the base station downlink scheduler. The standard specifies various QoS components and their functions. However, it does not specify algorithms for some important components, such as packet schedulers and connection admission control.

In this dissertation, we provided a packet scheduling algorithm for the uplink direction. The algorithm consists of two levels. In the first level, an inter-class scheduling algorithm distributes bandwidth among various service classes according to their QoS requirements and available resources. Then at the second level, class-specific scheduling algorithms take the bandwidth allocated by the inter-class algorithm and distribute it among the connections of the concerned class according to their QoS requirements. The inter-class scheduling algorithm implements service class priority order and prevents starvation of lower priority flows. The class-specific algorithms ensure that each connection gets at least the minimum guaranteed service level and the resource allocation is fair.

We also proposed a connection admission control algorithm and several components to facilitate the working of the proposed scheduling algorithm. The connection admission control algorithm only admits a new connection if QoS could be guaranteed to both the incoming and the established connections. We also developed a Markov chain based queuing model that could be used for the theoretical analysis of the proposed algorithms.

The proposed algorithms were validated by extensive simulations for each service class. The simulations were performed in Qualnet 5.0. From the simulation results it can be concluded that the proposed scheduling algorithms efficiently furnish the required QoS and ensure fairness of resource distribution. The simulations were also performed to compare the performance of the rtPS intra-class scheduling algorithm and EDF algorithm. The simulation results revealed that unlike EDF algorithm, the distribution done by the proposed algorithm is fair. Furthermore, the proposed algorithms provides better results for various performance indicators,

such as throughput, latency, and packet loss ratio.

We also provided an H.264 compatible video transmission framework that is based on the proposed rtPS intra-class scheduling algorithm. Different parts of an H.264 video stream have different importance. Therefore, the transmission framework transmits various layers of a video stream according to their importance. The simulation analysis of video transmission framework showed that the proposed scheduling algorithm efficiently transmits video streams.

6.2 Future Perspectives

Wireless medium has time-varying characteristics and consequently it is unpredictable in nature. Therefore, it is important to take into account the most recent channel state information while making scheduling decisions. This can improve throughput and overall system performance. However, channel-aware schemes have two major drawbacks, i.e. unfair allocation of resources and starvation of subscriber stations with poor channel conditions. An ideal scheduling algorithm would enhance network efficiency by using channel state information, while ensuring QoS to all QoS-enabled service classes. Therefore, TLSA should be modified to take into account the channel conditions.

Another important area of further research is to study the application of TLSA in heterogenous networks. A wireless heterogenous network is made up of several radio access technologies and architectures. Enabling convergence of heterogenous networks is considered as a major challenge in the evolution of telecommunication systems. The IEEE 802.21 working group is currently working to establish an interworking standard by introducing Media Independent Handover (MIH), which could provide transparent handover among various wireless networks [4]. Convergence of various heterogenous wired and wireless networks could provide *anywhere, anytime* connectivity using the most suitable available technology. It is an effective way of enhancing mobile network capacity and user experience. Three of the most important wireless interworking scenarios include WiMAX-WiFi, WiMAX-3G, and WiMAX-LTE. It would be interesting to extend the scope of the proposed algorithms to provide support for these interworking scenarios. The details of these scenarios are provided in the subsequent paragraphs.

WiFi and WiMAX have been optimized for different usage models. WiFi is designed for providing high-speed wireless local connectivity, while WiMAX for high-speed wide area connectivity. The combination of WiMAX and WiFi is an ideal choice to provide efficient and affordable broadband connectivity in more places. WiMAX-WiFi interworking enables consistent and user-friendly broadband connectivity to end-users.

Currently, WiMAX provides very high data transfer rates, however it has restricted coverage area as compared to 3G cellular networks. WiMAX-3G interworking can provide ubiquitous high-speed data connectivity and thus enlarge user base.

Long Term Evolution (LTE) is a wireless communication standard, which is based on GSM/EDGE and UMTS/HSPA network technologies. The goal of LTE is to offer performance levels as specified by IMT-Advanced specifications for 4G networks. An interworking between WiMAX and LTE is considered as an option towards realizing 4G data networks [3]. However, the two technologies have different protocol architectures and QoS frameworks. Therefore, protocol mapping would be required for realizing interworking.



Comparative Analysis of Analytical and Simulation Models

Parameter	Value
Total uplink bandwidth	1 Mbps
Frame duration	20 ms
TDD downlink duration	10 ms
MAC propagation delay	1 μ s
Cyclic prefix	8.0
Input queue size	50000 bytes
Antenna model	omni antenna
Sampling factor	8/7
Propagation model	Two ray ground
Shadowing model	constant
Shadowing mean	4.0 dB
Temperature	290K
Noise factor	10.0
Service flow timeout	15 s
Transmit power	20 dBm
Receive power threshold	205e-12
Carrier sense power threshold	0.9 * Receive power threshold
Handover RSS trigger	-78.0 dBm
Propagation limit	-111.0 dBm
Link adaptation	Enabled
Packing	Enable

Table A.1: Important simulation parameters for comparative analysis

In this appendix, we provide comparative analysis of the analytical model provided in Chapter 3 and the simulation model presented in Chapter 4. For the comparison, the total uplink channel capacity was set to 1 Mbps. The important simulation parameters are shown in Table A.1.

Figure A.1 presents the comparative analysis of bandwidth allocation by the inter-class scheduling algorithm. For the comparison, BE traffic is generated at an average rate of 200 Kbps. The value of β_{BE} is set to 90 Kbps to prevent starvation of BE class. While for nrtPS class, the MRTR is 375 Kbps and the average traffic rate is 500 Kbps. Comparative analysis are performed with increasing load of rtPS traffic. Initially, the average traffic rate of rtPS class is 300 Kbps, which is gradually increased to 600 Kbps. The MRTR for rtPS traffic is set to 300 Kbps, while the maximum permissible delay is set to 160 ms. For rtPS traffic, the simulation and analytical curves are almost identical. The average difference between the two curves is 8.34 Kbps with a standard deviation of 2.49 Kbps. For nrtPS class, the simulation curve follows the analytical curve. The average difference between two curves is 12.65 Kbps with a standard deviation of 16.27 Kbps. The analytical and simulation curves of bandwidth allocation to BE class show greater discrepancy in the beginning. The two curves converge and becomes identical at rtPS data generation rate of 440 Kbps. The average difference between the two curves is 16.69 Kbps with a standard deviation of 21.25 Kbps. Figure A.2 shows the percentage of lost

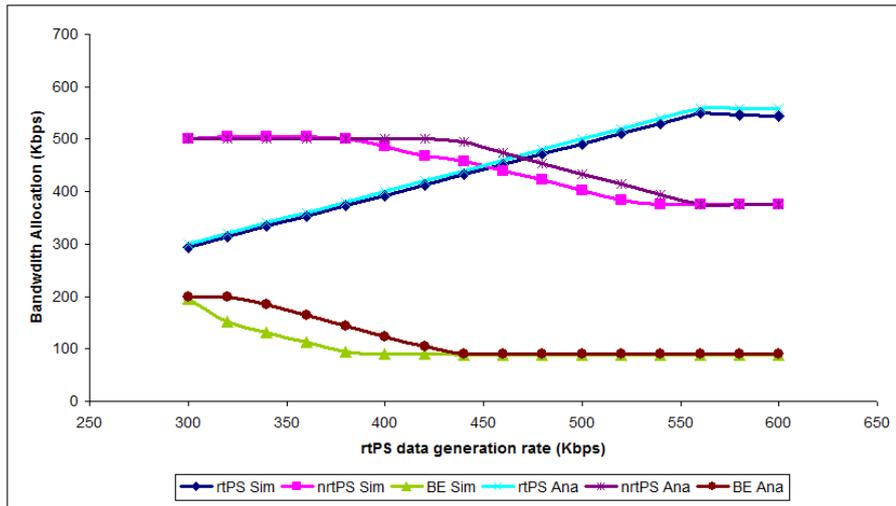


Figure A.1: Comparative analysis of inter-class bandwidth scheduling

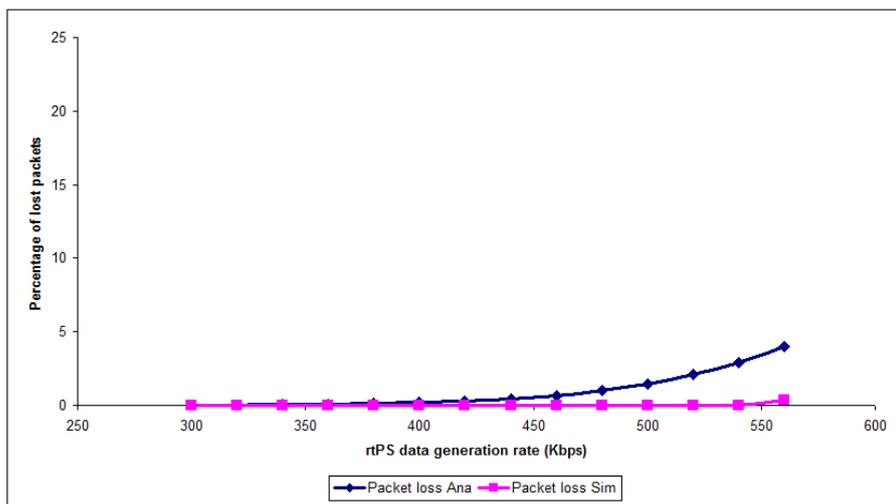


Figure A.2: Comparative analysis of packet lost in inter-class scheduling

packets as determined by the analytical model, and the actual lost observed in the simulations. Initially the simulation curve exactly follows the analytical curve. The deviation between the two curves starts at rtPS data generation rate of 460 Kbps. This is due to the time-dependent nature of percentage of lost packets. The average difference between the two curves is 0.93 pp with a standard deviation of 1.14 pp. The two figures suggest that the analytical model determines the bandwidth allocation of inter-class scheduling algorithm with good accuracy. Therefore, the model can be used for mathematical analysis of bandwidth distribution by the inter-class scheduling algorithm.

The comparison between analytical and simulation models of bandwidth allocation by the nrtPS intra-class scheduling algorithm is shown in Figure A.3. Four nrtPS connections are used for the comparison. The parameters of the connections are shown in Table A.2. The figure shows that the bandwidth allocation during sim-

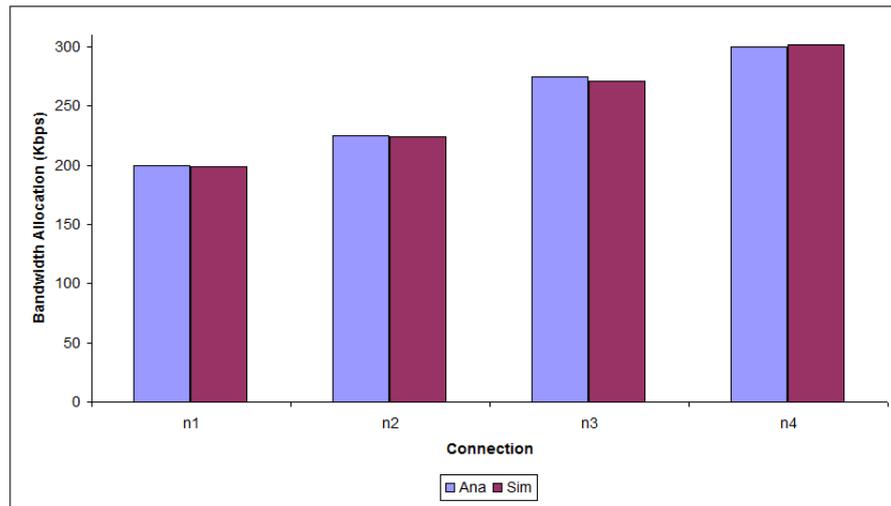


Figure A.3: Comparative analysis of nrtPS intra-class scheduling

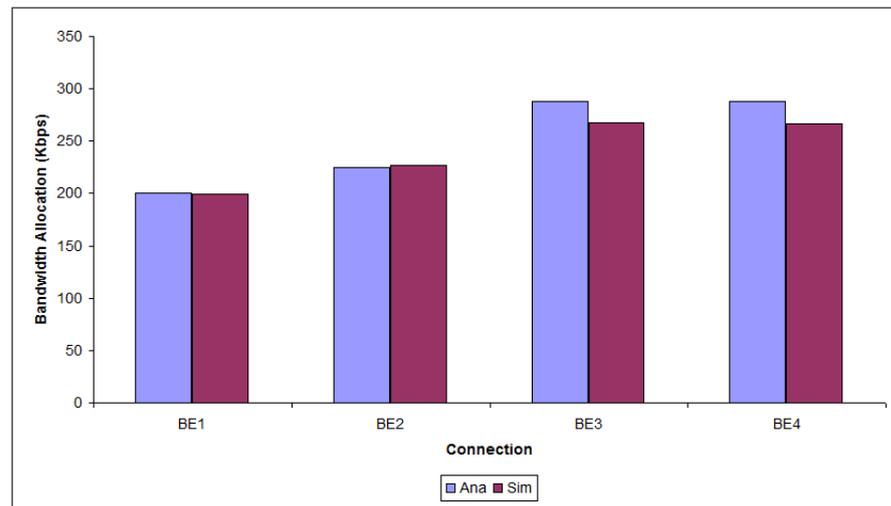


Figure A.4: Comparative analysis of BE intra-class scheduling

ulations by the nrtPS intra-class scheduling algorithm closely matches the analytical results. The average difference between two models is 1.09 Kbps with a standard deviation of 2.42 Kbps. The comparison between analytical and simulation models of bandwidth allocation by the BE intra-class scheduling algorithm is shown in Figure A.3. In this scenario four subscriber stations with one BE connection each are used to do the comparison. The average data generation rates are 200 Kbps, 225 Kbps, 275 Kbps, and 300 Kbps for connections *BE1*, *BE2*, *BE3* and *BE4* respectively. The figure shows that both models are in good agreement with each other. The average difference between the two models is 9.83 Kbps with a standard deviation of 12.14 Kbps.

The comparative analysis of bandwidth distribution among rtPS connections and the associated service ratios by the analytical and simulation models are shown in Figures A.5 and A.6, respectively. For this scenario, four subscriber stations host-

Connection	MRTR (Kbps)	Average Traffic Rate (Kbps)
<i>N</i> 1	140	200
<i>N</i> 2	200	225
<i>N</i> 3	225	275
<i>N</i> 4	250	300
Total	815	1000

Table A.2: Input traffic parameters for comparative analysis of nrtPS intra-class scheduling algorithm

Connection	MRTR (Kbps)	MSTR (Kbps)	Maximum Tolerable Delay (frames)
A	4000	9000	2
B	1000	3000	3
C	2000	4000	3
D	3000	5000	4

Table A.3: Input traffic parameters for comparative analysis of rtPS intra-class scheduling algorithm

ing one rtPS connection each are used and the total uplink bandwidth is 10Mbps. The parameters of the connections are shown in Table A.3. The average difference in bandwidth distribution between two the models is 0.35 Mbps with a standard deviation of 0.17 Mbps.

From the comparative analysis provided in this appendix, it can be concluded that the bandwidth distribution by the inter-class and the three intra-class scheduling algorithms is modeled with good accuracy by the analytical model presented in Chapter 3. The simulation analysis provide results in accordance with the analytical model and thus validates its accuracy.

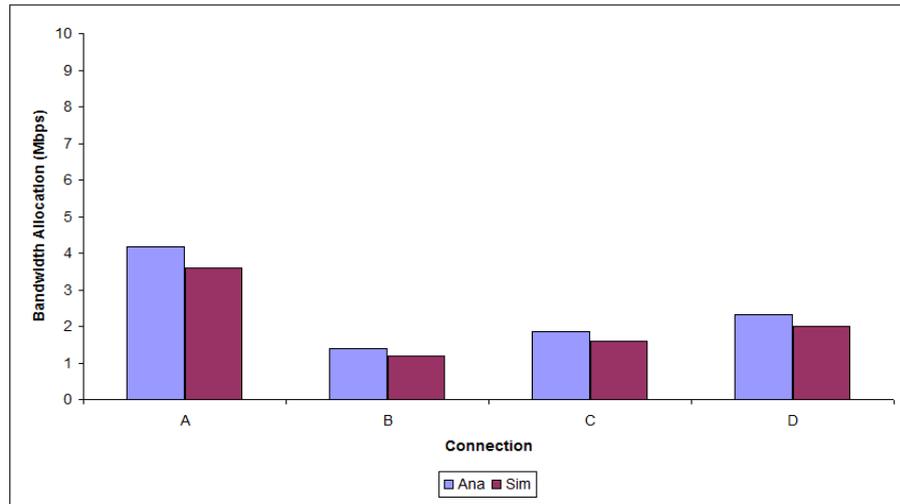


Figure A.5: Comparative analysis of rtPS intra-class scheduling

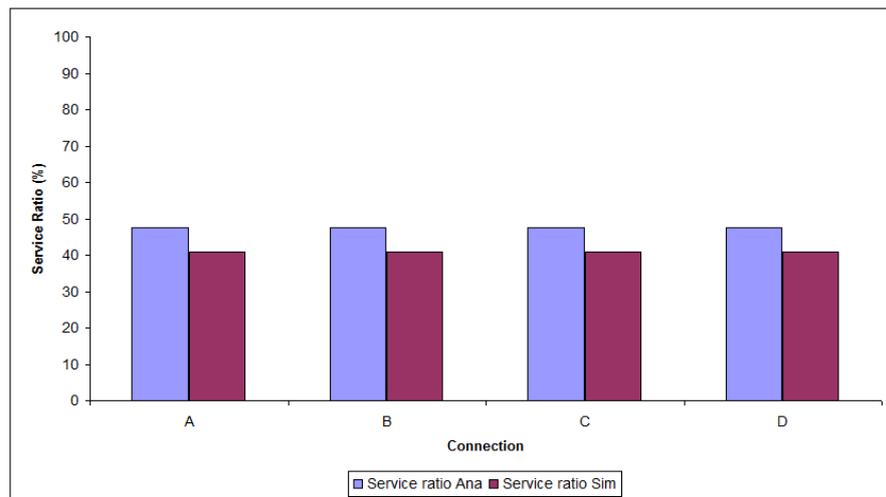


Figure A.6: Comparison of connection service ratios



B

Résumé en Français

Résumé de la thèse en français

Ce chapitre présente le résumé de la thèse en français en accord avec la loi Toubon.

1. Introduction

Un système de communication est une infrastructure complexe constituée de plusieurs sous systèmes permettant la communication entre 2 ou plusieurs entités. Dans cette thèse, nous nous intéressons aux réseaux sans fils qui procurent des bénéfices notables en mobilité et accessibilité. Les systèmes d'accès sans fil modernes tel que le WIMAX permettent l'échange de données, de voix en temps réel, de vidéo haute qualité et quantité d'autres services. Les services multimédias et particulièrement ceux qui sont temps réel sont demandeurs de fonctionnalités pointues en qualité de service (dénomé QoS dans le manuscrit). En conséquence, la QoS en WIMAX est cruciale et constitue le coeur de cette thèse. Cette QoS doit être mise en oeuvre par des technologies au niveau de la couche MAC et de la couche physique. L'état de l'art est organisé en deux parties. Dans la première partie, on rappelle l'architecture du standard IEEE 802.16 et de ses blocs principaux avant de donner le détail des technologies mises en oeuvre au niveau des couches physiques et MAC du WIMAX. Dans la seconde partie de l'état de l'art, on rappelle le problème de la qualité de service (QoS) et de l'architecture de QoS fournie par le standard IEEE 802.16. Les 2 éléments principaux sont le contrôle d'admission (dénomé CAC par la suite) et l'ordonnancement de paquets. Le standard ne spécifie pas les détails algorithmiques de ces composants qui ont permis beaucoup de design différents de la part des chercheurs et qui reste un champ ouvert de recherche. Une classification de ces propositions est fournie dans la seconde partie de l'état de l'art.

Dans la partie suivante, on détaille la partie théorique de notre contribution. Enfin nous concluons le chapitre par un bilan et des travaux futurs en lien avec ce travail.

2. La QoS dans les réseaux IEEE 802.16

Deux principaux composants sont essentiels dans l'architecture de QoS des réseaux IEEE 802.16. Il s'agit du contrôle d'accès des connexions (CAC) et l'ordonnancement des paquets. Ce chapitre a pour but de donner un aperçu de l'architecture de QoS et les algorithmes proposés pour le CAC et l'ordonnancement.

Le terme de qualité de service (QoS) se réfère à l'ensemble des normes et des techniques qui assurent des performances de qualité dans les réseaux de communication. Dans les réseaux sans qualité de service, tous les trafics sont traités sans aucune différenciation. Par conséquent, une seule demande de bande passante peut dégrader le niveau de service de l'ensemble du réseau. Dans les réseaux modernes (à QoS), les applications temps réel exigeantes en qualité de transmission obtiennent satisfaction.

Le problème de la garantie de la QoS consiste à déterminer un partage efficace des ressources physiques entre les utilisateurs, tels que les critères de performance comme le retard, la gigue, et le débit soient garantis. Les mécanismes de QoS peuvent être catégorisés comme suit :

- le contrôle d'accès : ces mécanismes permettent de décider comment et quand une nouvelle connexion est admise dans le réseau et avoir accès à des ressources.
- le contrôle de trafic : ces mécanismes déterminent la façon dont le paquet est classifié, ordonné, le contrôle de flux est également réalisé pour tout trafic qui entre dans le réseau.

Toute application souhaitant utiliser les ressources du réseau doit spécifier ses paramètres de QoS dont un sous ensemble est décrit ci-après :

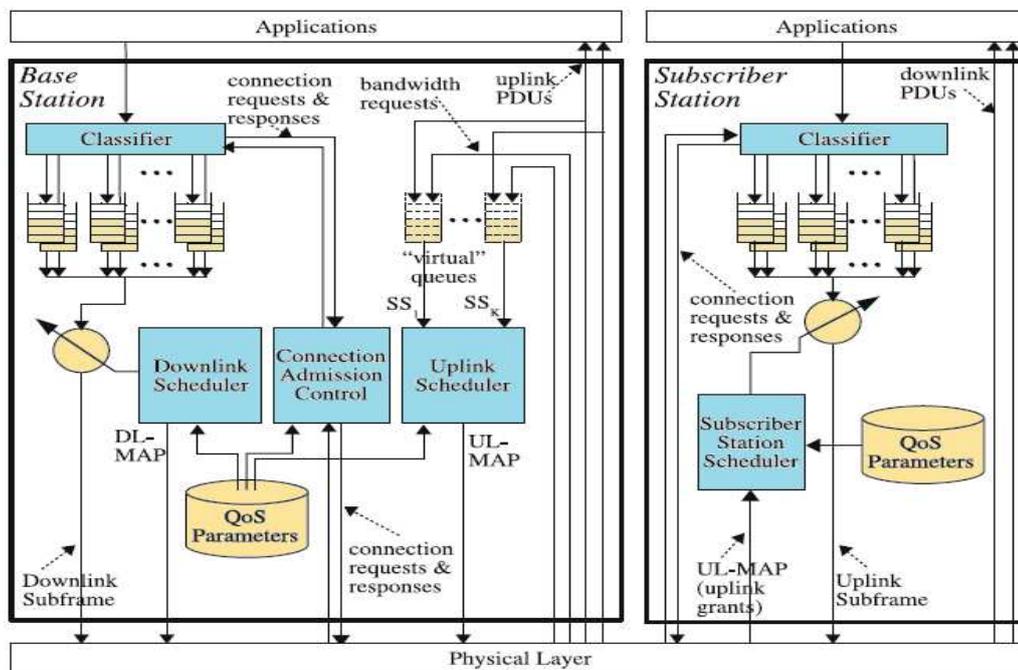
- Minimum Reserved Traffic Rate (MRTR) : le taux de trafic minimum garanti à un flux de service.
- Maximum Sustained Traffic Rate (MSTR) : le taux de trafic maximum associé à un flux de service.
- Maximum Tolerated Latency : le délai maximum acceptable entre une source et une destination.
- Tolerated Jitter : la gigue tolérée
- Unsolicited Grant Interval : L'intervalle de temps entre deux demandes successives de bande passante accorder à une connexion sur liaison descendante.
- Unsolicited Polling Interval : L'intervalle maximal entre deux demandes successives de bande passante pour une connexion sur la liaison montante.

Nous allons maintenant décrire les principales fonctions des deux mécanismes de contrôle de la QoS à savoir le mécanisme CAC et l'ordonnancement. Le mécanisme CAC est un premier composant très important dans la gestion de la QoS, il refuse ou accepte une nouvelle connexion que si les ressources sollicitées peuvent être garanties sans dégrader la

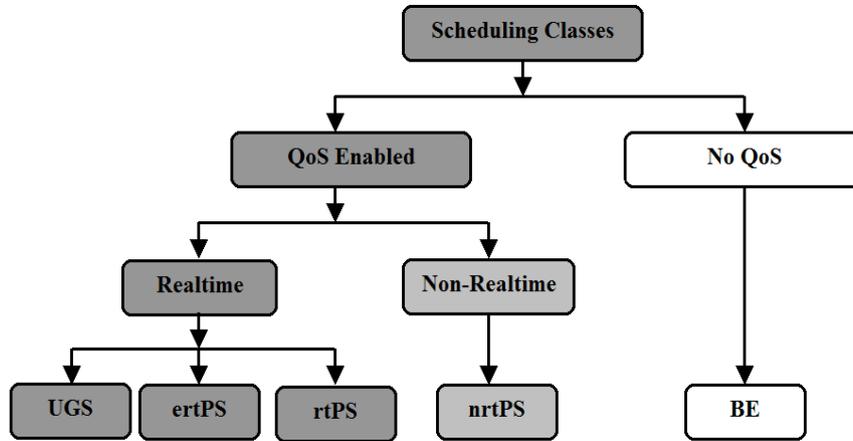
qualité de service des connexions en cours.

L'ordonnanceur : est le composant le plus important au niveau de la couche MAC. Il est constitué de deux parties essentielles, les files d'attente et l'algorithme d'ordonnement. Les files contiennent des informations tels que les données de contrôle, les données utilisateurs, les requêtes de bande passante. En utilisant plusieurs files d'attente, l'algorithme peut associer un traitement spécifique et une priorité aux applications critiques. Les décisions prises par l'ordonnanceur reposent sur des critères de performances tels que la taille de la file, le délai, MRTR, MSTR, les informations du canal radio.

L'architecture QoS WiMAX offre un service différencié. Elle supporte à la fois des applications temps réel et des applications non temps réel. Il prend en charge la transmission simultanée de données, voix et vidéo en utilisant les technologies adaptées de la couche MAC et physique. Les applications de type vidéo et voix ne peuvent tolérer une gigue importante mais par contre elles tolèrent certaines pertes. Alors que les applications de données ont une exigence complètement à l'inverse des applications temps réel. L'architecture de QoS est illustrée par la figure suivante :



La couche MAC WiMAX fournit des services d'ordonnement et des services de transmission de données pour assurer un envoi efficace des MAC PDU. Les premiers spécifient les procédures pouvant être utilisées pour les flux sur la liaison montante, tandis que les seconds concernent les flux sur la liaison montante et descendante.



Les classes de service d'ordonnancement sont récapitulées dans la figure ci-dessus :

Afin de bien gérer toutes les demandes, cinq catégories ont été définies et utilisées. Elles sont résumées dans le tableau suivant :

Scheduling service	Data delivery service	QoS parameters
Unsolicited grant service (UGS)	Unsolicited grant service	Maximum sustained traffic rate, latency tolerance, jitter tolerance
Extended realtime polling service (ertPS)	Extended realtime variable-rate service (ERT-VR)	Maximum sustained traffic rate, minimum reserved traffic rate, latency tolerance, jitter tolerance
Realtime polling service (rtPS)	Realtime variable-rate service (RT-VR)	Maximum sustained traffic rate, minimum reserved traffic rate, latency tolerance
Non-realtime polling service (nrtPS)	Non-realtime variable rate service (NRT-VR)	Maximum sustained traffic rate, minimum reserved traffic rate
Best effort service (BE)	Best effort service	Maximum sustained rate

Dans la littérature, les deux composants essentiels ont été traités et nous pouvons dans ce chapitre les résumer dans la suite.

1. Les **mécanismes CAC** peuvent être classés en deux catégories à savoir les

mécanismes dits flexibles et ceux dits conservateurs. Les mécanismes flexibles sont ceux qui, pour accepter une nouvelle connexion accepte de dégrader les performances de celles en cours afin d'obtenir plus de ressources. Alors que le second, rejète une nouvelle connexion dans le cas où les ressources ne sont pas disponibles.

Dans le cas des mécanismes flexibles, la dégradation peut être vue de trois façons différentes : (i) la dégradation de service (ii) l'emprunt de la bande passante (iii) et le vol de bande passante.

La première consiste à diminuer la bande passante allouée aux services de priorité inférieur à celle qui se présente. Y. Ge et G-S. Kuo ont suggéré que cette politique ne soit utilisée que pour des opérations de handover.

La seconde diminuer la bande passante aux connexions à taux adaptatif en cas de congestion. D. Niyato et E. Hossain ont défini schéma d'emprunt de bande passante comme une stratégie de jeux non coopérative [111]. L'objectif du jeux est de trouver un point d'équilibre entre les connexions rtPS et nrtPS pour trouver des ressources pour la nouvelle connexion. H. Wang et W. Li intègre l'emprunt de la bande passante avec une dégradation progressive [112]. Wang et al. propose un emprunt de la bande passante proportionnel avec une politique de sauvegarde d'une portion du canal pour des opération de handover [113].

La troisième politique est basée sur une approche à seau à jeton (*token bucket*). Jiang et Tsai ont fourni un schéma de vol de bande passante dans laquelle une connexion est caractérisée par deux paramètres : le taux du jeton et la taille du seau.

Dans les mécanismes conservateurs, les utilisateurs ne sont pas très satisfaits car une connexion est tout simplement rejetée si des ressources réseaux manquent. Un schéma similaire a été fourni par O. Yang and J. Lu [119]. K. Eunhyun et al. ont proposé une solution basée sur un modèle de Markov [120]. Un schéma CAC conservateur [121] qui prend en compte la puissance du canal radio a été proposé par Rong et al. Les auteurs ont proposé d'utiliser la puissance adaptative du canal afin de maximiser l'utilisation de la puissance de la station de base. Toutefois, le schéma proposé est limité à WiMAX fixe et aux stations d'abonnés ayant des besoins de bande passante non variables.

2. Les algorithmes d'ordonnement : très souvent, les algorithmes d'ordonnement proposés pour WiMAX distribuent en premier la bande passante disponible entre les différentes classes de service. Ensuite, des algorithmes spécifiques à chaque classe sont utilisés pour distribuer la bande passante, affecté à la première étape, parmi les flux de service.

- a) Les algorithmes inter-classes : plusieurs ordonnanceurs proposés pour WiMax utilisent une file d'attente unique pour chaque classe de trafic [127] [128] [129] [130] [131]. Cependant, dans [132], Y. Wang et al. ont proposé une file combinant rtPS, ertPS, and UGS dans le but de simplifier l'algorithme et la complexité de calcul de celui ci. De manière similaire, J. Borin and N. da Fonseca ont proposé d'utiliser une seule file d'attente pour rtPS and nrtPS classes. Une seconde file de plus haute priorité est utilisée pour les connexions UGS. Les paquets dont le délai toléré est dépassé sont déplacés dans la file UGS. De manière générale, le niveau de priorité détermine l'ordre d'attribution des ressources. Comme dans [132] [134] [135] [136] [137].

- b) Les algorithmes intra-classe : le plus simple des algorithmes est le Round Robin (RR). L'inconvénient est qu'il peut attribuer de la bande passante à des connexions qui n'ont pas de paquets à transmettre. Des chercheurs ont proposé l'utilisation de Weighted Round Robin (WRR) pour WiMAX [127] [128] [124] qui n'est pas par contre pas adapté au service temps réel. L'algorithme EDF (Earliest Deadline First) [141] est plus simple des algorithme basé sur le délai. Il permet l'ordonnement des paquets avec une tolérance en délai très faible. Des algorithmes basé sur le délai tel que Largest Weighted Delay First (LWDF), peut aussi être utilisé pour garantir un taux minimum de trafic [142]. Un autre algorithme basé sur le délai est Delay Threshold Priority Queueing (DTPQ) [143]. DTPQ prend en considération les délai tolérés des trafic temps réel que lorsque le paquet en tête de file a atteint la valeur seuil en délai. L'inconvénient majeur de ces algorithme est la complexité algorithmique qui peut constituée une limite dans le cadre des réseaux haut débit.

Basé sur les informations d'état du canal de transmission, les algorithmes d'ordonnement au niveau de la couche Mac peuvent être classés en deux catégories :

- a) Les algorithmes dépendant du canal (channel-aware) : ces algorithmes prennent des décisions selon les informations du canal. Atutrement dit, les paquets sont transmis que si les conditions du canal sont bonnes et les probabilités de perte de paquets est en dessous d'un seuil prédéfini. Il existe des modèles mathématiques [146] [147] et des heuristiques [148] [149] [150] qui permettent de prédire le futur état du canal. Les stations de base ont la possibilité de connaître l'état du canal sur la liaison descendante [151] à travers indicateur appelé CQI (Channel Quality Indicator). Ces algorithmes donnent la priorités aux stations avec les meilleures conditions de canal, ce qui évidemment sous alloue ou n'alloue pas du tout les stations dont les taux d'erreurs sont importants. Les algorithmes « channel-aware » peuvent être classés selon les paramètres qu'ils tentent d'optimiser. On cite l'équité, la maximisation des débits, la minimisation des délais.

2. Les algorithmes indépendant du canal (channel-unaware) : à l'inverse de la précédente catégorie, ces algorithmes ne prennent pas en considération l'état du canal pour prendre des décisions. Le canal est considéré sans erreur et la nature des liaisons sans fil est ignorée. Nous pouvons les classer en deux catégories les algorithmes homogènes et hybrides.

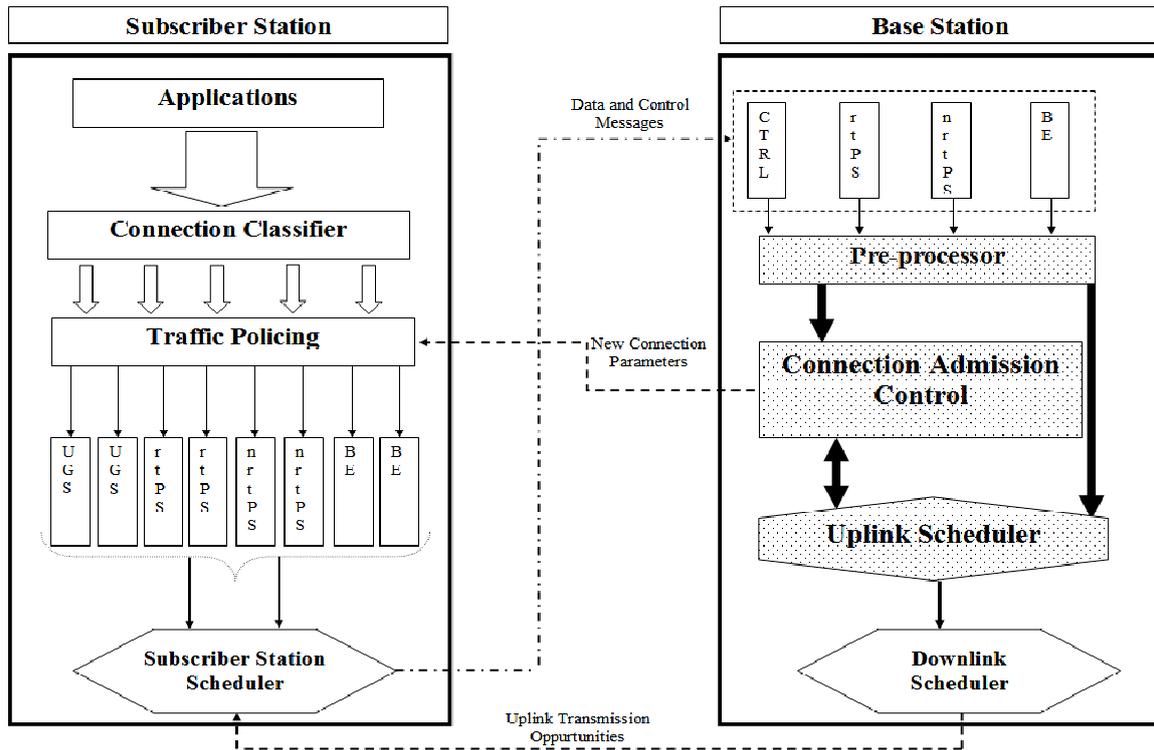
- Les algorithmes homogènes : dans ce cas un algorithme classique est assigné pour toutes les classes de service et ne considère pas les caractéristiques variables des classes de service WiMax. Nous pouvons citer à titre d'exemple les algorithmes RR, DRR, WRR, and EDF.
- Les algorithmes hybrides : dans ce cas différents algorithmes sont utilisés pour différentes classes de services. Par exemple, EDF est utilisé pour la classe rtPS, WRR pour la classe nrtPS et RR pour la classe BE. Très souvent, ces algorithmes utilisent des solutions basées sur la priorité pour

l'allocation des ressources inter-classes [176] [177] [130] [136] [178]. Ensuite allouent de bande passante aux connexions BE seulement si la qualité de service des classes prioritaires sont toutes garanties. De nombreux travaux de recherches ont essayé d'apporter des réponses mais aucune n'a été retenue par le groupe de travail IEEE 802.16 et par conséquent, ce point est encore un point sur lequel se penchent de nombreuses personnes du domaine. C'est dans ce cadre que se situe mon travail de thèse.

3. Algorithme d'ordonnancement à 2 niveaux (TSLA)

Dans ce chapitre, nous présentons l'algorithme d'ordonnancement à deux niveaux intitulé TSLA pour «Two-Level Scheduling Algorithm». Nous décrivons également l'algorithme de contrôle d'accès coopérant également avec le premier pour assurer une qualité de service (QoS) pour différents classes de trafic. Pour cela différents composants essentiels au bon fonctionnement de l'architecture de QoS de la liaison montante, sont décrits (voir figure ci-dessous) à savoir :

- le pré-processeur (Pre-processor)
- les files de données et de contrôle (Data and Control queues)
- le module de contrôle de trafic
- le contrôle d'accès des connexions (CAC)
- l'ordonnanceur de la station de base sur la liaison montante
- l'ordonnanceur de la station d'abonné



Le *Pre-processeur* est le composant responsable du contrôle et du traitement des messages de requête de bande passante émanant de la station d'abonné. Il extrait les données nécessaires de ces messages pour être traitées par les composants suivants. Ces données sont contenues dans une structure particulière appelée DSA. On trouve en particulier l'identificateur de la station d'abonné, de la classe de service, taux minimum de trafic requis, etc. Il est également responsable de l'extraction d'information des messages de requêtes de bande passante afin de déterminer la taille ainsi que le délai toléré des paquets arrivés dans la trame MAC. Si l'on considère que ρ_i est la taille de la file de données de la connexion i arrivées au début de la trame f et Y_i est la taille de la file de données arrivées au début de la trame $f-1$ alors nous pouvons écrire que la taille du trafic arrivé durant la trame $f-1$ est :

$$\xi_i[f-1, f] = (\rho_i[f] - \rho_i[f-1]) + Y_i[f-1, f] \quad (1.1)$$

Pour une connexion temps réel, l'ordonnanceur de la station d'abonné détruit tous les paquets dont le délai toléré a expiré. L'équation précédente devient donc :

$$\xi_i[f-1, f] = (\rho_i[f] - \rho_i[f-1]) + Y_i[f-1, f] + d_i[f-1, f] \quad (1.2)$$

La *gestion des files* est nécessaire aussi bien au niveau de la station d'abonné qu'au niveau de la station de base. Dans la station d'abonné, une file d'attente est associée à chaque connexion i . Alors que dans la station de base, toutes les requêtes de bande passante

appartenant à la même classe de trafic sont affectées à une même file en attente d'être traitées par l'ordonnanceur de la liaison montante. Ainsi l'algorithme d'ordonnement intra-classe est simplifié. Chaque nouvelle requête est ajoutée en queue de file associée et les requêtes sont traitées par l'ordonnanceur dans un ordre FIFO. Cependant, l'ordre de transmission est déterminé par la classe de trafic. Autrement dit, la file rtPS est traitée en premier ensuite nrtPS et la file BE en dernier respectant ainsi le niveau de QoS associé à chacune d'elles.

Le *contrôle de trafic* est un module qui permet d'assurer la surveillance du trafic émanant de chaque application. Il doit garantir que celle ci ne dépasse pas le taux spécifié lors de l'établissement de la connexion et qu'il soit conforme au contrat de trafic. Un lissage du trafic peut être opéré au niveau des connexions concernées par le dépassement pour éviter qu'il ne soit détruit.

Le *contrôle d'accès des connexions (CAC)* est le module qui décide d'accepter ou non une nouvelle demande de connexion. La décision est prise sur la base des exigences de qualité de service à la fois des connexions entrantes et existantes, et disponibilités des ressources de réseau. L'algorithme CAC proposée admet une nouvelle connexion uniquement si les deux conditions suivantes sont satisfaites :

(i) Les garanties demandées sur le délai et le taux de trafic minimum pourraient être fournies pour la connexion entrante. (ii) Les délais et les débits garantis pour les connexions existantes ne sont pas violés. Dans le cas où toute la bande passante montante est utilisé par les connexions établies, nous proposons d'utiliser la technique de vol de bande passante pour accepter une nouvelle connexion. Cependant, la technique doit respecter les garanties sur les niveaux de service minimum de connexions existantes. Autrement dit, si une nouvelle connexion arrive lorsque toutes les ressources du réseau sont en cours d'utilisation, le module CAC peut prendre les ressources des connexions déjà établies de sorte qu'au moins le niveau de service minimum pourrait être garanti à la fois à la nouvelle et à toutes les connexions existantes. Si le niveau de service minimum ne pouvait être garanti, la connexion est rejetée. Le vol de bande passante se fait toujours dans l'ordre BE, nrtPS et rtPS. La bande passante est d'abord prise de la classe BE jusqu'à ce que la bande passante totale allouée à cette classe soit réduit au minimum. Ensuite, on fait de même pour la bande passante de la classe nrtPS. Enfin, si il ya encore un besoin de bande passante pour les nouvelles connexions, on pourra la récupérer de la classe rtPS tant que les conditions (i) et (ii) sont satisfaites. On ne pourra pas « voler » de bande passante des classes UGS et ertPS.

Ci dessous la condition nécessaire qui pourrait être utilisée afin de déterminer s'il est possible de répondre aux exigences de retard d'une connexion rtPS :

$$\alpha_i'' = \left(\frac{\delta_i}{\gamma} - 1 \right) \left((\beta - \beta') - \psi' \sum_{j \in \Delta_{rtPS-i}} \alpha_j' \right) \quad (1.3)$$

ou,

$$\beta' = \sum_{k \in \Delta_{UGS}} \alpha_k + \sum_{m \in \Delta_{ertPS}} \alpha_m + \sum_{n \in \Delta_{nrtPS}} \alpha_n + \beta_{BE} \quad (1.4)$$

Voici pour chaque type de service, la procédure de contrôle d'admission associée :

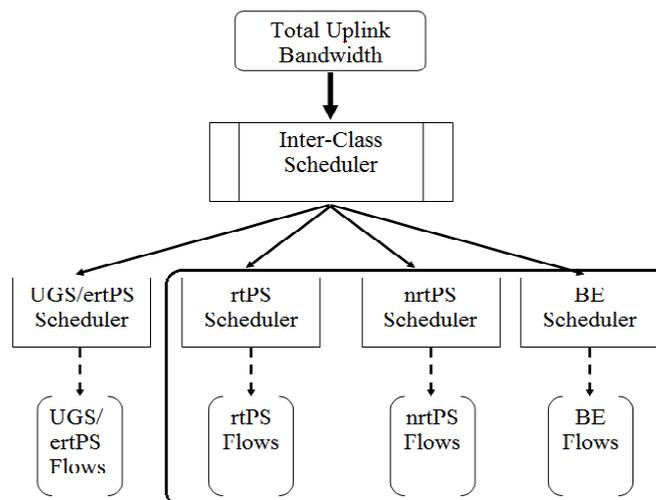
La *classe BE* : cette classe n'exige aucune garantie de débit et de délai. Par conséquent, le module CAC accepte toujours une connexion BE entrante.

La *classe nrtPS* : les connexions nrtPS exigent un débit minimum. Le module CAC admet une connexion nrtPS i , si la condition de l'équation 1.3 reste valable pour les connexions établies nrtPS et la condition donnée par l'équation 1.4 est satisfaite. Sinon, la demande est rejetée.

La *classe rtPS* : une connexion rtPS nécessite des garanties à la fois de taux de trafic minimum et de délai maximum. Si une nouvelle connexion (m) arrive, alors le module CAC opère comme suit :

1. Si le MRTR demandée est inférieure à la bande passante disponible alors aller à l'étape 2, sinon la connexion est refusée.
2. Si l'équation 1.3 est respectée pour la connexion entrante alors aller à l'étape 3 sinon la connexion est refusée.
3. Si l'équation 1.3 est respectée pour les connexions rtPS existantes alors accepter la connexion sinon la demande est rejetée.

L'*algorithme d'ordonnancement à 2 niveaux* : dans le but d'ordonner le trafic en tenant compte des différents besoins en QoS et de la priorité associés à chaque trafic, il est fourni un algorithme d'ordonnancement pour la liaison montante de la station de base. Au niveau niveau, l'algorithme inter-classe distribue la bande passante montante entre les différentes classes de service en fonction de leur exigences en qualité de service et des ressources disponibles. Le second niveau de l'algorithme concerne la distribution de la bande passante entre les connexions d'une classe donnée. Pour chacune des classe, un algorithme spécifique est proposé. Le principe de ce dernier est de distribuer la bande passante allouée par l'algorithme inter-classe, entre toutes les connexions actives appartenant à la même classe. Le schéma hiérarchique proposée dans la figure suivante :



Ci-après les détails des algorithmes d'ordonnement inter-classes et intra-classe.

- **L'algorithme d'ordonnement inter-classe** suit l'ordre de priorité des classes de service comme implicitement suggérée par la norme IEEE 802.16. Il veille à ce que des ressources suffisantes soient fournies pour chaque classe de service de sorte que les algorithmes d'ordonnement intra-classe peuvent assurer le niveau de service garanti pour chaque connexion. La répartition de la bande passante entre les classes de service se fait dans l'ordre UGS/ertPS, rtPS, nrtPS, et BE. Cela implique que les classes UGS et ertPS ont la plus haute priorité, alors que trafic BE la priorité la plus basse. Les principaux objectifs de l'algorithme d'ordonnement inter-classes sont comme suit :
 - Le niveau de qualité de service garantie par le module CAC est assurée pour toutes les classes de service ;
 - Les flux de priorité élevée ne sont pas affectés par les flux de faible priorité ;
 - Pas de famine pour aucune des classes de service ;
 - Une utilisation efficace de la bande passante.

L'ordonnement relatif aux classes UGS et ertPS est similaire et bien défini par la norme IEEE 802.16. Le standard spécifie une allocation fixe de la bande passante sur des intervalles périodique pour ces classes. Pour chaque connexion, cette allocation est égale à sa MRTR.

Pour la classe nrtPS, l'algorithme assure que suffisamment de bande passante est allouée à la classe de sorte que l'algorithme d'ordonnement intra-classe puisse fournir au moins la MRTR à chaque connexion. La largeur de bande minimale attribuée à la classe nrtPS est égale à la somme du minimum du taux de trafic de toutes les connexions nrtPS acceptée. Etant données que les connexions BE n'imposent pas un taux de trafic minimum, aucune allocation ne peut être faite dans ce cas. Néanmoins, pour empêcher la famine des connexions BE, l'algorithme se réserve une petite partie de la bande passante de la liaison montante pour cette classe de service.

- **L'algorithme d'ordonnement intra-classe**

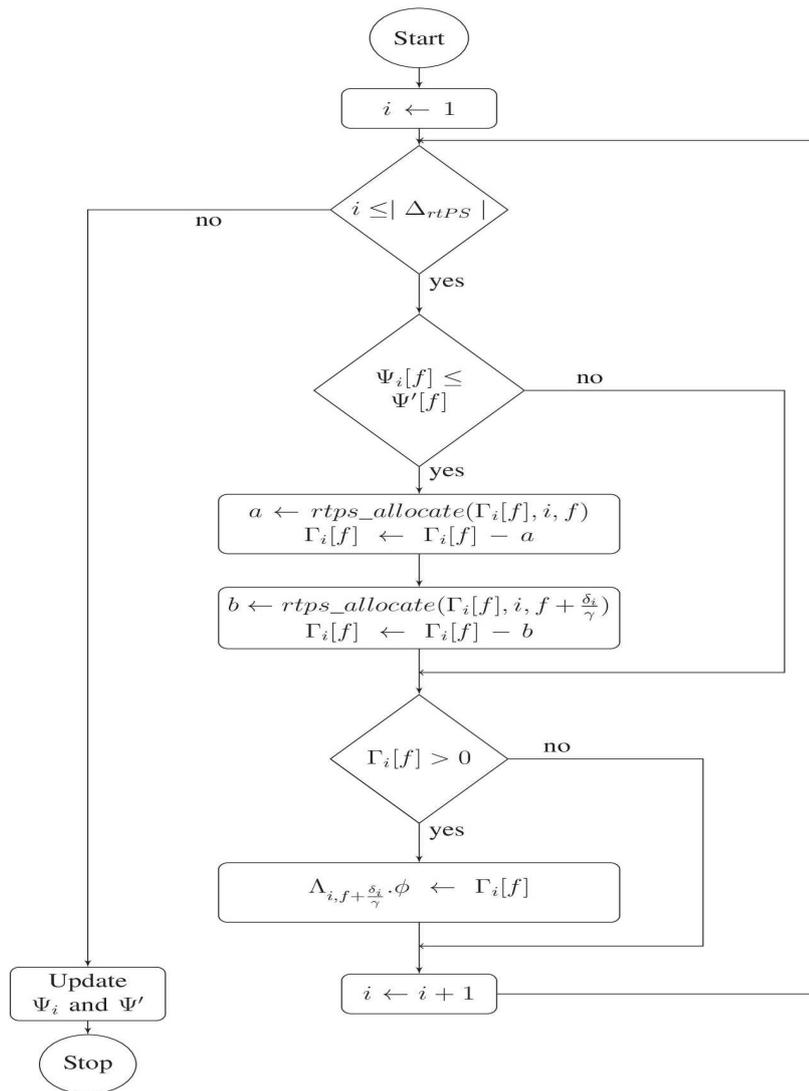
L'ordonnement relatif à la classe rtPS : nécessite pour sa description la présentation des paramètres utilisés par ce dernier permettant d'assurer l'équité d'allocation de bande passante entre les différentes connexions rtPS. Le principal paramètre que nous avons utilisé est le *taux de service*. Il est calculé pour chaque connexion et au début de chaque cycle de l'algorithme d'ordonnement en utilisant l'équation 1.5. Nous avons également considéré le paramètre *taux de service moyen* calculé comme le ratio du service total obtenu par toutes les connexions rtPS sur le total du service demandé par ces mêmes connexions (cf. équation 1.6). Au début d'une trame f , une connexion i est admissible à obtenir une bande passante de liaison montante si $i \leq f$ est vrai. Si $i > f$, ceci implique qu'il y a des connexions qui ont été moins servies que la connexion i , et par conséquent, ces connexions doivent avoir la priorité sur i . Ce qui doit être garanti et vérifié à chaque cycle de l'algorithme. Ainsi, l'utilisation de ces paramètres avec l'algorithme CAC garanti un taux de service minimum MRTR pour chaque connexion rtPS et une répartition équitable des

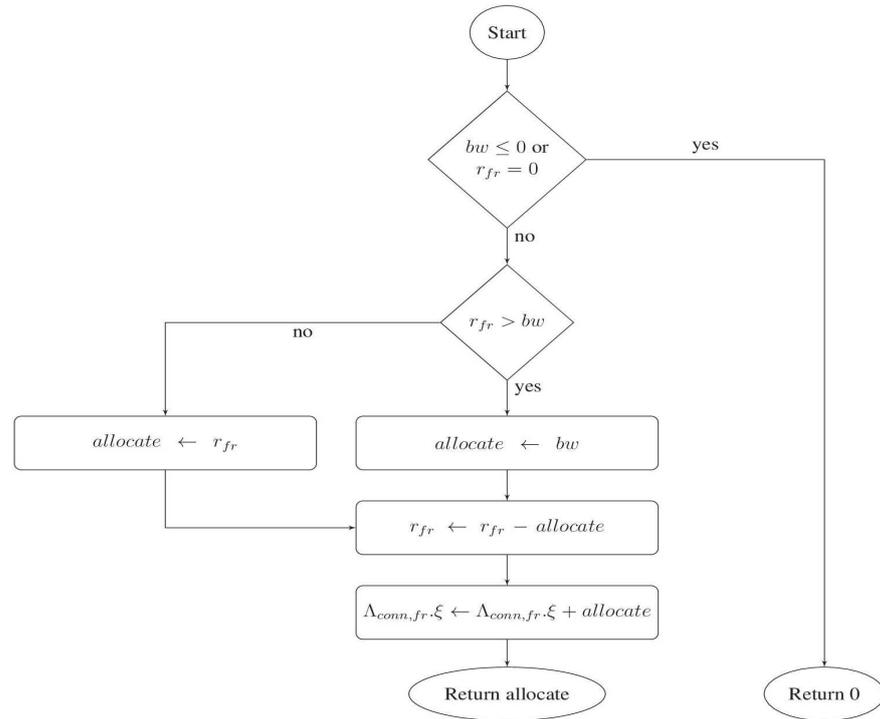
ressources au sein de la classe rtPS. Nous définissons ainsi l'allocation de bande passante *complètement équitable* si le taux de service de chaque connexion est égal au taux de service moyen.

$$\psi_i[f] = \frac{\sum_{t=1}^{f-1} Y_i[t]}{\sum_{t=1}^{f-1} \Gamma_i[t]} \quad (1.5)$$

$$\psi'[f] = \frac{\sum_{t=1}^{f-1} \sum_{i \in \Lambda_{rtPS}} Y_i[t]}{\sum_{t=1}^{f-1} \sum_{i \in \Lambda_{rtPS}} \Gamma_i[t]} \quad (1.6)$$

Maintenant que nous avons défini les paramètres d'équité, nous allons détailler l'algorithme d'allocation pour les connexions rtPS en tenant compte des délais tolérés de chacune d'elles. Pour respecter les délais des paquets rtPS et pour assurer un service déterministe pour les connexions rtPS, la station de base offre une possibilité de demande de bande passante périodiques pour ces connexions. Pour ces connexions, l'ordre de service est important si l'on doit tenir compte des délais tolérés et sachant que certaines ont une tolérance très faible. Par conséquent, la station d'abonné hébergeant les connexions rtPS sont interrogés dans l'ordre croissant de la latence maximale de connexions rtPS. De cette façon, la connexion qui a la plus faible valeur de latence maximale tolérable est interrogé en premier, ensuite la connexion avec la deuxième valeur plus faible et ainsi de suite. De cette façon, pour chaque trame de la liaison montante, la station de base reçoit et traite les demandes de bande passante dans l'ordre croissant des temps de latence acceptable. Cet ordre de traitement donne ainsi la priorité aux connexions avec des contraintes de délai très fort. L'algorithme est illustré par l'organigramme suivant :





L'ordonnancement relatif à la classe nrtPS : L'objectif principal de l'algorithme d'ordonnancement intra-classe pour la classe nrtPS est de garantir le taux de trafic minimum pour chaque connexion. Ensuite, l'algorithme alloue la bande passante résiduelle aux connexions nrtPS proportionnellement à la taille de leur file d'attente.

L'ordonnancement relatif à la classe BE : L'allocation de bande passante au niveau de la couche physique se fait en unités de temps. Une station d'abonné avec de mauvaises conditions de canal consomme plus d'unités de temps pour une quantité de données transmises relativement plus petite. D'autre part, un poste d'abonné avec de bonnes conditions de canal peut envoyer relativement plus de données avec le même nombre d'unités de temps. Par conséquent, pour améliorer l'efficacité de l'utilisation des ressources radio, nous proposons de répartir les créneaux les unités de temps de manière équitable entre les connexions BE. Cet ordonnancement évite donc aux stations d'abonnés avec un canal de mauvaises qualité d'affecter l'ensemble du réseau et dans le même temps évite la famine de ces stations d'abonnés.

†

Modèle analytique proposé : Nous proposons dans ce travail de thèse un modèle analytique qui n'est pas trivial pour les raisons de contexte suivantes :

- L'algorithme proposé se compose de deux niveaux. Au second niveau, chaque classe de service est géré par un algorithme d'ordonnancement différent.
- Chaque paquet dans la file d'attente de données rtPS dispose d'un délai associé.

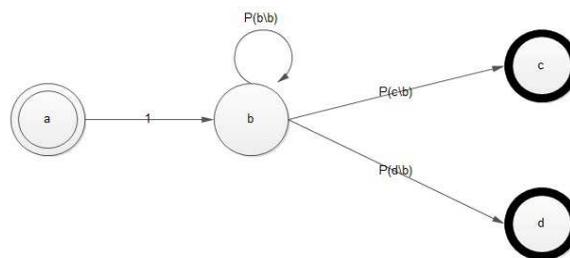
Un paquet est détruit s'il le délai toléré expire.

- La bande passante disponible pour chaque algorithme d'ordonnancement intra-classe dépend de la répartition de la bande passante par l'algorithme d'ordonnancement inter-classes.

Nous avons proposé un modèle de file d'attente pour lequel nous avons considéré plusieurs hypothèses que nous résumons comme suit :

- Le taux d'arrivée des paquets est indépendant du nombre de paquets déjà dans la file d'attente.
- Toutes les files d'attente sont de taille infinie.
- Pour chaque file d'attente, les paquets sont servis dans l'ordre FIFO.
- Paquets arrivent de façon aléatoire et indépendante l'un de l'autre. Ce qui correspond à la loi de Poisson et le temps de service suit une distribution exponentielle. Ainsi, une file d'attente peut être modélisée comme un système M/M/1. Le système global peut être modélisée comme un système de M/M/c.

Dans la norme IEEE 802.16, les données arrivent sous la forme de paquets et les paquets sont stockés dans les files d'attente et ordonnancés sur la base d'une trame. Pour analyser le système, il est important de comprendre le cycle de vie d'un paquet de données. Un paquet peut être dans l'un des quatre états comme le montre la figure suivante :



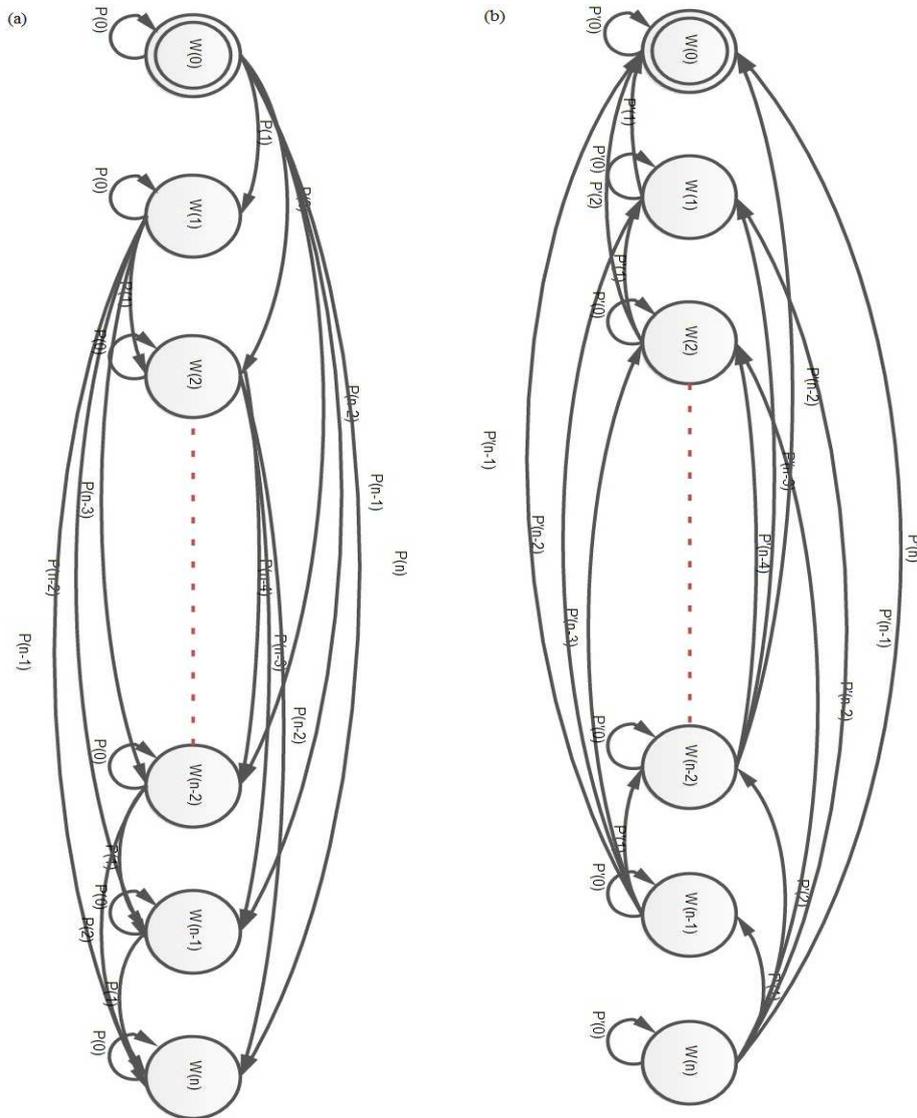
L'état 'a' correspond à la création d'un paquet. L'état 'b' correspond à l'état du paquet en attente dans la file d'attente. Nous supposons que les files d'attente sont de taille infinie, donc pas de rejet de paquet pour cause de manque de place dans la file. Une liaison orientée signifie une transition d'état avec une probabilité associée $P(y|x)$. $P(y|x)$ désigne la probabilité de passer à l'état y sachant que le paquet est dans l'état x . Notez que dans le diagramme d'état, $P(b|a) = 1$. Un paquet, qui est dans un état 'b' a trois transitions possibles: (i) Il reste dans le même état d'attente pour le service avec une probabilité $P(b|b)$ (ii) Il obtient un service (état 'c') avec une probabilité $P(c|b)$ (iii) Il est détruit pour cause du dépassement du délai toléré (état 'd') avec une probabilité $P(d|b)$. Pour les paquets non temps réel, il n'y a pas de paramètre délai limité associé, ce qui implique que $P(d|b) = 0$.

La figure suivante (a) montre le diagramme de transition vers l'état b sachant que la fonction de distribution des paquets est donné par l'équation suivante :

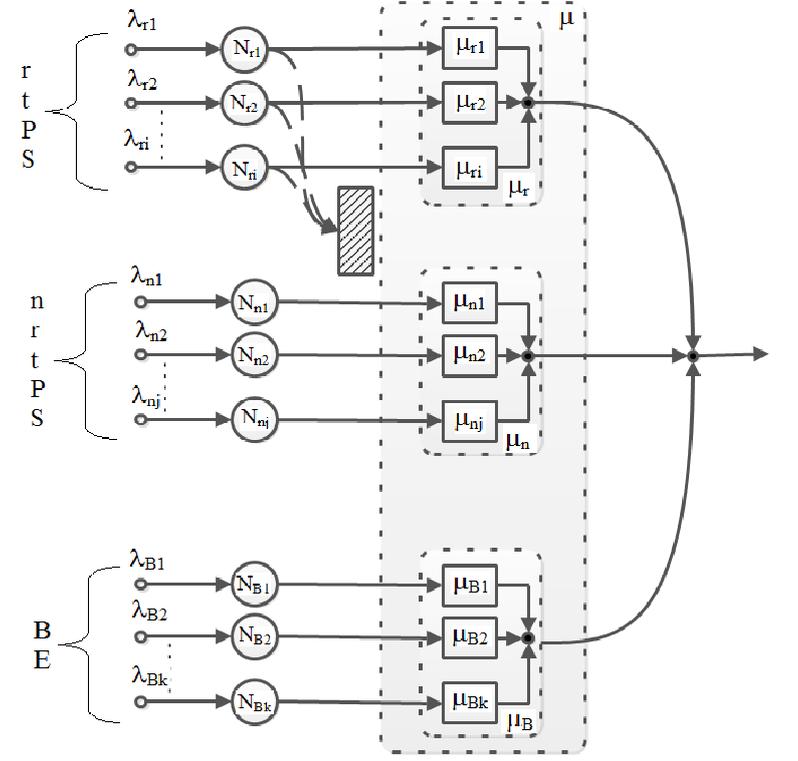
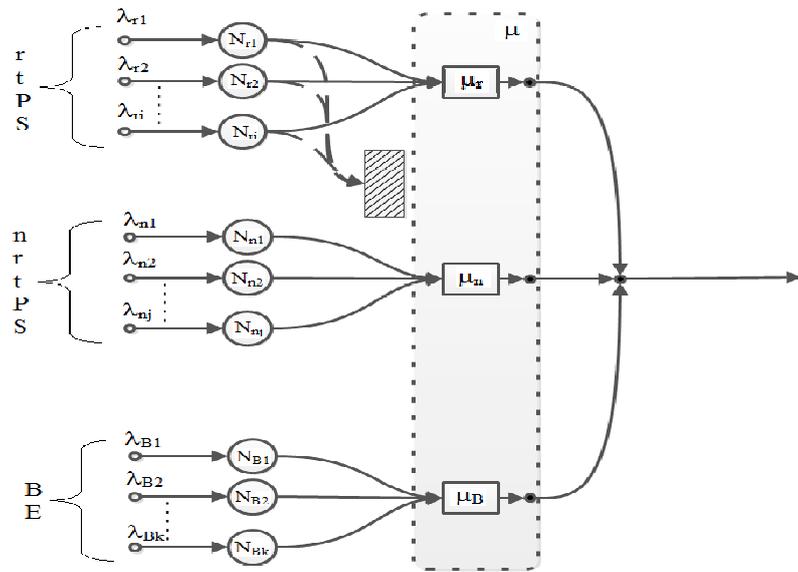
$$P(z) = \left(\frac{\lambda \gamma}{z!} \right) e^{-\lambda z}$$

De manière similaire ci-dessous la figure (b) montrant la transition de 'b' vers 'c' ou 'd'. Notons que la probabilité $P(x)$ correspond soit à la probabilité de rejets de x paquets ou à la

probabilité que les x paquets aient été servis.



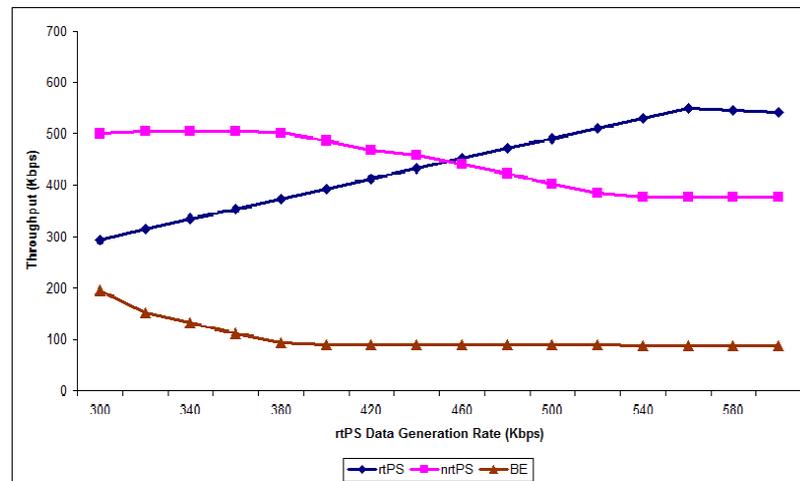
Ci dessous les figures représentant le modèle de files d'attente représentant l'algorithme d'ordonnancement inter-classes et intra-classe :



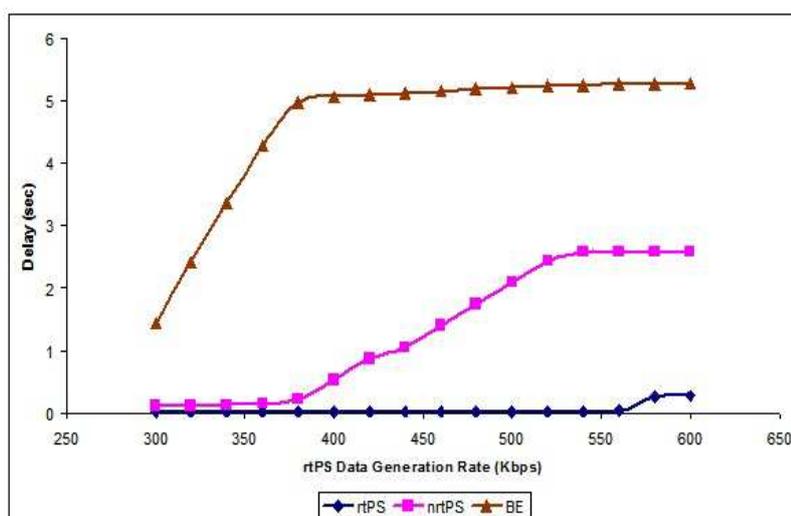
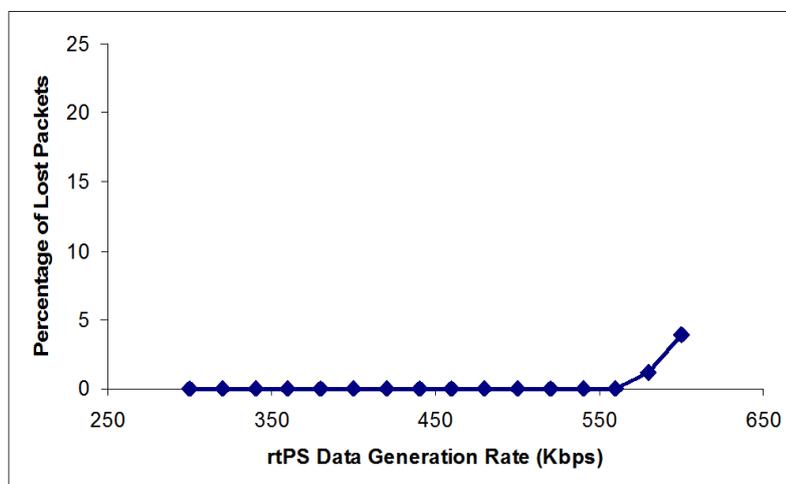
Evaluation de performances des algorithmes proposés : une étude de performance a été réalisée afin d'analyser et de valider nos propositions. Cette étude a été réalisée grâce à l'outil de simulation Qualnet. C'est une version commerciale de l'outil GloMoSim développé par la société SNT « Scalable Network Technologies ». Les différents indicateurs de performance utilisés pour analyser la performance l'algorithme d'ordonnancement proposé sont les suivants :

- Débit : le total des unités de données transmises dans la durée T divisé par T.
- Le délai de bout-en-bout : le délai moyen observé par paquets de la source à la destination. Il comprend délai d'attente et les délais de propagation.
- Taux de perte de paquets : Le nombre de paquets perdus sur le nombre total de paquets transmis par la source.
- Équité : l'équité dans l'allocation des ressources
- Evolutivité (scalability): la variation de performance de l'algorithme lorsque le nombre de stations d'abonnés augmente.

Nous allons montrer dans un premier temps les résultats de performance relatives à l'algorithme inter-classes. Pour ce faire, le trafic BE est généré avec un taux moyen de 200 Kbps tout au long de l'expérience. Afin d'éviter la famine de la classe BE, la valeur de son taux de trafic minimum est fixé à 90 Kbps. Alors que pour la classe nrtPS, le MRTR est de 375 Kbps et le taux de trafic moyen est de 500 Kbps. Des simulations sont effectuées en fonction de l'augmentation de la charge de trafic rtPS. Initialement, le taux de trafic moyenne de la classe rtPS est de 300 Kbps. Il est progressivement augmenté à 600 Kbps. Le MRTR pour le trafic rtPS est de 300 Kbps tout au long de l'expérience, alors que le délai maximum autorisé est fixé à 160 ms. La première figure montre la distribution de la bande passante entre les 3 classes de trafic rtPS, nrtPS et BE qui respecte l'attribution minimum pour chaque classe. La seconde figure montre le taux de perte des paquets en fonction de la génération de trafic de type rtPS. Cette figure montre que le taux de perte est négligeable tant que la demande n'excède pas la bande passante totale. Au delà, la perte est due essentiellement à l'expiration du délai de tolérance des paquets de type rtPS. La troisième figure montre le délai de bout en bout pour les 3 classes de trafic qui ne dépasse pas les 30 ms pour la classe rtPS mais ce délai augmente lorsque la génération de trafic de type rtPS



augmente. Dans ce cas, la perte augmente et donc le délai augmente également pour nrtPS et BE.



Nous avons également analysé les performances dans des conditions de mobilité des stations. Les résultats sont également intéressants et garantissent un taux de perte minimum ainsi qu'un délai très acceptable pour le trafic temps réel.

Des simulations sont également réalisées afin d'évaluer les performances de l'algorithme d'ordonnancement intra-classe. Nous avons réalisé cette étude pour le trafic nrtPS, BE et rtPS.

Regardons dans un premier temps les résultats pour le trafic nrtPS. Pour ce faire, quatre connexions nrtPS sont utilisées pour cette simulation. Les résultats montrent que le taux minimum est garanti pour toutes les connexions et que le taux de service moyen pour chacune est respectivement de 0.99, 0.99, 0.98, 0.98. Par conséquent, le délai moyen pour

chaque connexions est identique.

Nous avons également utilisé quatre connexions de type BE pour analyser les performances relatives à cette classe. Le taux de génération de paquets est respectivement de 200 Kbps, 225 Kbps, 275 Kbps et 300 Kbps. Une distribution équitable est assurée par cet algorithme cependant pour les deux dernières connexions n'obtiennent pas la totalité de la bande passante demandée étant donné que la répartition par l'algorithme CAC ne permet pas de satisfaire la somme des débits demandés par les quatre connexions. Le délai de bout en bout est par conséquent plus important pour les 2 dernières connexions et en particulier pour la quatrième.

Une dernière étude de simulation dans le cadre de l'analyse de l'algorithme d'ordonnancement intra-classe concerne les connexions de type rtPS. Nous avons considéré quatre connexions dont chacune dispose de 4 paramètres essentiels à savoir le MRTR, le MSTR, le délai maximum toléré. Les valeurs sont résumés dans le tableau suivant :

Connexions	MRTR (Kbps)	MSTR (Kbps)	Délai max toléré (trames)
A	4000	9000	2
B	1000	3000	3
C	2000	4000	3
D	3000	5000	4

Les résultats de simulations montrent que la répartition de la bande passante entre ces connexions est dynamique et équitable et chacune obtient le maximum possible de bande passante. Nous avons par la suite comparé le taux de service obtenu par notre algorithme et celui obtenu par l'algorithme EDF. Le résultat montre que le taux de service moyen est identique dans les deux algorithmes mais le taux de service associé à chaque connexion est quasi constant pour le TLSA mais pour EDF ce taux est plus important pour la connexion A qui est servi en premier. Cette dispersion en terme de taux de service s'explique par le fait que l'algorithme EDF a pour principal but de minimiser le délai et ne prend pas en compte le paramètre équité entre les connexions. D'autres critères de performances (le débit, de taux de perte, etc) sont utilisés pour comparer les deux algorithmes et les résultats montrent que TLSA présente de meilleurs performance lorsque le taux de génération de paquets est plus important.

4. Application au transfert vidéo

Le streaming vidéo est l'application la plus importante de la classe rtPS. Avec le développement des réseaux sans fil, la transmission de la vidéo sans fil est de plus en plus d'actualité. La convivialité et la flexibilité de ces systèmes ont permis plusieurs nouveaux services vidéo tels que : la téléphonie vidéo à l'aide d'appareils portables, le vidéo streaming avec les téléphones mobiles, la surveillance en temps réel par le personnel paramédicaux

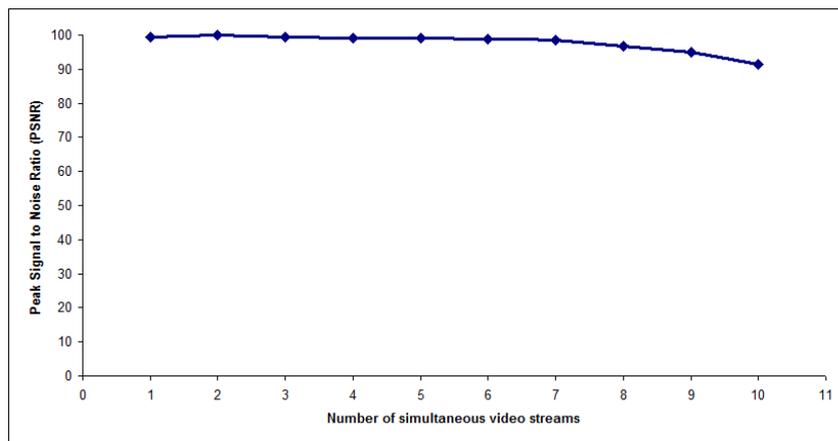
dans le service de réanimation. Le service vidéo sans fil est maintenant bien intégré dans la société et continuera à se répandre.

L'objectif de ce chapitre est d'étudier la performance de l'algorithme d'ordonnement intra-classe pour le trafic rtPS dans le cas de la transmission de flux vidéo. Le flux vidéo est compatible avec la norme de codage H.264.

Dans notre étude, nous avons fait usage de traces vidéo ainsi que d'une suite binaire. Les séquences vidéo sont dans un format « yuv ». Tout d'abord, le système code la séquence vidéo brut dans H.264/SVC bitstream en utilisant le logiciel de référence vidéo JSVM. Puis, « BitStreamExtractor », qui est un outil de génération de flux de bits dans la bibliothèque JSVM, est utilisé pour générer une trace de paquets. Cette trace, spécifie l'ordre, la taille et les paramètres de QoS de chaque paquet du train de bits. Un échantillon trace de paquets est illustré dans la figure suivante :

Start-Pos.	Length	LId	TId	QId	Packet-Type	Discardable	Truncatable
0x00000000	262	0	0	0	StreamHeader	No	No
0x00000106	14	0	0	0	ParameterSet	No	No
0x00000114	16	0	0	0	ParameterSet	No	No
0x00006fd4	238	1	0	0	SliceData	No	No
0x000070c2	9	0	1	0	SliceData	Yes	No
0x000070cb	979	0	0	0	SliceData	No	No
0x0000749e	9	0	1	0	SliceData	Yes	No
0x000074a7	121	0	0	0	SliceData	No	No
0x00007520	976	1	1	0	SliceData	Yes	No
0x000078f0	598	1	1	0	SliceData	Yes	No
0x00007b46	9	0	2	0	SliceData	Yes	No
0x00007b4f	679	0	0	0	SliceData	No	No
0x00007df6	979	1	2	0	SliceData	Yes	No
0x000081c9	9	0	2	0	SliceData	Yes	No
0x000081d2	768	0	0	0	SliceData	No	No
0x000084d2	983	1	2	0	SliceData	Yes	No
0x000088a9	97	1	2	0	SliceData	Yes	No

La simulation est réalisée pour évaluer la performance de l'algorithme d'ordonnement du trafic rtPS proposé représentant ici le trafic vidéo. Nous avons utilisé les séquences vidéo (videochat) obtenues à partir de la bibliothèque de trace vidéo de l'université d'Arizona. La simulation a été effectuée dans Qualnet avec un total de 20 stations d'abonnés et une station de base. Les 10 stations d'abonnées sont utilisées en tant que source, tandis que les 10 autres sont utilisées comme récepteurs. Le paramètre PSNR (Peak Service to Noise Ratio) est utilisé pour comparer la qualité de la vidéo reçue à celle transmise. Des valeurs élevées de PSNR implique que la qualité des vidéos reçues est bonne. La figure suivante montre le PSNR moyenne en fonction du nombre de flux de données vidéo simultanés. Le PSNR moyen est simplement la moyenne arithmétique de PSNR de tous les flux vidéo. On observe donc que les vidéos reçues ont des distorsions négligeables et sont presque identiques aux vidéos transmises. L'algorithme TLSA est donc efficace pour le transport du service vidéo.



Conclusion et travail futur

La norme IEEE 802.16 est une technologie d'accès haut débit sans fil qui prend en charge de nombreuses applications temps réel et non temps réel. Ces applications et services nécessitent une qualité de service différenciées. Sans mécanismes d'ordonnement de paquets appropriés, ces applications peuvent ne pas offrir une qualité acceptable pour les utilisateurs finaux. Dans la norme 802.16, cette tâche est complexe et est répartie en trois niveaux : l'ordonnanceur au niveau de la station de base pour la liaison montante, l'ordonnanceur au niveau de la station de base pour la liaison descendante et l'ordonnanceur au niveau de la station d'abonné. La norme spécifie les divers composants de qualité de service et de leurs fonctions. Toutefois, il ne détaille pas les algorithmes pour certains éléments, tels que l'ordonnement des paquets et le contrôle d'admission des connexions.

Dans cette thèse, nous avons fourni un algorithme d'ordonnement de paquets pour la liaison montante. L'algorithme se compose de deux niveaux. Dans le premier niveau, un

l'algorithme d'ordonnancement inter-classes distribue la bande passante entre les différentes classes de service selon à leurs exigences de qualité de service et des ressources disponibles. Ensuite, au deuxième niveau, l'algorithme à chaque classe prend en compte la bande passante allouée par l'algorithme inter-classe et la distribue entre les connexions de la classe concernée en fonction de leurs exigences en qualité de service. L'algorithme d'ordonnancement inter-classe implémente ce service en considérant l'ordre de priorité des classes tout en empêchant la famine des flux moins prioritaires. L'algorithme spécifique à une classe fait en sorte que chaque connexion reçoit au moins le minimum niveau de service garanti et que l'allocation des ressources soit équitable.

Nous avons également proposé un algorithme de contrôle d'admission des connexions qui accepte une connexion que si la qualité de service peut être garantie à la fois la connexion en question qu'aux connexions établies. Nous avons également proposé un modèle analytique basé sur les chaînes de Markov.

Les algorithmes proposés ont été validés par des simulations réalisées dans Qualnet 5.0. Nous pouvons conclure que les algorithmes d'ordonnancement proposés assurent efficacement un niveau requis de qualité de service et une garantie d'équité dans la répartition des ressources. Une étude comparative a été également menée avec l'algorithme EDF pour comparer la performance de l'algorithme intra-classe rtPS. Les résultats des simulations ont révélé que contrairement à l'algorithme EDF, la répartition faite par l'algorithme proposé est équitable. Par ailleurs, les algorithmes proposés donnent de meilleurs résultats pour les différents critères de performance tels que le débit, le délai et le taux de perte de paquets.

Nous avons également fourni un système de transmission vidéo compatible H.264 basé sur l'algorithme d'ordonnancement intra-classe rtPS proposé. Les différentes parties d'un flux vidéo H.264 ont une importance différente et donc une priorité différente. Par conséquent, la transmission des différentes couches d'un flux de données vidéo se fait en fonction de leurs caractéristiques. L'analyse de performance dans le cadre de la transmission vidéo a montré que l'algorithme d'ordonnancement transmet efficacement des flux vidéo.

Les supports physiques sans fils ont des caractéristiques changeant dans le temps et donc, ils sont variables. Par conséquent, il est important de tenir compte des informations sur l'état du canal les plus récentes. Cela peut améliorer le débit et les performances globales du système. Un algorithme d'ordonnancement idéal serait d'améliorer l'efficacité du réseau en utilisant des informations d'état de canal, tout en assurant une qualité de service à toutes les classes de service. Par conséquent, TLSA devrait être modifiée pour tenir du canal physique.

Un autre point important qui peut constituer une piste de recherche intéressante est de d'étudier la généralisation de TLSA dans les le cadre de réseaux hétérogènes. Un réseau hétérogène sans fil est constitué de plusieurs technologies d'accès radio et plusieurs architectures. Cette convergence dans le cadre hétérogène est considéré comme un enjeu majeur dans l'évolution des systèmes de télécommunications. Le groupe IEEE 802.21

travaille actuellement à établir un interfonctionnement en introduisant des médias indépendants Handover (MIH), qui pourrait fournir un handover entre les différents réseaux sans fil. Cette convergence de divers réseaux hétérogène des réseaux câblés et sans fil pourrait fournir une connectivité n'importe où et n'importe quand en utilisant la technologie disponible la plus appropriée. Il s'agit d'un moyen efficace de renforcer la capacité du réseau mobile et Trois scénarii les plus importants d'interfonctionnement concernent WiMAX-WiFi, WiMAX-3G, et WiMAX-LTE.

WiFi et WiMAX ont été optimisés pour différents modèles d'utilisation. WiFi est conçu pour offrir une connectivité locale haut débit sans fil, tandis que WiMAX pour une connectivité à haute vitesse. La combinaison de la technologie WiMAX et Wi-Fi est un choix idéal pour fournir une connectivité efficace et abordable à large bande à plusieurs endroits. WiMAX-WiFi haut débit permet l'interfonctionnement cohérente et conviviale pour les utilisateurs finaux.

Actuellement, le WiMAX fournit des taux de transfert de données très élevés, mais il a une zone de couverture limité par rapport aux réseaux cellulaires 3G. L'interfonctionnement WiMAX-3G peut fournir une connectivité omniprésent des données à grande vitesse donc des conditions très intéressante pour l'utilisateur.

Long Term Evolution (LTE) est une norme de communication sans fil, qui est basée sur les technologies réseau GSM/EDGE et UMTS/HSPA. Le but de la technologie LTE est d'offrir des niveaux de performance telque spécifié par les spécifications IMT-Advanced pour les réseaux 4G. Un interfonctionnement entre WiMAX et LTE est considérée comme l'option en vue de la réalisation de réseaux de données 4G. Toutefois, les deux technologies ont différentes architectures de qualité de service. Par conséquent, le mapping entre ces deux technologies serait nécessaire pour réaliser l'interfonctionnement.

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List of Publications

Z. Ahmed and S. Hamma, "Efficient and fair scheduling of rtPS traffic in IEEE 802.16 point-to-multipoint networks", 4th Joint IFIP/IEEE wireless and mobile networking conference (WMNC), Toulouse, France, 2011.

Z. Ahmed and S. Hamma, "Two-level scheduling algorithm for different classes of traffic in WiMAX networks", IEEE Int. Symp. on Performance Evaluation of Computer and Telecommunication Systems (SPECTS), Genoa, Italy, pp.90-96, 2012.

Z. Ahmed and S. Hamma, "Fair uplink scheduling for rtPS traffic in IEEE 802.16 networks", First Sino-Franco Workshop on Education and Research Collaborations in Information and Communication Technologies (SIFWICT), Nantes, France, 2011.

Z. Ahmed and S. Hamma, "Fair uplink scheduling for rtPS traffic in IEEE 802.16 broadband wireless networks", Colloque Francophone sur l'Ingénierie des Protocoles (CFIP), Sainte-Maxime, France, 2011.

Thèse de Doctorat

Zeeshan Ahmed

Quality of Service in WiMAX for Multimedia Services

Qualité de Service en WiMAX pour les Services Multimédias

Résumé

Le standard IEEE 802.16 est une importante technologie d'accès haut débit sans fil. Il prend en charge de nombreuses applications avec des besoins de qualité de service (QoS) très différents demandant des mécanismes efficaces et une bonne architecture de QoS. Toutefois, certains détails algorithmiques tels que l'ordonnancement et les mécanismes de contrôle d'admission (CAC) restent libres. Nous proposons un algorithme à deux niveaux (TLSA) pour l'ordonnancement des paquets en voie montante. Au premier niveau, un algorithme d'ordonnancement inter-classe distribue la bande passante entre les différentes classes de trafic en fonction de leurs besoins en QoS et des ressources disponibles. Cet ordonnancement utilise le niveau de priorité des classes de service en évitant la famine des flux de faible priorité. Au second niveau et pour chaque classe de service, des algorithmes spécifiques distribuent la bande passante entre les connexions appartenant à cette classe. Ces algorithmes s'assurent qu'au moins le niveau de service garanti est fourni à chaque connexion et que l'allocation de bande passante est équitable. Nous proposons également un schéma de CAC qui facilite les opérations de notre algorithme d'ordonnancement. Nous avons développé un cadre de transmission vidéo (VTF) basé sur le TLSA permettant de transmettre des flux vidéo H.264 sur des réseaux 802.16. L'implémentation des algorithmes a été réalisée sur QualNet et une évaluation de performances a été réalisée. Les résultats de simulation valident l'efficacité et l'équité des algorithmes proposés. Par ailleurs, les résultats révèlent que VTF peut transmettre efficacement les flux vidéo.

Mots clés

IEEE 802.16, WiMAX, QoS, Ordonnancement

Abstract

The IEEE 802.16 standard is an important broadband wireless access technology. It supports variety of applications with different, often conflicting, QoS needs. Therefore, effective QoS mechanisms are important in 802.16 networks. The standard incorporates a QoS architecture that specifies various QoS components and their functions. However, it does not specify the algorithmic details of some important components, such as packet schedulers and connection admission control (CAC). In this dissertation, we provide a two-level scheduling algorithm (TLSA) for uplink packet scheduling. At the first level, an inter-class scheduling algorithm distributes bandwidth among various classes of traffic according to their QoS needs and available resources. The inter-class scheduling algorithm enforces service class priority order and prevents starvation of lower priority flows. Then at the second level, various class-specific algorithms distribute bandwidth among connections of the associated class. The class-specific algorithms make sure that at least the guaranteed service level is provided to each connection and the bandwidth allocation is fair. We also propose a CAC scheme and several components that facilitate the operations of the proposed scheduling algorithm. Furthermore, we developed a TLSA based video transmission framework (VTF) that can transmit H.264 video streams over 802.16 networks. The proposed algorithms were extensively validated by simulations in Qualnet. The simulation results validate the efficiency and fairness of the proposed algorithms. Furthermore, the results reveal that VTF can efficiently transmit video streams.

Key Words

IEEE 802.16, WiMAX, QoS, Packet Scheduling