

**A Performance Study of Uplink Scheduling Algorithms
in
Point to Multipoint WiMAX Networks**

by

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A thesis submitted to the School of Computing
in conformity with the requirements for
the degree of Master of Science

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Abstract

Applications such as video and audio streaming, online gaming, video conferencing, Voice over IP (VoIP) and File Transfer Protocol (FTP) demand a wide range of QoS requirements such as bandwidth and delay. Existing wireless technologies that can satisfy the requirements of heterogeneous traffic are very costly to deploy in rural areas and “last mile” access. Worldwide Interoperability for Microwave Access (WiMAX) provides an affordable alternative for wireless broadband access supporting a multiplicity of applications. The IEEE 802.16 standard provides specification for the Medium Access Control (MAC) and Physical (PHY) layers for WiMAX. A critical part of the MAC layer specification is scheduling, which resolves contention for bandwidth and determines the transmission order of users.

It is imperative for a scheduling algorithm to have a multi-dimensional objective of satisfying QoS requirements of the users, maximizing system utilization and ensuring fairness among the users. In this thesis, we categorize and study various scheduling algorithms for the uplink traffic in WiMAX in view of these objectives. The algorithms are studied under different mixes of traffic and for various characteristics of the IEEE 802.16 MAC layer such as uplink burst preamble, frame length, bandwidth request mechanisms etc. Simulation results indicate that legacy algorithms are not suitable for the multi-class traffic in WiMAX as they do not explicitly incorporate the WiMAX QoS parameters. We provide recommendations for enhancing existing scheduling schemes in WiMAX, and shed light on some of the open issues that need to be addressed.

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List of Acronyms

AAS	Adaptive Antenna System
AMC	Adaptive Modulation and Coding
AMR	Adaptive Multi-Rate
ARQ	Automatic Repeat Request
ATDD	Adaptive Time Division Duplex
ATM	Asynchronous Transfer Mode
BE	Best Effort
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
BS	Base Station
CBR	Constant Bit Rate
DCD	Downlink Channel Descriptor
DHCP	Dynamic Host Configuration Protocol
DRR	Deficit Round Robin
DSL	Digital Subscriber Line
EDF	Earliest Deadline First
ertPS	extended real-time Polling Service
ETSI	European Telecommunication Standards Institute
FBWA	Fixed Broadband Wireless Access
FDD	Frequency Division Duplex
FEC	Forward Error Correction

GPC	Grant Per Connection
GPS	Generalized Processor Sharing
GPSS	Grant Per Subscriber Station
HiperMAN	High performance radio Metropolitan Area Network
HRR	Hierarchical Round Robin
HOL	Head Of Line
IE	Information Element
IEEE	Institute of Electrical and Electronic Engineers
LOS	Line-of-Sight
MAC	Medium Access Control
MAN	Metropolitan Area Network
MIB	Management Information Base
MRTR	Minimum Reserved Traffic Rate
MSTR	Maximum Sustained Traffic Rate
MUFSS	Multi-class Uplink Fair Scheduling Structure
MWFQ	Modified Weighted Fair Queuing
MWRR	Modified Weighted Round Robin
NLOS	Non-Line-of-Sight
nrtPS	non real-time Polling Service
OFDM	Orthogonal Frequency Division Multiplex
PDU	Protocol Data Unit
PFS	Proportional Fair Scheduler
PHY	Physical Layer

PMP	Point to MultiPoint
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RLC	Radio Link Control
RR	Round Robin
rtPS	real-time Polling Service
SC	Single Carrier
SCa	Single Carrier access
SDMA	Spatial Division Multiple Access
SDU	Service Data Unit
SINR	Signal to Interference Noise Ratio
SLA	Service Level Agreement
SNMP	Simple Network Management Protocol
SS	Subscriber Station
STC	Space Time Coding
TDD	Time Division Duplex
TFTP	Trivial File Transfer Protocol
UCD	Uplink Channel Descriptor
UGS	Unsolicited Grant Service
UNNI	Unlicensed National Information Infrastructure
VoIP	Voice over Internet Protocol
VWRR	Variably Weighted Round Robin

WFQ	Weighted Fair Queuing
WF ² Q	Worst-case Fair Weighted Fair Queuing
WiBRO	Wireless Broadband
WiMAX	Worldwide Interoperability Microwave Access
WirelessHUMAN	Wireless High Speed Unlicensed Metro Area Network
WirelessMAN	Wireless Metropolitan Area Network
WRR	Weighted Round Robin

Chapter 1. Introduction

WiMAX (Worldwide Interoperability for Microwave Access), is a cell-based technology aimed at providing last-mile wireless broadband access at a cheaper cost. The “last mile” is the final leg of delivering connectivity from the service provider to the customer. This leg is typically seen as an expensive undertaking because of the considerable costs of wires and cables. The core of WiMAX technology is specified by the IEEE 802.16 standard that provides specifications for the Medium Access Control (MAC) and Physical (PHY) layers. The term WiMAX was created by the WiMAX forum that promotes conformance and interoperability of the standard. Wireless Broadband (WiBro) is a technology developed by the Korean telecommunications industry that mirrors the specifications of the IEEE 802.16 standard. Efforts are already underway to define interoperability of WiMAX and WiBro equipment.

Broadband wireless networks (including WiMAX) can be categorized into a single-hop, Figure 1-1 or a multi-hop network Figure 1-2. A single-hop network contains a central entity such as a Base Station (BS) that makes and delivers decisions to all the nodes such as Subscriber Stations (SSs) in its cell. A cell is a basic geographic unit of a cellular system. On the other hand, in a cellular multi-hop network, some SSs are not in direct contact with the BS. For example, an object such as a building could be blocking the path from the BS to the SS. In such a network, a relay mechanism is required at intermediate SSs that will relay information to other SSs that are not in direct contact with the BS.

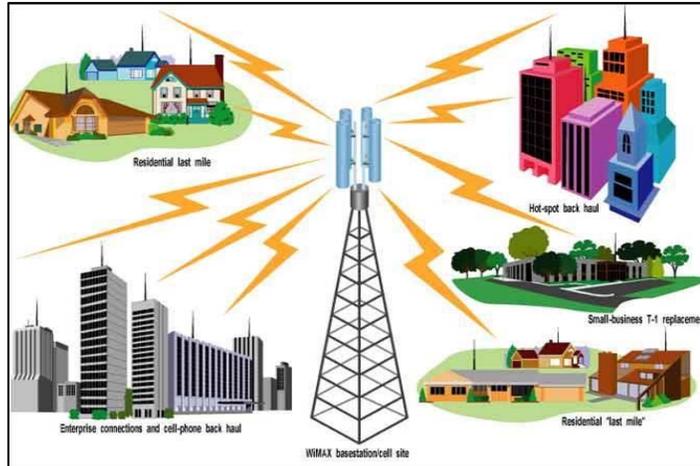


Figure 1-1: Single-hop cellular network

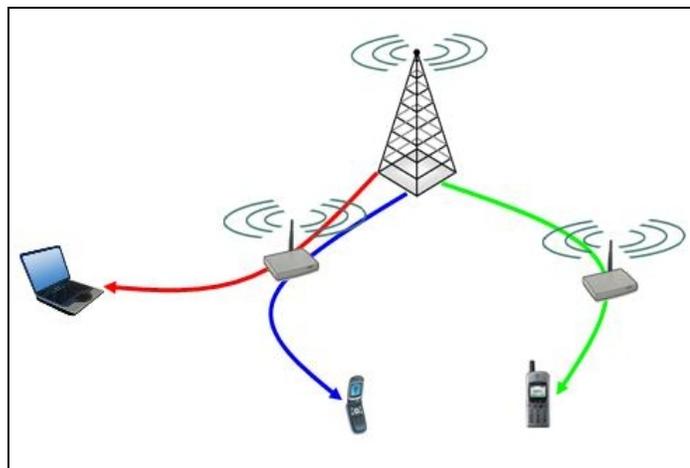


Figure 1-2: Multi-hop cellular network

In a cellular network such as WiMAX, traffic from the BS to the SSs is classified as downlink traffic while that from the SSs to the BS is classified as uplink traffic. A scheduling algorithm implemented at the BS has to deal with both uplink and downlink traffic. In some cases, separate scheduling algorithms are implemented for the uplink and downlink traffic. Typically, a Call Admission Control (CAC) procedure is also implemented at the BS that ensures the load supplied by the SSs can be handled by the network. A CAC algorithm will admit a SS into the network if it can ensure that the minimum Quality of Service (QoS) requirements of the SS can be satisfied and the QoS

of existing Ss will not deteriorate. The performance of the scheduling algorithm for the uplink traffic strongly depends on the CAC algorithm.

Packet scheduling is the process of resolving contention for shared resources in a network. The process involves allocating bandwidth among the users and determining their transmission order. Scheduling algorithms for a particular network need to be selected based on the type of users in the network and their QoS requirements. QoS requirements vary depending on the type of application/user. For real-time applications such as video conferencing, voice chat and audio/video streaming, delay and delay jitter are the most important QoS requirements. Delay jitter is the inter-packet arrival time at the receiver and is required to be reasonably stable by the real-time applications. On the other hand, for non-real time applications such as file transfer (FTP), throughput is the most important QoS requirement. Some applications, such as web-browsing and email do not have any QoS requirements. In a network, different types of applications, with diverse QoS requirements, can co-exist. A scheduling algorithm's task in a multi-class network is to categorize the users into one of the pre-defined classes. Each user is assigned a priority, taking into account its QoS requirements. Subsequently, bandwidth is allocated according to the priority of the users as well as ensuring that fairness between the users is maintained.

Besides having a very close coupling with the QoS requirements of the users, the design of a scheduling algorithm also depends on the type of network it is intended for. Packet scheduling algorithms can usually be distinguished based on their characteristics. Some of the desirable qualities a scheduling algorithm must possess and the issues that need to be addressed by the algorithms include [1],[2]:

- **Flexibility:** A scheduling algorithm should be able to accommodate users with diverse QoS requirements and also meet the minimum requirements of users. Ideally, the design of a scheduling algorithm should be flexible enough so that it requires minimal changes to be deployed in a different network or even a different technology.
- **Simplicity:** A scheduling algorithm should be simple, both conceptually and mechanically. Conceptual simplicity allows manageable analysis of the algorithm such that distribution or worst case analysis bound for parameters such as delay and throughput can be derived. Mechanical simplicity allows efficient implementation of the algorithm on a large scale.
- **Protection:** A scheduling algorithm needs to be able to protect well-behaving users from sources of variability such as best effort traffic, misbehaving users and fluctuations in the network load. Upon admission into the network, users enter into a service level agreement (SLA) that they will adhere to (e.g., a user will specify peak and mean traffic rates). Sometimes a user will not abide by the SLA, causing unpredicted traffic fluctuations in the network. A scheduling algorithm needs to ensure that such fluctuations do not affect well-behaving users in the network.
- **Fairness:** Besides satisfying the QoS requirements of users, a scheduling algorithm needs to ensure that a reasonable level of fairness is maintained among the users. Fairness measures the difference between users with respect to the resources allocated to them. In a wireless network, due to the presence of variations in channel quality, users experiencing poor channel quality might be denied service by the scheduling algorithm. This is because bandwidth allocated to users with inferior channel quality will essentially be wasted as the data will be lost or corrupted prior to reaching the

destination. A scheduling algorithm needs to have a mechanism to compensate users that have lost service and maintain fairness among all users.

- **Link Utilization:** A scheduling algorithm is required to assign bandwidth to the users such that maximum link utilization is realized. Link utilization is a critical property for the service providers as it is directly linked to the revenue generated. A scheduling algorithm needs to ensure that resources are not allocated to users that do not have enough data to transmit, thus resulting in wastage of resources.
- **Power conservation on the mobile device:** Due to limited power available on the mobile device, a scheduling algorithm needs to ensure that limited processing is done on the device.
- **Device mobility:** Different cells can have a different notion of time, i.e., the BSs of different cells are not required to be synchronized. When a mobile device moves from one cell to another, packets need be time-stamped based on the notion of time in the new cell. Scheduling algorithms that allocate bandwidth to the users according to the time-stamp of the packets (e.g. schedule users based on their packet deadlines) will not function as expected if the packets are not stamped with the correct notion of time.

1.1 Thesis Contribution

In this thesis, we focus on evaluating scheduling algorithms for the uplink traffic in WiMAX. We evaluate a number of WiMAX uplink scheduling algorithms in a single-hop network, which is referred to as Point to Multipoint (PMP) mode of WiMAX. The main contributions of our work are:

- Survey and categorize scheduling algorithms for the uplink traffic in WiMAX. Identify the strengths and weaknesses of the algorithms based on which the algorithms will be selected for evaluation.
- Identify performance metrics that will effectively evaluate the scheduling algorithms. The scheduling algorithms will be evaluated with respect to various characteristics of WiMAX as specified in the IEEE 802.16 standard.
- Update the Network Simulator 2 (NS2) extension of WiMAX by incorporating the implementation of representative scheduling algorithms. A traffic model based on the WiMAX specifications is also implemented.
- Based on the simulation results, highlight the effects of the characteristics of WiMAX on the performance of the scheduling algorithms. We also identify open issues and provide suggestions to improve the performance of the evaluated algorithms.

1.2 Thesis Organization

The rest of the thesis is organized as follows. In chapter 2, we provide a description of the IEEE 802.16 standards with a focus on the IEEE 802.16-2004[3] standard. The IEEE 802.16-2004 standard provides specification for the PMP mode of operation of WiMAX. We also categorize and discuss scheduling algorithms for uplink traffic in WiMAX. In chapter 3, we provide detailed information about the scheduling algorithms selected for evaluation. Such information includes the pseudo-code of the algorithms and any assumptions made. In chapter 4, we describe the simulation environment, including the traffic model and values of the MAC and PHY layer parameters, and define the performance metrics to be used in evaluating the algorithms. We then discuss the results

of the experiments. In chapter 5, we highlight the shortcomings of the algorithms and propose enhancements. We also discuss some of the open issues and suggest future research directions in the area.

Chapter 2. Background

In this chapter we will discuss the characteristics of the MAC and PHY layers as specified in the IEEE 802.16 standard. More specifically, we will describe the various scheduling services, bandwidth request mechanisms and the transmission modes in the IEEE 802.16 standard. We will then discuss packet scheduling algorithms for the uplink traffic in WiMAX Point to Multipoint (PMP) mode. We categorize the algorithms into three classes and discuss the strengths and weaknesses of each algorithm.

2.1 Worldwide Interoperability of Microwave Access (WiMAX)

Prior to the introduction of the IEEE 802.16 standard, the most effective ways to obtain access to broadband internet service were mainly through T1, Digital Subscriber Line (DSL), or cable modem based connections. However, these wired infrastructures are considerably more expensive, especially for deployment in rural areas and developing countries. This limitation propelled the industry to devise an alternative means of obtaining broadband internet access and the approach taken was via the wireless medium. The IEEE 802.16 standard provides specification for the MAC and PHY layers for the air interface. The standard includes details about the various flavors of PHY layers supported and characteristics of the MAC layer such as bandwidth request mechanisms and the scheduling services supported.

The WiMAX forum, a consortium of about 420 members including major corporations like AT&T, Fujitsu, Intel and Siemens, was set up in June 2001 to support

the WiMAX technology and promote its commercial use. The forum is responsible for preparing profiles for systems that comply with the IEEE 802.16 standard and create interoperability tests to ensure different vendors' implementation can work together. The first version of the IEEE 802.16 standard was completed in October 2001 and since then several version have emerged addressing issues such as Non-Line Of Sight (NLOS) operation, mobility, multiple traffic classes for QoS, operation in the licensed and unlicensed frequency bands.

2.1.1 The Evolution of IEEE 802.16 standard

The IEEE 802.16 Working Group is the body responsible for developing the IEEE 802.16 standard and its extensions. In this section, we provide an overview of some of functionalities specified in the IEEE 802.16 standard and its extensions. The following is a chronological ordering of the standards and their respective content:

IEEE 802.16a [4]: This was the first extension of the IEEE 802.16 standard that allowed the use of licensed and license-free frequencies from 2GHz to 11GHz. Most of the industrial interest is in the unlicensed frequencies. The lower frequencies allow signals to penetrate obstacles and thus do not require a Line-Of-Sight (LOS) between the Subscriber Station (SS) and the Base Station (BS). This extension also allows mesh deployment, whereby the subscribers can act as relay points by transferring information from the BS to other SSs not in direct path of the BS.

IEEE 802.16c [5]: This extension contains specifications to allow the technology to inter-operate in the licensed frequency band of 10GHz to 66GHz. The extension clearly identifies the mandatory and optional features of the technology so that implementation and interoperability are clearer. The extension also addresses issues such as performance

evaluation, testing and detailed system profiling along with adding support for Multiple Input Multiple Output (MIMO) antennas.

IEEE 802.16-2004 [1]: This extension of the standard, popularly known as “Fixed WiMAX”, is the combination of 802.16a and 802.16c extensions with some modifications. The standard supports both Time Division Duplex (TDD) and Frequency Division Duplex (FDD) services. The product profile, as specified by the standard, utilizes the OFDM 256-FFT (Fast Fourier Transform) system profile. One of the enhancements in this extension is the concatenation of Protocol Data Unit (PDU) and Service Data Unit (SDU) which reduces the MAC overhead. This extension provides a substantial improvement for the polling mechanism. It allows the SS to be polled individually or in groups. It also allows piggybacking bandwidth requests over data packets thus reducing collisions and system overhead.

IEEE 802.16e-2005 [6]: This extension, known as “Mobile WiMAX”, adds mobility support for the technology. The extension preserves the technical aspects of “Fixed WiMAX” while adding support for mobile broadband wireless access. The standard specifies the use of OFDMA technology with support for 2000-FFT, 1000-FFT, 512-FFT and 128-FFT system profiles. The OFDMA technology allows signals to be divided into many sub-channels to increase resistance to multi-path interference. The specification in the standard supports mobile device speeds up to 100 km/h.

IEEE 802.16f [7]: The extension is currently an active amendment with the intention to support Management Information Base (MIB). A MIB is a database of information used for managing all the devices in the network. The extension aims to provide a detailed

description of the protocol for managing information between the Subscriber Stations (SSs) and the Base Station (BS).

IEEE 802.16g [8]: This extension is currently under development and it aims to improve the co-existence mechanisms for license exempt operations. More specifically, the extension is attempting to find approaches to allow coexistence between fixed wireless access networks operating in the license exempt bands, primarily the 5GHz frequency band.

Our work is based on the 802.16-2004 standard [1], popularly known as “Fixed WiMAX”, that provides bandwidth up to 75Mbps, without mobility. This extension of the IEEE 802.16 standard is geared to providing broadband internet access to residential and commercial buildings where mobility is not a requirement. In the next few sections, we highlight some of the main features of the PHY and MAC layer as specified in the 802.16-2004 standard.

2.1.2 IEEE 802.16 PHY Layer

The purpose of the PHY layer is the physical transport of data. In the IEEE 802.16-2004 standard, the PHY layer is defined for frequencies ranging from 2 to 66 GHz with the sub-range 10-66 GHz requiring Line Of Sight (LOS) propagation and possibility of Non Line Of Sight (NLOS) propagation in the 2-11 GHz frequency range.

The principle technologies behind the PHY layer of WiMAX are Orthogonal Frequency Division Multiplexing (OFDM) and Orthogonal Frequency Division Multiple Access (OFDMA). OFDM is a multi-carrier transmission technique that has recently gained popularity for high-speed bidirectional wireless data communication [9]. OFDM

basically squeezes multiple modulated carriers together reducing the required bandwidth and at the same time keeping the modulated signals orthogonal to each other so they do not interfere with each other. OFDM is based on a technology called Frequency Division Multiplexing (FDM) that uses many frequencies to transmit signals in parallel. OFDM is more efficient than FDM as it allows sub-channels to be spaced much close to each other by finding orthogonal frequencies. On the other hand, OFDMA allows certain sub-carriers to be assigned to different users. A group of sub-carriers constitutes a sub-channel with each sub-channel belonging to a particular SS.

Both Time Division Duplex (TDD) and Frequency Division Duplex (FDD) are specified in the IEEE 802.16-2004 standard. TDD is a technique in which the system receives and transmits within the same frequency channel, assigning time slices for transmit and receive modes. In FDD, two separate frequencies are required to transmit and receive, usually separated by 50 to 100 MHz within the operating band. The downlink and uplink frame structures in FDD are similar except they are transmitted in separate channels. When half duplex FDD (H-FDD) is used at the SSs, the BS must make sure that it does not schedule the SSs to transmit and receive at the same time.

Adaptive Antenna System (AAS) is used in WiMAX to specify the beam-forming techniques whereby a set of antennas is used at the BS to increase the gain to the SSs, at the same time reducing interference to and from other SSs. AAS can be used to enable Spatial Division Multiple Access (SDMA) so that multiple SSs that are in different space can receive and transmit on the same sub-channel simultaneously.

The IEEE 802.16-2004 standard [1] specifies 5 variants of the PHY layer distinguished by whether the PHY layer is Single Carrier (SC) or uses OFDM technology. The variants with a brief description follow:

Wireless Metropolitan Area Network – Orthogonal Frequency Division Multiplexing (WirelessMAN-OFDM): The WirelessMAN-OFDM PHY is based on OFDM technology designed mainly for fixed SSs, where the SSs are deployed in residential areas and businesses. The OFDM PHY supports sub-channelization in the uplink with 16 sub-channels. It also supports TDD and FDD frame structures with both FDD and H-FDD options. The modulation schemes supported are BPSK, QPSK, 16-QAM and 64-QAM.

Wireless Metropolitan Area Network – Orthogonal Frequency Division Multiple Access (WirelessMAN-OFDMA): This variant is based on OFDMA technology and offers sub-channelization in both uplink and downlink. The OFDMA PHY supports both TDD and FDD frame structures, with both FDD and H-FDD options. The variant is different from WirelessMAN-OFDM in that it supports sub-channelization in both the uplink and downlink directions resulting in broadcast messages being transmitted at the same time as data.

Wireless High Speed Unlicensed Metro Area Network (WirelessHUMAN): This specification of the PHY layer is similar to the OFDM based layer except it is focused on Unlicensed National Information Infrastructure (UNII) devices and other unlicensed bands.

Wireless Metropolitan Area Network – Single Carrier (WirelessMAN-SC): This variant specifies the use of the technology in the frequency range 10-66GHz. The PHY

Layer design supports Point to Multi-Point (PMP) architecture whereby the BS acts as the coordinator for all the SSSs in its cell. In this design, the BS transmits a Time Division Multiplexing (TDM) signal in which the SSSs are allocated time slots serially. This variant provides support for both TDD and FDD frame structures. Both TDD and FDD support adaptive burst profiles whereby the modulation and coding options can be dynamically assigned on a burst by burst basis.

Wireless Metropolitan Area Network - Single Carrier Access (WirelessMAN-SCa):

This variant of the PHY layer uses single carrier modulation in the 2-11GHz frequency range and it is intended for Non Line-Of-Sight (NLOS) operations. It supports both FDD and TDD frame structures with TDMA in the uplink and TDM or TDMA in the downlink. The PHY specification includes Forward Error Correction (FEC) coding for both uplink and downlink and framing structures that allow improved channel estimation performance over NLOS operations.

2.1.3 IEEE 802.16 MAC Layer

The MAC layer in WiMAX basically provides intelligence to the PHY layer. It contains 2 sub layers: service-specific convergence sub-layer and the MAC common part sub-layer. The service-specific convergence sub-layer is responsible for interfacing with upper layers while the MAC common part sub layer carries out the key MAC functions [10].

Service specific convergence sub-layer: The IEEE 802.16-2004 standard specifies 2 types of service specific convergence sub-layers for mapping service to and from the MAC layer; the ATM sub-layer for mapping ATM services and the packet sub-layer for

mapping packet services such as IPv4, IPv6 and Ethernet. The main task of the sub-layer is to map Service Data Units (SDUs) to MAC connections, and to enable QoS and bandwidth allocation based on the parameters received from the upper layers. The convergence sub-layer also has the ability to perform more complicated tasks such as payload header compression.

MAC common part sub-layer: The MAC protocol, according to the IEEE 802.16-2004 standard, is mainly designed for Point to Multi-Point (PMP) operation. The MAC layer is connection-oriented including connection-less services mapped to a connection which allows a way of requesting bandwidth and mapping the QoS parameters of the connection. Connections contain a 16-bit Connection Identifiers (CIDs) which act as the primary addresses used for all operations. Each SS has a 48-bit MAC address which is mainly used as an equipment identifier. Upon entering the network, an SS is assigned 3 management connections in each of the uplink and downlink directions. These connections are:

Basic Connection: This connection is responsible for transferring critical MAC and Radio Link Control (RLC) messages.

Primary Management Connection: This connection is responsible for transferring longer and more delay-tolerant control messages such as those used for authentication and connection setup.

Secondary Management Connection: This connection is used for transfer of standard based management messages such as Dynamic Host Configuration Protocol (DHCP),

Trivial File Transfer Protocol (TFTP) and Simple Network Management Protocol (SNMP).

The functionalities supported by the Common Part Sub-layer are:

Channel Acquisition: The MAC protocol includes an initialization procedure that allows an SS, upon installation, to scan its frequency list to find an operating channel. Once the SS is synchronized, it will periodically search for Downlink Channel Descriptor (DCD) and Uplink Channel Descriptor (UCD) messages that inform the SS about the modulation scheme used on the channel.

Ranging and Negotiation: After determining the parameters to use for ranging, the SS will search for ranging opportunities by scanning the UL MAP messages in every frame. The ranging response from the SS allows the BS to find out the distance between itself and the SS.

SS authentication and Registration: Each SS contains a manufacturer issued digital certificate that establishes a link between the 48-bit MAC address of the SS and its public RSA key. After verifying the identity of the SS, if the SS is admitted into the network, the BS will respond to the request with an Authorization Reply containing an Authorization Key (AK), encrypted with the SS's public key.

Bandwidth Grant and Request: The MAC layer distinguishes between 2 classes of SS, one that accepts bandwidth for a connection and the other one that accepts bandwidth for the SS. For the Grant per Connection (GPC) class, bandwidth is granted explicitly to the connection of the SS. With the Grant per SS (GPSS) class, bandwidth is granted to the SS and then it's up to the SS on how to distribute the bandwidth among its connections. The SS can use the bandwidth for the connection that requested it or for another connection.

The IEEE 802.16-2004 standard allows SSs to request bandwidth via contention or piggyback mechanisms. In contention mechanism, the SSs will contend for a slot to send the bandwidth request while in the piggyback mechanism the SSs will attach their bandwidth request onto data packets. A bandwidth request can also be either incremental or aggregate. When the BS receives an incremental bandwidth request, it will add the quantity of the bandwidth requested to its current perception of the bandwidth needs of the SS. When the BS receives an aggregate bandwidth request, it will replace its perception of the bandwidth needs of the SS with the quantity of bandwidth requested. Piggyback bandwidth requests are always incremental.

Frame structure and MAP messages: IEEE 802.16 MAC supports both TDD and FDD frame structures. The MAC starts building the downlink sub-frame with DL-MAP (Downlink MAP) and UL-MAP (Uplink MAP) messages. The DL-MAP indicates the PHY transitions on the downlink while the UL-MAP indicates bandwidth allocation and burst profiles in the uplink. In a TDD frame structure, the frame is divided into uplink and downlink sub-frames along the time axis (see Figure 2-1). The frame starts with the downlink sub-frame followed by a short gap called transmit/receive transition gap (TTG). The downlink sub-frame contains a preamble followed by a header and one or more downlink bursts. Following TTG, the uplink sub-frame contains one or more uplink bursts. Each uplink burst contains a preamble that allows the BS to synchronize with each SS. The uplink sub-frame is then followed by a short gap called the receive/transmit transition gap (RTG) before the BS can start transmitting again.

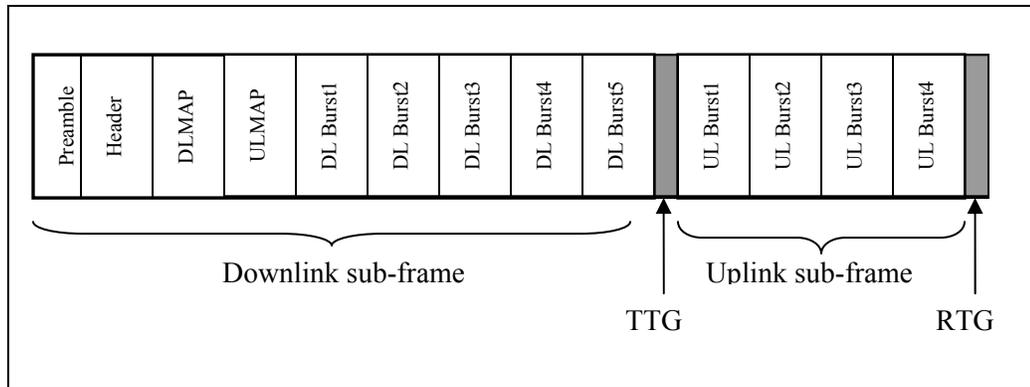


Figure 2-1: TDD Frame Structure

Packing and Fragmentation of MAC SDUs: This functionality is executed in tandem with the bandwidth allocation process to maximize efficiency, flexibility and effectiveness. Fragmentation is the process by which a MAC SDU is divided into one or more SDU fragments. Packing is the process in which multiple MAC SDUs are packed into a single MAC PDU payload. Either functionality can be initiated by the BS in the downlink or by the SS in the uplink.

Scheduling Services: The Common Part Sub-layer maps each connection to a scheduling service, where a scheduling service is associated with pre-determined QoS parameters. The IEEE 802.16-2004 standard specifies four scheduling services: Unsolicited Grant Service (UGS), real-time Polling Service (rtPS), non-real time Polling Service (nrtPS) and Best Effort (BE). The bandwidth allocation mechanism for the UGS service as specified in the IEEE 802.16-2004 standard [1] requires the BS to send fixed size grants to the SSs periodically.

2.1.4 IEEE 802.16 Service Classes

The IEEE 802.16-2004 standard [1] specifies the provision of four scheduling services:

- **Unsolicited Grant Service (UGS):** This scheduling service is designed to support applications that generate fixed-size data packets periodically such as T1/E1 and VoIP without silence suppression. To support the real-time needs of such applications and reduce overhead by the bandwidth request-grant process, the BS allocates fixed size data grants without receiving explicit requests from the SS. The size of the grants is based on the maximum rate that can be sustained by the application and is negotiated at connection setup.
- **real-time Polling Service (rtPS):** This scheduling service is designed to support real-time applications that generate variable size packets on a periodic basis such as MPEG video or VoIP with silence suppression. The BS allows the SSs to make periodic unicast requests and allows them to specify the size of the desired grant. Since a dedicated grant request is contention-free, the bandwidth request is guaranteed to be received by the SS in time. SSs belonging to this class are prohibited from using contention request opportunities.
- **non real-time Polling Service (nrtPS):** nrtPS is designed to support non-real time applications that require variable size data grant bursts on a regular basis. This scheduling service supports applications that are delay tolerant but may need high throughput such as File Transfer Protocol (FTP) applications. The BS allows the SS to make periodic unicast grant requests, just like the rtPS scheduling service, but the requests are issued at longer intervals. This will ensure that the SSs receive request

opportunities even during network congestion. SSSs of this class are also allowed to use contention request opportunities.

- **Best Effort (BE):** This traffic class contains applications such as telnet or World Wide Web (WWW) access that do not require any QoS guarantee. The bandwidth request by such applications is granted on space-available basis. The SS is allowed to use both contention-free and contention based bandwidth requests, although contention-free is not granted when the system load is high.

In Table 2-1, we list some of the applications belonging to each class, including suggested delay guarantees that they require under both premium and basic service [11].

Table 2-1: Traffic classes in 802.16 and their QoS requirements

Class of Service	QoS Requirements			Kinds of services
	QoS Factors	Premium	Basic	
UGS	Packet Delay	< 150ms	< 250ms	VoIP
	Delay Jitter	< 30ms	< 50ms	
	Packet Loss	< 0.3%	< 0.5%	
	Guarantee	> 99.9%	> 99.5%	
rtPS	Packet Delay	< 300ms	< 600ms	Video Telephony, Video game, VOD, AOD, Internet shopping, bank and stock transaction
	Delay Jitter	< 50ms	< 100ms	
	Packet Loss	< 1%	< 5%	
	Guarantee	> 99%	> 95%	
nrtPS	Packet Delay	N/A	N/A	High-speed file transfer, Multi- media messaging, E-commerce
	Delay Jitter	N/A	N/A	
	Packet Loss	0-2%	0-5%	
	Guarantee	> 98%	> 82%	
BE	Packet Delay/ Delay Jitter/ Packet Loss/ Guarantee	N/A	N/A	Web-browsing, SMS

2.2 Packet scheduling for uplink traffic in WiMAX

Packet scheduling algorithms are implemented at both the BS and SSs. A scheduling algorithm at the SS is required to distribute the bandwidth allocation from the BS among its connections. A scheduling algorithm at the SS is not needed if the BS grants bandwidth to each connection of the SS separately i.e. the Grant Per Connection (GPC) procedure is followed. If the Grant Per Subscriber Station (GPSS) procedure is followed, the scheduling algorithm at the SS needs to decide on the allocation of bandwidth among its connections. The scheduling algorithm implemented at the SS can be different than that at the BS.

The focus of our work is on scheduling algorithms executed at the BS for the uplink traffic in WiMAX i.e. traffic from the SSs to the BS. A scheduling algorithm for the uplink traffic is faced with challenges not faced by an algorithm for the downlink traffic. An uplink scheduling algorithm does not have all the information about the SSs such as the queue size. An uplink algorithm at the BS has to coordinate its decision with all the SSs where as a downlink algorithm is only concerned in communicating the decision locally to the BS.

Based on our comprehensive survey, we have classified scheduling algorithms for the uplink traffic in WiMAX into three categories (see Figure 2-2):

- **Homogeneous scheduling algorithms:** These are legacy scheduling algorithms that attempt to address issues such as providing QoS, flow isolation and fairness. The algorithms were originally proposed for wired networks but are used in WiMAX principally to satisfy QoS requirements of the four traffic scheduling services. Algorithms in this category do not address the issue of link channel quality.

- **Hybrid scheduling algorithms:** This category contains algorithms that use a combination of legacy scheduling algorithms in an attempt to satisfy QoS requirements of the four scheduling services. Some of the algorithms in this category also address the issue of variable channel conditions in WiMAX. An important aspect of algorithms in this category is the overall allocation of bandwidth among the scheduling services. Once bandwidth has been assigned to each class, a legacy algorithm is executed for SSs of the class to determine the bandwidth allocation in that class.
- **Opportunistic scheduling algorithms:** Scheduling algorithms in this category are primarily focused on exploiting the variability in channel conditions in WiMAX. The algorithms also attempt to satisfy QoS requirements of the four scheduling services and maintain fairness between the SSs.

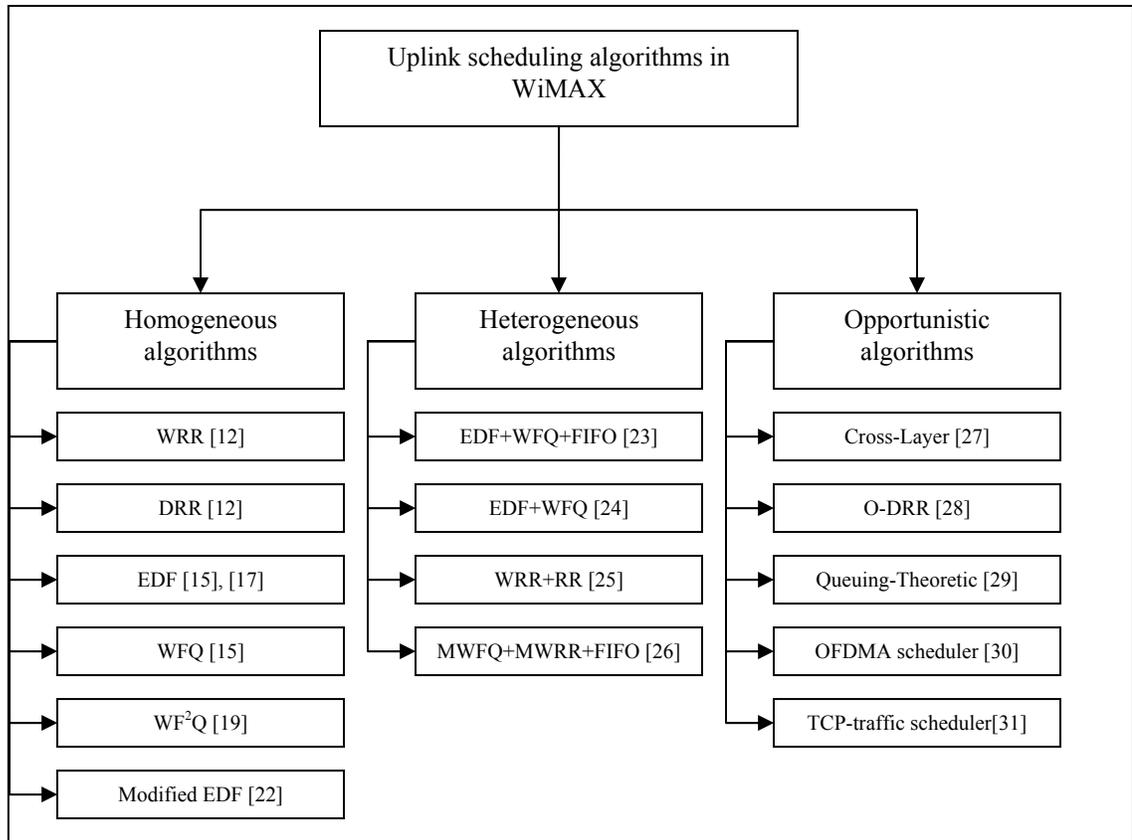


Figure 2-2: Taxonomy of uplink scheduling algorithms in WiMAX

2.2.1 Homogeneous scheduling algorithms

Most of the legacy scheduling algorithms can be classified into this category. We will only discuss those that have been proposed and evaluated in WiMAX. Weighted Round Robin (WRR) and Deficit Round Robin (DRR) algorithms are evaluated on a WiMAX system in [12]. WRR is evaluated for the uplink traffic while DRR is evaluated for the downlink traffic. WRR is an extension of the Round Robin (RR) algorithm. It is a work-conserving algorithm in that it will continue allocating bandwidth to the SSs as long as they have backlogged packets. The WRR algorithm assigns weight to each SS and the bandwidth is then allocated according to the weights. The algorithm was originally proposed for ATM networks that have fixed size packets [13]. Since the

bandwidth is assigned according to the weights only, the algorithm will not provide good performance in the presence of variable size packets. It has been discussed in [12] that weight to a SS can be assigned to reflect its relative priority i.e. a higher weight assigned to SSs of the rtPS class compared to the weight assigned to SSs of the nrtPS and BE classes. DRR, a variation of RR, is also a work-conserving algorithm with a constant processing time [14]. DRR is similar to the RR algorithm in that a quantum of service is assigned to each SS. The difference between the two algorithms is that when a SS is not able to send a packet, the remainder quantum is stored in a deficit counter. The value of the deficit counter is added to the quantum in the following round. The algorithm is flexible enough as it allows provision of quanta of different sizes depending on the QoS requirements of the SSs. DRR is mostly suited for datagram networks where packet sizes vary. Since the DRR algorithm requires accurate knowledge of packet size, it is not suitable for the uplink traffic.

N. Ruangchaijatupon *et al.* [15] evaluate the performance of the Earliest Deadline First (EDF) algorithm. EDF is a work conserving algorithm originally proposed for real-time applications in wide area networks [16]. The algorithm assigns deadline to each packet and allocates bandwidth to the SS that has the packet with the earliest deadline. Deadlines can be assigned to packets of a SS based on the SS's maximum delay requirement. The EDF algorithm is suitable for SSs belonging to the UGS and rtPS scheduling services, since SSs in this class have stringent delay requirements. Since SSs belonging to the nrtPS service do not have a delay requirement, the EDF algorithm will schedule packets from these SSs only if there are no packets from SSs of UGS or rtPS

class. With a large number of SSs from the UGS or rtPS class, SSs from the nrtPS or BE class can potentially starve.

T. Tsai *et al.* [17] propose an uplink scheduling algorithm and a token bucket based Call Admission Control (CAC) algorithm. The CAC algorithm assigns thresholds to each class to avoid starvation of lower priority classes. The EDF scheduling algorithm is used for the rtPS class. The scheduling algorithm first grants bandwidth to SSs of the UGS class. The algorithm will then allocate bandwidth to SSs of the rtPS class using the EDF algorithm and restricting the allocation to the maximum grant size. Finally, the algorithm will allocate minimum required bandwidth to SSs of the nrtPS and BE classes in order. Each SS is controlled by a token rate and bucket size. A mathematical model is proposed that estimates the appropriate token rate based on the queuing delay and the packet loss requirements, for both finite and infinite queue length.

N. Ruangchajaturon *et al.* [15] also evaluate Weighted Fair Queuing (WFQ) for the uplink traffic in WiMAX. WFQ is a packet-based approximation of the Generalized Processor Sharing (GPS) algorithm [18]. GPS is an idealized algorithm that assumes a packet can be divided into bits and each bit can be scheduled separately. This is an impractical assumption since a packet needs to be scheduled in its entirety. WFQ is a practical implementation of GPS as it assigns finish times to packets and selects packets in increasing order of their finish times. The finish times of packets of a SS are calculated based on the weight assigned to the SS and the size of the packets. The WFQ algorithm results in superior performance compared to the WRR algorithm in the presence of variable size packets. The finish time of a packet is essentially the time the packet would have finished service under the GPS algorithm. The disadvantage of the WFQ algorithm

is that it will service packets even if they wouldn't have started service under the GPS algorithm. This is because the WFQ algorithm does not consider the start time of a packet.

Y. Shang and S. Cheng in [19] propose a hierarchical model for packet scheduling based on the proposal of J. Bennet and H. Zhang in [20]. The model is comprised of three scheduling servers: a hard-QoS scheduling server, a soft-QoS scheduling server and a best effort scheduling server. The UGS traffic is mapped to the hard-QoS server and the rtPS traffic is mapped to the soft-QoS server. The nrtPS traffic can be mapped to either the soft-QoS server or the best effort server. All the servers implement the Worst-case Fair Weighted Fair Queuing (WF^2Q) scheduling algorithm [21]. WF^2Q is another extension of GPS that addresses the issue of the WFQ algorithm servicing packets even if they wouldn't have started service under the GPS scheme. O. Yang and J. Lu in [22] propose a joint CAC and scheduling algorithm based on the concept of EDF to satisfy the QoS requirements of real-time video applications in IEEE 802.16 networks. The algorithm claims to cover throughput requirement, delay constraint and maintain fairness among the SSs simultaneously.

2.2.2 Hybrid scheduling algorithms

The hybrid class of scheduling algorithms uses a combination of legacy scheduling algorithms to satisfy QoS requirements of the multi-class traffic as specified in the IEEE 802.16-2004 standard [1]. The most critical part of hybrid algorithms is the distribution of uplink bandwidth among different traffic classes.

K. Wongthavarawat and A. Ganz propose a hybrid scheduling algorithm in [23] that combines EDF, WFQ and FIFO scheduling algorithms. The overall allocation of

bandwidth is done in a strict priority manner i.e. all the higher priority SSs are allocated bandwidth until they do not have any packets to send. The EDF scheduling algorithm is used for SSs of the rtPS class, WFQ is used for SSs of the nrtPS class and FIFO for SSs of the BE class. Besides the scheduling algorithm, an admission control procedure and a traffic policing mechanism are also proposed. All these components together constitute the proposed QoS architecture. A drawback of this algorithm is that lower priority SSs will essentially starve in the presence of a large number of higher priority SSs due to the strict priority overall bandwidth allocation.

K. Vinay *et al.* [24] propose a hybrid scheme that uses EDF for SSs of the rtPS class and WFQ for SSs of the nrtPS and BE classes. This algorithm differs from the one in [23] in a couple of ways. First, the WFQ algorithm is used for SSs of both nrtPS and BE classes. Secondly, the overall bandwidth allocation is not done in a strict priority manner. Although the details of overall bandwidth allocation are not specified, it is briefly mentioned that the bandwidth is allocated among the classes in a fair manner. Since SSs of the BE class do not have any QoS requirements, using a computationally complex algorithm such as WFQ for them is not needed.

M. Settembre *et al.* [25] propose a hybrid scheduling algorithm that uses WRR and RR algorithms with a strict priority mechanism for overall bandwidth allocation. In the initial portion of the algorithm, bandwidth is allocated on a strict priority basis to SSs of the rtPS and nrtPS classes only. After that the WRR algorithm is used to allocate bandwidth among SSs of rtPS and nrtPS classes until they are satisfied. If any bandwidth remains, it is distributed among the SSs of the BE class using the RR algorithm. The hybrid algorithm is opportunistic in nature as the WRR and RR algorithms select the SSs

with the most robust burst profiles. This algorithm will starve lower priority SSs in the presence of a large number of higher priority SSs. The algorithm can also result in low fairness among SSs as it selects SSs with the most robust burst profiles first.

J. Lin and H. Sirisena [26] propose an architecture called Multi-class Uplink Fair Scheduling Structure (MUFSS) to satisfy throughput and delay requirements of the multi-class traffic in WiMAX. The proposed scheduling discipline at the BS is Modified Weighted Round Robin (MWRR), although details of the modifications to the WRR discipline are not provided by the authors. The model is based on Grant Per Subscriber Station (GPSS) bandwidth grant mode and thus schedulers are implemented at the SSs to distribute the bandwidth granted among their connections. At the SS, Modified WFQ (MWFQ) is used for UGS and rtPS connections, MWRR is used for nrtPS connections and FIFO is used for BE connections.

Normally, implementing the scheduling algorithms to cater to all the service classes and on a per connection basis is not a trivial task. The IEEE 802.16 standard supports changing a connection's QoS parameters during its lifetime. Thus, the scheduling algorithms are required to adapt to these changes by reassigning the slots. For instance, in the case of the WFQ algorithm, dynamic weights may be the solution, which considerably increases the algorithm's complexity and renders its implementation challenging. The standard also requires the frame size to be constant. This would result in some of the time slots not being utilized, which makes the performance of the algorithms closer to that of non work-conserving schedulers, whereas the proposed homogeneous and hybrid algorithms are work-conserving. Therefore, most of the proposals make significant assumptions that simplify the implementation of the algorithms. In [12], the

authors implement DRR as the downlink scheduler, without considering the WiMAX QoS parameters, and WRR as the uplink scheduler with constant weights, without clarifying how the weights are chosen, or providing justification for adopting constant weights.

2.2.3 Opportunistic scheduling algorithms

Opportunistic scheduling algorithms proposed for WiMAX exploit variations in channel quality giving priority to users with better channel quality, while attempting to satisfy the QoS requirements of the multi-class traffic.

A cross-layer scheduling algorithm is proposed in [27] whereby each SS is assigned a priority based on its channel quality and service status. The SS with the highest priority is scheduled each time. SSs of the UGS class are assigned a fixed number of time slots based on their required rate. The priority function for the rtPS class takes the deadline and waiting time of the HOL packet as input while the priority function for the nrtPS class takes the average throughput and the minimum required throughput as inputs. The priority function for the BE class only takes the channel quality of the SS as input. A coefficient is included in the priority functions to reflect the relative priority of the classes, i.e., a higher coefficient is used for the rtPS class compared to the coefficients of nrtPS and BE classes. One of the drawbacks of this algorithm is that it results in poor channel utilization since it schedules only one SS every frame. This will also result in low average throughput and higher average delay as SSs will have to wait for longer period of time before they are scheduled.

H. Rath et al. [28] propose to use an opportunistic extension of the Deficit Round Robin (DRR) algorithm with the purpose of satisfying delay requirements of multi-class

traffic in WiMAX. The proposed algorithm uses a deficit counter and a quantum size just like the DRR algorithm. A critical part of the algorithm is the polling interval. At the beginning of a polling interval, a set of schedulable SSs are selected that forms a schedulable set. Until the next polling interval, SSs are selected only from the schedulable set. SSs that are not selected will have their quantum accumulated in the deficit counter. A major limitation of this algorithm is selecting an appropriate polling interval. If a large polling interval is selected, it will result in some of the SSs being denied service for a long period of time. Even if a SS becomes eligible to be added to the schedulable set, it will not be considered until the next polling interval.

D. Niyato and E. Hossain [29] propose a joint bandwidth allocation and connection admission control algorithm based on queuing theory. In order to limit the amount of bandwidth allocated per class, a bandwidth threshold is assigned to each class. Each SS is assigned a utility and bandwidth is allocated to the SS with the least utility. Utility for SSs of the rtPS and nrtPS classes is calculated using a sigmoid function that takes the average delay and maximum latency for the rtPS class and average throughput and minimum required throughput for the nrtPS class as inputs. SSs of the BE class are assigned the highest utility as long as 1 unit of bandwidth is assigned to them. The unit of bandwidth used in the algorithm is a fixed size packet called a Protocol Data Unit (PDU). In the initial stages of the algorithm, minimum bandwidth is allocated to the SSs in a strict priority order. Afterwards, the algorithm assigns the residual bandwidth in PDU units to the SSs. SSs are sorted in ascending order based on their utility and they are allocated the PDUs based on a water filling mechanism until the bandwidth of each class exceeds the class threshold. The thresholds for the classes are calculated such that QoS

requirements of the SSs are met and the system revenue is maximized. If a packet is too large and the minimum bandwidth allocated to the SS is not enough to transmit the large packet, the allocation will essentially be wasted. Another drawback of the algorithm is that it will result in maximum MAC and PHY layer overhead since it selects all the SSs in the network based on the first phase of the algorithm as aforementioned.

V. Singh and V. Sharma [30] propose scheduling algorithms for the OFDMA system with a TDD frame structure for both uplink and downlink traffic in WiMAX. An optimization problem is first formulated for the allocation of resources to the SSs. The formulation considers the channel quality in terms of bytes/slot a SS can send on a channel, the number of slots allotted to the SS and the total bandwidth demanded by the SS. A linear programming solution to the optimization problem is devised that will maximize the throughput of the SSs. Due to the high complexity of the linear programming solution, a heuristic algorithm is considered that provides close to optimal performance. The heuristic algorithm will maximize the throughput of the SSs but is not fair which leads to the design of fair versions of the algorithm. A definition of fairness is also proposed that requires assigning bandwidth to a SS proportional to its throughput requirement. The bandwidth allocation among the scheduling services is done on a priority basis, similar to that of the hybrid algorithm proposed in [23]. More specifically, the BS first attempts to satisfy the requirements of UGS connections followed by the requirements of rtPS and nrtPS connections. Finally, any residual bandwidth is distributed among the BE connections. This mechanism of allocating bandwidth might lead to fairness among the connections but can result in starvation of lower priority connections in the presence of a large number of high priority connections.

S.Kim and I.Yeom propose an uplink scheduling algorithm for TCP traffic for the BE class [31]. The proposed algorithm does not require explicit bandwidth request from a SS. It estimates the amount of bandwidth required by the SS based on the current sending rate of the connection at the SS. The purpose of the algorithm is to provide reasonable fairness among the SSs based on the max-min fairness criteria while providing high frame utilization. To provide fairness, the algorithm requires knowing the demand of each SS. The demand is defined as the amount of bandwidth requested for achieving the maximum throughput so as not to be limited by the channel bandwidth. It is shown via simulation that the algorithm provides reasonable fairness among SSs of the BE class.

2.3 Summary

In this chapter we provide details of the MAC and PHY layer characteristics according to the specifications in the IEEE 802.16-2004 standard [1]. We then categorize and discuss various proposals of uplink scheduling algorithms for WiMAX. The homogeneous class of algorithms contains legacy algorithms that were originally proposed for wired networks. The algorithms proposed in this category attempt to satisfy the QoS requirements of the multi-class traffic by assigning deadlines to packets or weights to the SSs. The hybrid class of algorithms contains schemes that basically combine two or more legacy algorithms. An important aspect of hybrid algorithms is the allocation of bandwidth among the traffic classes, which can significantly affect the QoS received by the SSs. The opportunistic category of algorithm exploits variations in channel quality as well as attempts to satisfy QoS requirements of the multi-class traffic.

Chapter 3. Representative WiMAX uplink scheduling algorithms

In this chapter, we provide detailed information of the representative uplink scheduling algorithms in WiMAX. As part of the implementation details, we highlight any assumptions and implementation decisions made in the process.

The IEEE 802.16-2004 [1] standard specifies provision of four traffic classes, Unsolicited Grant Service (UGS), real time Polling Service (rtPS), non real time Polling Service (nrtPS) and Best Effort (BE). According to the standard, the BS will provide Data Grant Burst IEs to the SSs at periodic intervals based upon the Maximum Sustained Traffic Rate (MSTR) of the SS. The size of the grants will be sufficient enough to hold the fixed length data associated with the SS. The IEEE 802.16-2005 standard [6] specifies an additional scheduling service called extended real-time Polling Service (ertPS). The ertPS is a scheduling service that builds on the efficiency of both UGS and rtPS services. For SSs belonging to the ertPS class, the BS can provide unicast grants in an unsolicited manner like in UGS, thus saving the latency of a bandwidth request. The SS can also change the size of the uplink allocation by using the piggyback mechanism or explicitly requesting bandwidth. The ertPS scheduling service is designed to support real-time applications that generate variable size data packets on a periodic basis, such as VoIP with silence suppression. Since the bandwidth allocation mechanism is specified for UGS connections, the implementation details in this chapter focus on ertPS, rtPS, nrtPS and BE traffic classes only. The standard requires each SS to be associated with certain

QoS parameters, whose values are specified by the SS upon admission into the network.

These parameters are:

- **Minimum Reserved Traffic Rate (MRTR):** This parameter specifies the minimum rate reserved for the SS. The rate is usually expressed in bits per second and specifies the minimum amount of data to be transported on behalf of the SS when averaged over time. The MRTR rate will only be honored when sufficient data is available for scheduling. When insufficient data exists, the requirement of this parameter will be satisfied by transmitting the available data as soon as possible.
- **Maximum Sustained Traffic Rate (MSTR):** This parameter specifies the peak information rate of the SS. The value, expressed in bits per second, does not limit the instantaneous rate of the SS but it is used to police the SS to ensure that it conforms to the value specified, on average, over time.
- **Maximum Latency (d_i^{req}):** This parameter specifies the maximum latency between the reception of a packet by the SS on its network interface and the forwarding of the packet to its RF interface.
- **Packet Loss (L):** This parameter, expressed as a percentage, indicates the percentage of packets dropped from the queue as a result of exceeding the maximum latency. Packet dropping is performed only for SSs with real-time needs as for such SSs delayed packets are useless.

The SSs specify values for all four parameters except the maximum latency and packet loss, which are specified by SSs of ertPS and rtPS classes only.

Following are variables/functions used in the pseudo-code for the scheduling algorithms:

- C: the uplink channel capacity.

- Ω_{Total} : the set of all admitted SSs
- Ω_{ertPS} : set of all SSs belonging to the ertPS class.
- Ω_{rtPS} : set of all SSs belonging to the rtPS class
- Ω_{nrtPS} : set of all SSs belonging to the nrtPS class.
- Ω_{BE} : set of all SSs belonging to the BE class.
- b_i^{alloc} : bandwidth allocated to SS i .
- $\text{deque}_i(P)$: remove packet P from the queue of SS i .
- $\text{enqueue}_i(P)$: insert packet P in the queue of SS i .
- $\text{size}_i(P, \gamma_i)$: retrieve the size of packet P from the queue of SS i . Convert the size of the packet to number of symbols based on the Signal to Interference Noise Ratio (SINR (γ_i)), of SS i .
- P_i^{HOL} : Head Of Line (HOL) packet of SS i .
- $\text{CreateIE}(\text{size}_i(P, \gamma_i), i)$: create an Information Element (IE) for SS i with $\text{size}_i(P, \gamma_i)$ number of symbols. After creation, the Information Element (IE) is added to the ULMAP message.
- $\text{Next}_i(P)$: Retrieve the next packet from the queue of SS i .
- $\text{drop}(\text{ertPS}, \text{rtPS})$: drop packets from the queue of all SSs of ertPS and rtPS class.

Due to the open-ended nature of the standard and ambiguity of some of the proposals, we have made some implementation decisions. To ensure that the number of SSs and their minimum required bandwidth do not exceed the channel capacity, we perform an admission control calculation as follows:

$$\sum_{i=1}^n (\text{MRTR}_i \text{ in symbols}) \leq \text{total uplink data symbols} \quad (3.1)$$

The number of uplink symbols available for data is calculated by subtracting overhead symbols (preamble, bandwidth request symbols and symbols for contention request opportunity) from the total uplink capacity.

Each SS can have multiple connections, all of them of the same traffic class. The number of connections per SS will be varied in the experiments so that the load per SS can be changed. Even though a SS can have multiple connections, bandwidth requests will be granted to a SS and not to each connection separately i.e. Grant Per SS (GPSS) is implemented. One slot per SS is reserved for bandwidth request and contention request opportunity.

A Time Division Duplex (TDD) frame structure is adopted, where the BS and SS each transmit on the same frequency separated in time. The IEEE 802.16-2004 standard [1] specifies the use of Information Elements in the Uplink MAP (ULMAP) message. For our purpose, the most important components of an IE are the Connection ID (CID) and the number of symbols allocated to the SS. The list of IEs constitutes the ULMAP message that is broadcast to all the SSs. In the pseudo-codes that follow, the function that creates the ULMAP message will be referred to as CreateIE().

3.1 Homogeneous Algorithms

This category contains scheduling algorithms originally proposed for wired networks, but have also been widely adopted by cellular technologies such as GSM, UMTS and WiMAX. We have chosen to implement Weighted Round Robin (WRR), Earliest Deadline First (EDF) and Weighted Fair Queuing (WFQ) schemes. The three

schemes selected satisfy QoS requirements of their users in diverse ways. The EDF algorithm allocates bandwidth according to the delay requirements of the SSs whereas the WRR and WFQ algorithms allocate bandwidth according to the weight assigned to the SSs. It is important to study schemes with such diverse behaviors so that we can determine which ones or a combination of them are suitable for the multi-class traffic in WiMAX.

3.1.1 Weighted Round Robin (WRR)

The WRR scheduling algorithm originally proposed for ATM traffic in [13] has been implemented in [12] to evaluate the IEEE 802.16 MAC layer on how effectively it supports QoS requirements of the multi-class traffic (see Figure 3-1). The algorithm is executed at the beginning of every frame at the Base Station (BS). At the start of a frame, the WRR algorithm determines the allocation of bandwidth among the SSs based on their weights. A critical portion of the WRR scheme is assigning weights to each SS. The weights are assigned to reflect the relative priority and QoS requirements of the SSs. Since MRTR is one of the parameters specified by a SS that reflect its QoS requirements, we assign weight to each SS with respect to its MRTR as follows:

$$W_i = \text{MRTR}_i / \sum_{j=1}^n \text{MRTR}_j \quad (3.2)$$

where, W_i = weight of SS i.

n = number of SSs.

```

1. //Drop packets of ertPS and rtPS SSs that have missed their deadline
2. drop(ertPS, rtPS)
3. Assign weights to SSs using (3.2)
4. //Allocate bandwidth according to the weight of SSs
5. for  $i \in \Omega_{\text{Total}}$ 
6.      $b_i^{\text{alloc}} = b_i^{\text{alloc}} + W_i * C$ 
7. end for Still here you need to check for the capacity
8. //Select packets for transmission based on bandwidth allocated
9. for  $i \in \text{Conn}$  do
10.     CreateIE() //create IEs based on allocated bandwidth
11. end for

```

Figure 3-1: Pseudo-code of WRR algorithm

3.1.2 Earliest Deadline First (EDF)

Earliest Deadline First (EDF) is one the most widely used scheduling algorithms for real-time applications as it selects SSs based on their delay requirements. The algorithm assigns deadline to arriving packets of a SS (see Figure 3-2). Since each SS specifies a value for the maximum latency parameter, the arrival time of a packet is added to the latency to form the tag of the packet. The value of maximum latency for SSs of the nrtPS and BE classes is set to infinity. In the following pseudo-code, $\min_{\text{deadline}}(P)$ refers to the packet with the earliest deadline. The algorithm below is executed upon arrival of every packet.

```

1. //Drop packets of ertPS and rtPS SSs that have missed their deadline
2. drop(ertPS, rtPS)
3. Assign deadline upon arrival of a packet.
4. //Assign bandwidth to the SS with the packet with earliest deadline
5. while(C>0)
6.      $b_i^{\text{alloc}} = b_i^{\text{alloc}} + \text{size}_i(\min_{\text{deadline}}(\text{P}), \gamma_i)$ 
7.     CreateIE()
8.      $C = C - \text{size}_i(\min_{\text{deadline}}(\text{P}), \gamma_i)$ 
9. end while

```

Figure 3-2: Pseudo-code of EDF algorithm

3.1.3 Weighted Fair Queuing (WFQ)

Both WFQ and WRR scheduling algorithms assign weights to SSs. Unlike the WRR algorithm, the WFQ algorithm also considers the packet size and the channel capacity when allocating bandwidth to the SSs (see Figure 3-5). An arriving packet is tagged with finish time that is calculated based on the weight of the SS, the packet size and the uplink channel capacity. In WFQ, the weight of a SS is calculated in the same way as it is in WRR. Once the weight is assigned, arriving packets of the SS are stamped with virtual finish time as follows:

$$S_k^i = \max \{F_i^{k-1}, V(a_i^k)\} \quad (3.3)$$

$$F_i^k = S_k^i + L_k^i / \Phi_i \quad (3.4)$$

where, S_k^i is the Start time of k^{th} packet of SS i ;

F_i^{k-1} is the Finish time of $(k-1)^{\text{th}}$ packet of SS i ;

$V(a_i^k)$ is the Virtual time of the k^{th} packet of SS i ;

a_i^k arrival time of k^{th} packet of SS i ;

F_i^k is the Finish time of k^{th} packet of SS i ;

L_k^i is the Length of k^{th} packet of SS i ;

Φ_i is the reserved rate of SS i , where $\Phi_i = C * W_i$;

W_i is the weight assigned to SS i .

The complexity of WFQ is high due to two main reasons: selection of the next queue to serve and computation of the virtual time. The complexity of the former is fixed to $O(\log N)$ whereas the complexity of the latter is $O(N)$, where N is the number of SSs. We use an approach described in [32] that provides a practical implementation of WFQ for packet scheduling in IP networks. The transmission of a packet will trigger an update to the virtual time of its SS. Since the scheduling algorithm is implemented at the BS, the virtual time will be updated once a packet has been selected for transmission. For a time interval τ , the virtual time $V(t)$ is updated as follows:

$$V(t_{j-1} + \tau) = V(t_{j-1}) + \frac{\tau}{\sum_{i \in B_j} \Phi_i} \quad (3.5)$$

where, $\tau \leq t_j - t_{j-1}$, $j = 2, 3, \dots$

B_j is the set of busy SSs.

Figures 3-3 and 3-4 describe the pseudo-code of the actions that need to be performed upon arrival and selection of a packet, respectively. In the following pseudo code, $\min_{\text{finishtime}}(P)$ refers to the packet with the smallest finish time.

1. //Reset the virtual and finish time
2. If system idle
3. $V(t) = 0$
4. $F_i^{k-1} = 0$
5. end if
6. Calculate S_k^i and F_i^k using (3.3) and (3.4)
7. if $i \in B$ then
8. $B = B + i$ //Add SS i to the busy set
9. end if
10. assign F_i^k to packet k

Figure 3-3: Action upon arrival of packet k of SS i – arrive(i,k)

1. Update $V(t)$ using (3.5)
2. CreateIE()
3. if connection i not backlogged
4. $B = B - i$ //Remove i from the busy set
5. end if

Figure 3-4: Action upon selection of packet k of SS i – select (i,k)

1. //Drop packets of ertPS and rtPS SSs that have missed their deadline
2. drop(ertPS, rtPS)
3. Upon arrival of packet k of SS i ,
4. arrive(i,k)
5. enqueue(k)
6. //Assign bandwidth to the SS with the packet with earliest finish time
7. while($C > 0$)
8. //Allocate bandwidth to SS i i.e. SS with the packet with earliest finish time
9. $b^{\text{alloc}} = b^{\text{alloc}} + \text{size}(\min_{\text{finishtime}}(P), \gamma)$
10. $C = C - \text{size}(\min_{\text{finishtime}}(P), \gamma)$
11. select(i,k)
12. end while

Figure 3-5: Pseudo-code of WFQ algorithm

3.2 Hybrid Algorithms

The two hybrid algorithms selected for evaluation use a different mechanism of overall bandwidth allocation. An important aspect of hybrid algorithms is allocation of bandwidth among the traffic classes of WiMAX. The two algorithms we have selected perform this task in different ways. The hybrid (EDF+WFQ+FIFO) algorithm uses a strict priority mechanism for inter-class bandwidth allocation, whereas the hybrid (EDF+WFQ) algorithm allocates bandwidth among traffic classes based on the number of SSs and their MRTR in each class.

3.2.1 Hybrid (EDF+WFQ+FIFO)

The hybrid algorithm proposed in [23] uses strict priority mechanism for overall bandwidth allocation (see Figure 3-6). The EDF scheduling algorithm is used for SSs of ertPS and rtPS classes, the WFQ algorithm is used for SSs of nrtPS class and FIFO is used for SSs of BE class. The EDF and WFQ algorithms are implemented as described in sections 3.1.2 and 3.1.3 respectively. FIFO is used for BE class as SSs of this class do not have any QoS requirements. In the following pseudo-code, `queue(ConnertPS)`, `queue(ConnrtPS)`, `queue(ConnnrtPS)` and `queue(ConnBE)` refer to packet queues of SSs from ertPS, rtPS, nrtPS and BE classes, respectively. The bandwidth distribution among the traffic classes is executed at the beginning of every frame whereas the EDF, WFQ and FIFO algorithms are executed at the arrival of every packet.

```

1. //Drop packets of ertPS and rtPS SSs that have missed their deadline
2. drop(ertPS,rtPS)
3. Upon arrival of packet  $k$  of connection  $i$ 
4.     if( $i \in \Omega_{ertPS}, \Omega_{rtPS}$ )
5.         Assign deadline to packet  $k$ 
6.     end if
7.     if( $i \in \Omega_{nrtPS}$ )
8.         arrive( $i,k$ )
9.     end if
10. enqueue $_i(k)$ 
11.
12. //Execute the EDF algorithm for ertPS and rtPS SSs
13. for  $i \in \Omega_{ertPS}, \Omega_{rtPS}$ 
14.     while( $C > 0$  and (queue( $\Omega_{ertPS}$ ) or queue( $\Omega_{rtPS}$ )) != NULL)
15.          $b_i^{alloc} = b_i^{alloc} + size_i(\min_{deadline}(P), \gamma_i)$ 
16.         CreateIE()
17.          $C = C - size_i(\min_{deadline}(P), \gamma_i)$ 
18.     end while
19. end for
20.
21. //Execute the WFQ algorithm for nrtPS SSs
22. for  $i \in \Omega_{nrtPS}$ 
23.     while( $C > 0$  and queue( $\Omega_{nrtPS}$ ) != NULL)
24.          $b_i^{alloc} = b_i^{alloc} + size_i(\min_{finishtime}(P), \gamma_i)$ 
25.          $C = C - size_i(\min_{finishtime}(P), \gamma_i)$ 
26.         select( $i,k$ )
27.     end while
28. end for
29.
30. //Execute the FIFO algorithm for BE SSs
31. while( $C > 0$  and queue( $\Omega_{BE}$ ) != NULL)
32.      $b^{alloc} = b^{alloc} + deque(P)$ 
33.     CreateIE()
34.      $C = C - size(P, \gamma)$ 
35. end while

```

Figure 3-6: Pseudo-code of hybrid (EDF+WFQ+FIFO) algorithm

3.2.2 Hybrid (EDF+WFQ)

K.Vinay *et al.* [24] propose a hybrid algorithm that uses the EDF scheduling algorithm for SSs of ertPS and rtPS classes and WFQ algorithm for SSs of nrtPS and BE classes (see Figure 3-7). Although the mechanism of overall bandwidth distribution is not specified, it is mentioned in [24] that bandwidth is allocated in a fair manner. Just as the hybrid (EDF+WFQ+FIFO) algorithm, the overall bandwidth distribution is executed at the beginning of every frame while the EDF and WFQ algorithms are executed at the arrival of every packet. The following is the overall bandwidth allocation scheme adopted in our implementation:

$$BW_{ertPS,rtPS} = C * \left(\sum_{i \in ertPS,rtPS} MRTR_i \right) / \left(\sum_{j=1}^n MRTR_j \right) \quad (3.6)$$

$$BW_{nrtPS,BE} = C * \left(\sum_{i \in nrtPS,BE} MRTR_i \right) / \left(\sum_{j=1}^n MRTR_j \right) \quad (3.7)$$

```

1. //Drop packets of ertPS and rtPS SSs that have missed their deadline
2. drop(ertPS,rtPS)
3. Assign overall bandwidth using (3.6) and (3.7)
4. Upon arrival of packet  $k$  of SS  $i$ 
5.   if( $i \in \Omega_{ertPS}, \Omega_{rtPS}$ )
6.     Assign deadline to packet  $k$ 
7.   end if
8.   if( $i \in \Omega_{nrtPS}, \Omega_{BE}$ )
9.     arrive( $i,k$ )
10.  end if
11. enqueue( $k$ )
12.
13. //Execute the EDF algorithm for ertPS and rtPS SSs
14. for  $i \in \Omega_{ertPS}, \Omega_{rtPS}$ 
15.   while ( $BW_{ertPS,rtPS} > 0$  and ( $queue(\Omega_{ertPS})$  or  $queue(\Omega_{rtPS})$ ) != NULL)
16.      $b_i^{alloc} = b_i^{alloc} + size_i(\min_{deadline}(P), \gamma_i)$ 
17.     CreateIE()
18.      $BW_{ertPS,rtPS} = BW_{ertPS,rtPS} - size_i(\min_{deadline}(P), \gamma_i)$ 
19.   end while
20. end for
21.
22. //Carry-over any bandwidth remaining from execution of EDF (line 13-19)
23. if( $BW_{ertPS,rtPS} > 0$ )
24.    $BW_{nrtPS,BE} = BW_{nrtPS,BE} + BW_{ertPS,rtPS}$ 
25. end if
26.
27. //Execute WFQ algorithm for nrtPS and BE SSs
28. for  $i \in \Omega_{nrtPS}, \Omega_{BE}$ 
29.   while ( $BW_{nrtPS,BE} > 0$  and ( $queue(\Omega_{nrtPS})$  or  $queue(\Omega_{BE})$ ) != NULL)
30.      $b_i^{alloc} = b_i^{alloc} + size_i(\min_{finishtime}(P), \gamma_i)$ 
31.     CreateIE()
32.      $BW_{nrtPS,BE} = BW_{nrtPS,BE} - size_i(\min_{finishtime}(P), \gamma_i)$ 
33.     select( $i,k$ )
34.   end while
35. end for

```

Figure 3-7: Pseudo-code of hybrid (EDF+WFQ) algorithm

3.3 Opportunistic Algorithms

The two opportunistic algorithms selected for evaluation use priority and utility functions in calculating the relative priority of the SSs. The Queuing Theoretic algorithm uses queuing theory and sigmoid function in assigning utility to the SSs whereas the cross layer algorithm uses delay of HOL packet and average throughput. Both these algorithms use distinguishing mechanisms in satisfying the QoS requirements of the SSs and therefore provide a good representation of the algorithms in this category.

3.3.1 Cross-Layer scheduling algorithm

The algorithm proposed in [27] uses a priority function that incorporates delay of HOL packet and the minimum required throughput of the SSs in its formulation. The SS with the highest priority is selected to transmit in the frame (see Figure 3-8). The priority of a SS is calculated based on the traffic class it belongs to. Although the priority function for SSs of ertPS class is not defined in [27], we use the same function specified for SSs of rtPS class. We have chosen to evaluate this algorithm as it incorporates all the required QoS parameters in the priority functions. More specifically, the MRTR and maximum latency is used in the priority function for ertPS and rtPS SSs and only the MRTR is used in the priority function for nrtPS SSs. The priority of BE SSs depends only on the channel quality of the SS as the BE scheduling service does not have any QoS requirements. The algorithm is executed at the Base Station (BS) at the beginning of every frame whereby priority is assigned to each SS. Subsequently, the SS with the highest priority is selected for transmission in the frame. The scheme uses a coefficient for each class in the priority functions as shown in the following equations:

Priority function for SSs of ertPS/rtPS class:

$$\phi_i(t) = \begin{cases} (\beta_{\text{ertPS}}, \beta_{\text{rtPS}}) * \frac{R_i(t)}{R_N} * \frac{1}{F_i(t)} & \text{if } F_i(t) \geq 1, R_i(t) \neq 0 \\ (\beta_{\text{ertPS}}, \beta_{\text{rtPS}}) & \text{if } F_i(t) < 1, R_i(t) \neq 0 \\ 0 & \text{if } R_i(t) = 0 \end{cases} \quad (3.8)$$

$$F_i(t) = d_i^{\text{req}} - \Delta d_i - W_i(t) + 1 \quad (3.9)$$

Priority function for SSs of nrtPS class:

$$\phi_i(t) = \begin{cases} \beta_{\text{nrtPS}} * \frac{R_i(t)}{R_N} * \frac{1}{F_i(t)} & \text{if } F_i(t) \geq 1, R_i(t) \neq 0 \\ \beta_{\text{nrtPS}} & \text{if } F_i(t) < 1, R_i(t) \neq 0 \\ 0 & \text{if } R_i(t) = 0 \end{cases} \quad (3.10)$$

$$F_i(t) = \hat{\eta}_i(t) / \eta_i \quad (3.11)$$

Priority function for SSs of BE class:

$$\phi_i(t) = \beta_{\text{BE}} * \frac{R_i(t)}{R_N} \quad (3.12)$$

where,

$\phi_i(t)$ is the priority of SS i at time t .

$\beta_{\text{ertPS}}, \beta_{\text{rtPS}}, \beta_{\text{nrtPS}}, \beta_{\text{BE}}$ are coefficients for SSs of ertPS, rtPS, nrtPS and BE classes

respectively such that $\beta_{\text{ertPS}}, \beta_{\text{rtPS}}, \beta_{\text{nrtPS}}, \beta_{\text{BE}} \in [0,1]$

$R_i(t)$ refers to the amount of data that can be carried by one time slot based on the channel quality of the SS.

R_N refers to the amount of data that can be carried by one time slot based on the highest modulation and coding scheme.

$F_i(t)$ refers to the delay satisfaction indicator for SSs of ertPS and rtPS classes and throughput satisfaction indicator for SSs of nrtPS class.

d_i^{req} refers to the delay requirement of SS i

Δd_i refers to the guard time ahead of T_i . $\Delta T_i \in [0, T_i]$

$\hat{\eta}_i(t)$ refers to the average throughput of SS i at time t .

η_i refers to the Minimum Reserved Traffic Rate (MRTR) of SS i .

$W_i(t)$ refers to the delay of the Head Of Line (HOL) packet of SS i at time t , such that

$W_i(t) \in [0, T_i]$.

The class coefficients $(\beta_{\text{ertPS}}, \beta_{\text{rtPS}}, \beta_{\text{nrtPS}}, \beta_{\text{BE}})$ can be set to reflect the priority of the traffic classes. We adopt the same coefficients as in [27] and normalize them to reflect the relative priority of the SSs. Although, coefficient for SSs of ertPS class is not defined in [27], we use a higher value than coefficients for the other 3 classes to reflect their relative priority.

```

1. //Drop packets of ertPS and rtPS SSs that have missed their deadline
2. drop(ertPS,rtPS)
3.
4. //Assign priority to ertPS and rtPS SSs
5. for  $i \in \Omega_{ertPS}, \Omega_{rtPS}$ 
6.     assign  $\phi(t)$  according to (3.8),(3.9)
7. end for
8.
9. //Assign priority to nrtPS SSs
10. for  $i \in \Omega_{nrtPS}$ 
11.     assign  $\phi(t)$  according to (3.10),(3.11)
12. end for
13.
14. //Assign priority to BE SSs
15. for  $i \in \Omega_{BE}$ 
16.     assign  $\phi(t)$  according to (3.12)
17. end for
18.
19. //Select the SS with the highest priority
20.  $i_{max} = \max_i(\phi(t))$ 
21.
22. //Allocate bandwidth to the SS with highest priority as long as it has
    backlogged packets
23. while ( $C > 0$  and Next (P)  $\neq \phi$ )
24.      $b^{alloc} = b^{alloc} + \text{size}(\text{Next (P)}, \gamma)$ 
25.     CreateIE()
26.      $C = C - \text{size}(\text{Next (P)}, \gamma)$ 
27. end while

```

Figure 3-8: Pseudo-code of Cross Layer algorithm

3.3.2 Queuing Theoretic scheduling algorithm

This uplink scheduling algorithm [29] uses a queuing model to satisfy the QoS requirements of the multi-class traffic. The algorithm uses sigmoid functions to assign utility to each SS (see Figure 3-9). The input to the sigmoid function depends on the traffic class the SS belongs to. The SS with lowest utility is given the highest transmission priority.

The proposed resource management model contains a closely coupled scheduling algorithm and a CAC scheme. The focus of our study is to evaluate the performance of the scheduling algorithm only, and therefore the CAC is not implemented. We have chosen to evaluate this algorithm as it uses queuing theory to satisfy the QoS requirements of the scheduling services. It also uses thresholds to limit the bandwidth allocated to SSs of each class. This is a unique way of limiting bandwidth allocation and ensuring that lower priority SSs do not starve. The algorithm is executed at the Base Station (BS) at the beginning of every frame. The utility of each SS is calculated at the start of a frame and the bandwidth is allocated accordingly. The utility function of a SS is defined as follows:

$$U_{BE}(b_i) = \begin{cases} 1 & \text{if } b_i \geq 0 \\ 0 & \text{Otherwise} \end{cases} \quad (3.13)$$

$$U_{ertPS,rtPS}(b_i) = 1 - \frac{1}{1 + \exp(-g_{rt}(d(\gamma, \lambda, b_i) - d_i^{req} - h_{rt}))} \quad (3.14)$$

$$U_{nrtPS} = \frac{1}{1 + \exp(-g_{nrt}(\tau(\gamma, \lambda, b_i) - MRTR_i - h_{nrt}))} \quad (3.15)$$

where,

U_{BE} refers to utility for SSs of BE class.

$U_{ertPS,rtPS}$ refers to utility for SSs of ertPS and rtPS classes respectively.

U_{nrtPS} refers to utility for SSs of nrtPS class.

g_{rt}, g_{nrt} are parameters of sigmoid function that determine the steepness (sensitivity of the utility function to delay or throughput requirement).

h_{rt}, h_{nrt} are parameters of the sigmoid function that represent the centre of the utility function.

$d(\gamma, \lambda, b_i)$ is the average delay as a function of SINR(γ), PDU arrival rate (λ) and bandwidth allocated (b).

$\tau(\gamma, \lambda, b_i)$ is the average throughput as a function of SINR(γ), PDU arrival rate (λ) and bandwidth allocated (b).

MRTR_{*i*} is the Minimum Reserved Traffic Rate of SS *i*.

The goal of the scheme is to maximize the utility of all the SSs in the network. For this purpose, an optimization problem is formulated. Due to the exponential time complexity of the optimal approach, the authors implement the water-filling algorithm [29]. An important part of the algorithm is determining bandwidth thresholds of each traffic class. An optimization approach is discussed that will calculate thresholds to maximize the average system revenue under connection level QoS constraints such as call blocking probability. Since we do not implement a CAC scheme and therefore call blocking probability is not a parameter in our system, to be able to compare all the algorithms under the same constraints, we calculate the thresholds as follows:

$$T_{ertPS} = \frac{\sum_{i=1}^{n_{ertPS}} MRTR_i}{\sum_{j=1}^n MRTR_j} * C \quad (3.16)$$

$$T_{rtPS} = \frac{\sum_{i=1}^{n_{rtPS}} MRTR_i}{\sum_{j=1}^n MRTR_j} * C \quad (3.17)$$

$$T_{nrtPS} = \frac{\sum_{i=1}^{n_{nrtPS}} MRTR_i}{\sum_{j=1}^n MRTR_j} * C \quad (3.18)$$

$$T_{BE} = \frac{\sum_{i=1}^{n_{BE}} MRTR_i}{\sum_{j=1}^n MRTR_j} * C \quad (3.19)$$

where,

$\sum_{j=1}^n MRTR_j$ refers to the sum of MRTR of all the SSs in the network.

$\sum_{i=1}^{n_{ertPS}} MRTR_i, \sum_{i=1}^{n_{rtPS}} MRTR_i, \sum_{i=1}^{n_{nrtPS}} MRTR_i, \sum_{i=1}^{n_{BE}} MRTR_i$ refers to the sum of MRTR of SSs from

ertPS, rtPS, nrtPS and BE traffic class, respectively.

```

1. //Drop packets of ertPS and rtPS SSs that have missed their deadline
2. drop(ertPS,rtPS)
3.
4. //Allocate bandwidth to ertPS SSs equivalent to their MRTR
5. for  $i \in \Omega_{ertPS}$ 
6.     if ( $C > 0$ )
7.          $b_i^{alloc} = b_i^{alloc} + MRTR_i$ 
8.         CreateIE()
9.          $C = C - MRTR_i$ 
10.    end if
11. end for
12.
13. //Allocate bandwidth to rtPS SSs equivalent to their MRTR
14. for  $i \in \Omega_{rtPS}$ 
15.     if ( $C > 0$ )
16.          $b_i^{alloc} = b_i^{alloc} + MRTR_i$ 
17.         CreateIE()
18.          $C = C - MRTR_i$ 
19.     end if
20. end for
21.
22. //Allocate bandwidth to nrtPS SSs equivalent to their MRTR
23. for  $i \in \Omega_{nrtPS}$ 
24.     if ( $C > 0$ )
25.          $b_i^{alloc} = b_i^{alloc} + MRTR_i$ 
26.         CreateIE()
27.          $C = C - MRTR_i$ 
28.     end if
29. end for
30.
31. //Allocate bandwidth to BE SSs equivalent to size of one packet
32. for  $i \in \Omega_{BE}$ 
33.     if ( $C > 0$ )
34.          $b_i^{alloc} = size_i(Next_i(P), \gamma_i)$ 
35.         CreateIE()
36.          $C = C - size_i(Next_i(P), \gamma_i)$ 
37.     end if
38. end for

```

```

40. //Assign residual bandwidth according to the utility of the SSs
41. while(C>0 and  $\Omega_{\text{Total}} \neq \text{Empty}$ )
42.   Calculate utility using (3.13), (3.14), (3.15)
43.    $i = \min(U(b_i))$  //Select connection with minimum utility
44.    $b_i^{\text{alloc}} = b_i^{\text{alloc}} + \text{size}_i(\text{Next}_i(P), \gamma_i)$ 
45.   if( $b_i \geq \text{MSTR}_i$ )
46.      $\Omega_{\text{Total}} = \Omega_{\text{Total}} - i$ 
47.   end if
48.   if( $\sum_{i \in \text{Conn}_{\text{ertPS}}} b_i = T_{\text{ertPS}}$ )
49.      $\Omega_{\text{Total}} = \Omega_{\text{Total}} - \Omega_{\text{ertPS}}$ 
50.   end if
51.   if( $\sum_{i \in \text{Conn}_{\text{rtPS}}} b_i = T_{\text{rtPS}}$ )
52.      $\Omega_{\text{Total}} = \Omega_{\text{Total}} - \Omega_{\text{rtPS}}$ 
53.   end if
54.   if( $\sum_{i \in \text{Conn}_{\text{nrtPS}}} b_i = T_{\text{nrtPS}}$ )
55.      $\Omega_{\text{Total}} = \Omega_{\text{Total}} - \Omega_{\text{nrtPS}}$ 
56.   end if
57.   if( $\sum_{i \in \text{Conn}_{\text{BE}}} b_i = T_{\text{BE}}$ )
58.      $\Omega_{\text{Total}} = \Omega_{\text{Total}} - \Omega_{\text{BE}}$ 
59.   end if
60. end while

```

Figure 3-9: Pseudo-code of Queuing Theoretic scheduling algorithm

3.4 Complexity Analysis

The complexity of legacy algorithms is well known based on comprehensive analysis done in the literature. The complexity of the WRR algorithm is known to be constant with respect to the number of SSs i.e. $O(1)$ [33]. It has been discussed in [34] that the complexity of the WFQ algorithm is $O(N)$, where N is the number of SSs. The

complexity of the EDF algorithm is also $O(N)$ [35]. The hybrid (EDF+WFQ+FIFO) and hybrid (EDF+WFQ) have a complexity of $O(N)$, just as their legacy components.

The computational complexity of the opportunistic algorithms has not been discussed by their authors. Based on our analysis of the algorithms, the complexity of the Cross Layer algorithm is $O(N)$. The complexity of the Cross Layer algorithm is dominated by the portion of the code that assigns priority to the SSs (lines 1-10 in Figure 3-8) that loops through all the SSs. The complexity of the Queuing Theoretic algorithm is $O(N^2)$. Its complexity is dominated by the portion of the algorithm that assigns one unit of bandwidth to the SS with the minimum utility, until there are no backlogged packets or all the available bandwidth has been assigned (lines 30-50 in Figure 3-9). Since the number of iterations of the while loop (starting at line 30) at the most can be equivalent to the number of SSs and within each iteration utility is assigned to all the SSs, it results into a complexity of $O(N^2)$. Table 3-1 lists the scheduling algorithms in increasing order of their complexity, where N is the number of SSs.

Table 3-1: Complexity of representative uplink scheduling algorithms

<u>Algorithm</u>	<u>Complexity</u>
WRR	$O(1)$
EDF	$O(N)$
WFQ	$O(N)$
Cross Layer	$O(N)$
Hybrid (EDF+WFQ)	$O(N)$
Hybrid (EDF+WFQ+FIFO)	$O(N)$
Queuing Theoretic	$O(N^2)$

3.5 Summary

In this chapter we described implementation details of the scheduling algorithms considered for evaluation. The algorithms in each category have been chosen based on how effectively they demonstrate characteristics of the respective category. We also made some implementation decisions in the process. These include a common CAC algorithm, allowing multiple connections per SS and adoption of TDD frame structure. In chapter 4, we will analyze the performance of the scheduling algorithms with respect to the characteristics of the IEEE 802.16 MAC layer.

Chapter 4. Performance Analysis

In this chapter we study the performance of the scheduling algorithms discussed in Chapter 3 under different conditions. These conditions include studying the performance of the algorithms under various concentrations of traffic and under the characteristics of the IEEE 802.16 MAC layer such as uplink burst preamble, frame length and bandwidth request mechanisms. The simulation tool chosen for the experiments is Berkeley's Network Simulator 2 (NS-2) [36] with an add-on developed by members of Network and Distributed Systems Laboratory at Chang Gung University in Taiwan [37]. Numerous modifications to the NS-2 extension for WiMAX were made. We will provide a detailed description of the changes made to the tool, including the traffic model, transmission modes and choice of MAC layer parameters such as length of uplink preamble and allocation start time. We will also discuss the metrics used to evaluate the schemes. The experiments are run on a Linux PC with a 2 GHz Intel Core Duo processor and Random Access Memory (RAM) of 2 Giga bytes. The version of Linux installed on this PC is Ubuntu. Each experiment takes approximately 10 minutes to complete execution i.e. a simulation time of 50 seconds corresponds to 10 minutes in real-time. The execution time varies depending on the number of SSs and the traffic load used in the experiment.

4.1 NS-2 and WiMAX

The NS-2 extension for WiMAX PMP mode [37], version 1.06, implements the Convergence Sub layer (CS), the MAC Common Part Sub layer (CPS) and the PHY layer. The module implements functionalities such as ranging, MAC Management,

Scheduler, IP-SFID mapping and SFID-TCID mapping. MAC management contains messages such as Uplink Channel Descriptor (UCD), Downlink Channel Descriptor (DCD), Bandwidth Request (BW-Req) message and Uplink MAP (UL-MAP) message and Downlink MAP (DL-MAP) message. IP-Service Flow ID (IP-SFID) mapping is used to record the characteristics of the packets coming from the upper layers while Service Flow ID-Transport Connection ID (SFID-TCID) mapping is used to map the QoS characteristics of the SSs to one of the four traffic classes (ertPS, rtPS, nrtPS and BE). Our modifications to the extension include changes to the uplink scheduler, support for OFDM, additional parameters associated with a SS such as maximum latency, packet loss, MRTR and MSTR. MAC layer parameters such as allocation start time and uplink burst preamble were added to the tool and their values set according to the IEEE 802.16-2004 standard [1]. Our major contribution to the NS-2 extension for WiMAX is the addition of seven representative uplink scheduling algorithms from the three categories discussed in chapter 3. An important modification to the extension was to change the bandwidth management function so that the list of Information Elements (IEs) in the ULMAP message is created according to the transmission order determined by the scheduling algorithm. A traffic model was also added to the extension. This model implements VoIP traffic for the ertPS class, video streaming traffic for the rtPS class, FTP traffic for the nrtPS class and HTTP traffic for the BE class. This traffic model is representative of the traffic characteristics of the scheduling classes as specified in the IEEE 802.16-2004 standard [3]. Details of the traffic model are discussed later in the chapter.

The wireless channel is modeled using a block-fading model, which is suitable for slow varying channel conditions as in WiMAX PMP mode. The channel quality of each SS remains constant per frame, but is allowed to vary from frame to frame based on the transmission modes defined in Table 4-1. Therefore, Adaptive modulation and Coding (AMC) is implemented based on six transmission modes as described in the IEEE 802.16-2004 standard [1]. The channel quality is captured using a single parameter, the instantaneous Signal to Noise Ratio (SNR), which remains constant for the duration of the frame.

Each packet, upon arrival, is tagged with a key and a deadline. Packets are inserted in the queue of the SS in increasing order of the key. To enable this, a priority queue class was implemented that would allow us to insert packets in order of the key. Additionally to collect statistics for analysis, we store the average throughput, average delay and packet loss for each SS.

4.2 Simulation Model

4.2.1 Transmission Modes

We use six transmission modes in our implementation whose values are defined in Table 4-1. The raw bit rate varies depending on the modulation technique and coding rate used. These values are in conformance with the IEEE 802.16-2004 standard for channel bandwidth of 20MHz.

Table 4-1: IEEE 802.16-2004 Transmission Modes

Mode	1	2	3	4	5	6
Modulation	QPSK	QPSK	16QAM	16QAM	64QAM	64QAM
Coding rate	1/2	3/4	1/2	$\frac{3}{4}$	2/3	3/4
Raw Bit Rate (Mbits/sec)	15.36	23.04	30.72	46.08	61.44	69.12

4.2.2 Traffic Model

We have implemented four different traffic sources, one for each of the traffic classes. VoIP traffic is modeled for SSs of ertPS class, video streaming for SSs of rtPS class, FTP for SSs of nrtPS class and HTTP for SSs of BE class. The values of all the traffic parameters are based on one connection per SS.

VoIP Traffic – ertPS class (Class 2)

An important characteristic of VoIP traffic is the presence of talk spurt and silence spurt. The length of the talk spurt and silence spurt depends on the encoding scheme used. There are numerous encoding schemes such as G.711, G.722, and Adaptive Multi-Rate (AMR) that result in different bandwidth requirements. We have used the model described in [38] that assumes the AMR codec and a packet size of 23 bytes. To suit our simulation needs, we have modified the model and added some parameters associated with a connection as specified in the IEEE 802.16-2004 standard.

Table 4-2: VoIP traffic parameters

Parameter	Value (1 connection per SS)
Minimum Reserved Traffic Rate (MRTR)	25 Kbps
Average Traffic rate	44 Kbps
Maximum Sustained Traffic Rate (MSTR)	64 Kbps
Maximum Latency	100ms
Tolerated packet loss	10%
Talk spurt length	Exponential random with $\mu=147$ ms
Silence length	Exponential random with $\mu=167$ ms

Video Streaming – rtPS class (Class 3)

The values of a traffic source modeled for video streaming highly depend on the video trace. Based on the discussions in [38] and [39], video streaming has been divided into two broad categories, low quality (64-500Kbps) and high quality (1.5-10Mbps). We implement the low quality video streaming model with packet size uniformly distributed between 150 bytes and 300 bytes.

Table 4-3: Video Streaming parameters

Parameter	Value (1 connection per SS)
Minimum Reserved Traffic Rate (MRTR)	64 Kbps
Average traffic rate	282 Kbps
Maximum Sustained Traffic Rate (MSTR)	500 Kbps
Maximum Latency	150ms
Tolerated packet loss	5%

FTP – nrtPS class (Class 4) and HTTP – BE class (Class5)

We have implemented an FTP traffic generator with a constant packet size of 150 bytes. A value of 45 Kbps for Minimum Reserved Traffic Rate (MRTR) and 500Kbps for Maximum Sustained Traffic Rate is used for each FTP source. An HTTP traffic model is used for the BE class. A value of 64 Kbps for MSTR and a packet size of 100 bytes are adopted in the mode. Although SSs of the BE class do not have MRTR parameter associated with them, since implementation of schemes such as WRR and WFQ require a weight to be assigned to the SSs based on the MRTR, a value of 1 Kbps is used for MRTR.

4.2.3 Simulation Parameters

Table 4-4 lists parameters that are constant throughout the simulation study whereas table 4-5 lists parameters whose values are varied in the experiments to study the performance of the scheduling algorithms. According to the IEEE 802.16-2004 standard [1], the allocation start time for WirelessMAN-OFDM PHY layer can either be the start of the uplink sub-frame in the current frame or start of the uplink sub-frame in the next frame. The allocation start time is the reference point for the information in the UL-MAP message. In our experiments, the value for allocation start time is set such that all the allocation in the UL-MAP will start in the current frame after the last specified allocation in the DL-MAP. This point in the frame is referred to as Adaptive Time Division Duplexing (ATDD) split. The default values as specified in NS-2 are used for PHY layer parameters such as radio propagation model and antenna type.

Table 4-4: Fixed Simulation parameters

Parameter	Value
Physical Layer	WirelessMAN-OFDM
Uplink burst preamble	16 symbols
Allocation Start Time	ATDD split
Frame structure	TDD
Bandwidth	20MHz
DL:UL frame ratio	1:1
OFDM symbol duration	12.5 μ s
Node placement	Random
Simulation grid size	1000mx1000m
Simulation time	50 seconds

Table 4-5: Variable Simulation parameters

Parameter	Value
Number of SSs	1-36
Ratio of SS (ertPS:rtPS:nrtPS:BE)	1:1:1:3, 1:1:2:2, 1:1:3:1, 1:2:2:1, 1:3:1:1, 2:2:1:1, 3:1:1:1
Frame Length	2.5ms,4ms,5ms,8ms,10ms,12ms,20ms

4.2.4 Performance Metrics

The following statistics are collected for each simulation run:

Average Throughput ($\hat{\tau}$): The amount of data selected for transmission by a user per unit time. The value is expressed in Kbps and calculated using an exponential moving average as follows:

$$\hat{\tau}_t = \alpha * (\tau_t) + (1 - \alpha) \hat{\tau}_{t-1}$$

where, $\hat{\tau}_t$ and $\hat{\tau}_{t-1}$ are the average throughput at frame t and $t-1$ respectively.

τ_t is the attainable rate at frame t . This value corresponds to the raw bit rate based on the channel quality and thus the modulation and coding scheme as described in Table 4-1.

A value of 0.001 for α as used in [27] is adopted for all our experiments with a window size of 1000ms. This metric will allow us to understand the effect of preamble overhead and the relative priority given to the SSs.

Average Queuing delay (\hat{d}): The time between the arrival of a packet in the queue to the departure of the packet from the queue. The value is reported in milliseconds (ms) and is calculated for each SS as follows:

$$\hat{d} = \frac{\sum_{i=1}^N (f_i - a_i)}{N}$$

where, \hat{d} is the average queuing delay.

f_i is the time packet i leaves the queue.

a_i is the arrival time of packet i in the queue.

N is the number of packets.

Packet Loss (ρ): The percentage of packets dropped from the queue out of all the packets that arrived in the queue. The metric indicates the percentage of packets that missed their deadlines and is calculated as follows:

$$\rho = \frac{\sum_{i=1}^m \omega_i}{\sum_{j=1}^n \kappa_j}$$

where,

$\sum_{i=1}^m \omega_i$ is the sum of packets dropped.

$\sum_{j=1}^n \kappa_j$ is the sum of packets arrived in the queue.

Both average delay and packet loss will allow us to determine how effectively a scheduling algorithm satisfies the QoS requirements of real-time SSs.

Frame Utilization: The number of symbols utilized for data out all the symbols in the uplink sub-frame. The metric reported as a percentage is calculated as follows:

$$F_{\text{util}} = \frac{\sum_{i=1}^n \varpi_i}{C} \times 100\%$$

where, ϖ_i is the number of data symbols allocated to a SS

C is the total number of symbols in the uplink sub-frame.

n is the number of connections.

This metric will allow us to determine how effectively the scheduling algorithm utilizes the frame.

Fairness Index: Fairness is measured between users of the same traffic class (intra-class fairness) and as an overall measure, between all the users (inter-class fairness). We will use Jain's fairness index [39] to calculate inter-class fairness and Min-Max fairness index to calculate intra-class fairness. Due to the high similarity of legacy scheduling algorithms with respect to intra-class fairness, we use Min-Max index as it is more sensitive to service degradation and service unfairness [40]. The fairness indices are defined as follows:

$$\text{Jain's fairness index} = \frac{\left(\sum_{i=1}^n x_i \right)^2}{n * \sum_{i=1}^n (x_i)^2}$$

$$\text{Min-Max fairness index} = \frac{x_{\min}}{x_{\max}}$$

To calculate inter-class fairness using Jain's index, we use normalized average throughput for x_i . The average throughput of a SS is normalized with respect to the

MRTR of the SS i.e. $x = \frac{\hat{\tau}}{\text{MRTR}}$. Min-Max fairness calculates fairness between the SS

with the maximum average throughput (x_{\max}) in the class and the SS with the minimum average throughput (x_{\min}) in the class.

Honoured Requests: To evaluate the scheduling algorithms with respect to various bandwidth request mechanisms, we calculate the percentage of successful requests out of all the requests made by the SS. This metric will be referred to as percentage of honoured requests in the discussion.

4.3 Simulation Results

In this section, we study the performance of the scheduling algorithms under the context of IEEE 802.16-2004 MAC layer. More specifically, the effect of preamble symbols, frame length and bandwidth request mechanisms on the scheduling algorithms is studied. The experiments are executed under different network conditions such as light/heavy load and number of SSs in the network. Except for the bandwidth request analysis experiment, one slot per SS is reserved for bandwidth request and contention

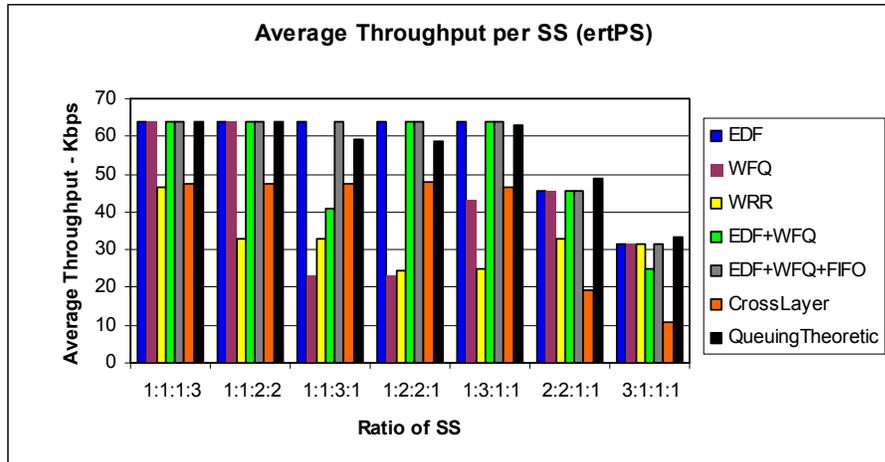
request opportunities. In all the experiments 95% confidence levels are maintained with 5% confidence intervals based on 10 independent runs [41].

4.3.1 The effect of SS ratio

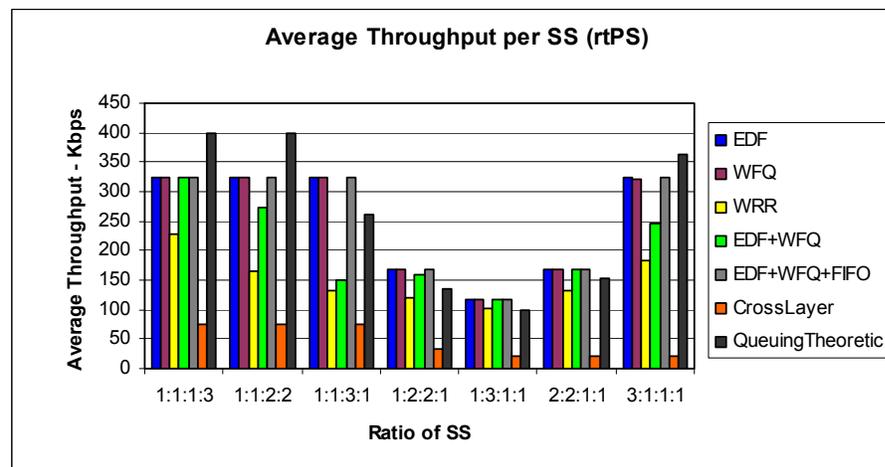
The objective of this experiment is to study the performance of the scheduling algorithms under different mixes of traffic. The experiment is conducted with a total traffic arrival rate of 4850 Kbps, of which 350Kbps is supplied by the ertPS class, 2200 Kbps is supplied by the rtPS class, 2200 Kbps is supplied by the nrtPS class and 100 Kbps is supplied by the BE class. We use a value of 15Kbps for the MRTR of ertPS SSs for this experiment. The traffic arrival rate is calculated based on the number of symbols available in the uplink sub-frame after subtracting the preamble symbols, bandwidth request and contention request symbols. Each SS consists of one connection. This is determined based on the simple CAC algorithm as described in chapter 3.

Table 4-6: The effect of SS Ratio – Parameters

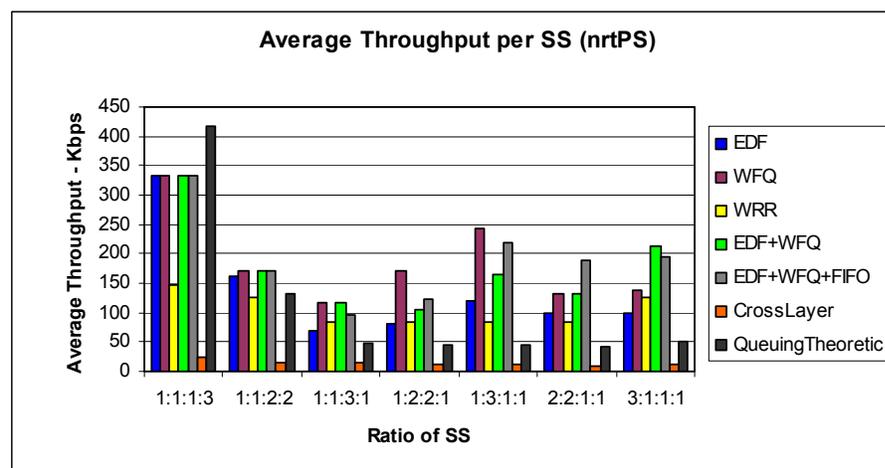
Parameter	Value
Number of SSs	36
Ratio of SS (ertPS:rtPS:nrtPS:BE)	1:1:1:3, 1:1:2:2, 1:1:3:1, 1:2:2:1, 1:3:1:1, 2:2:1:1, 3:1:1:1
Frame Length	20 ms



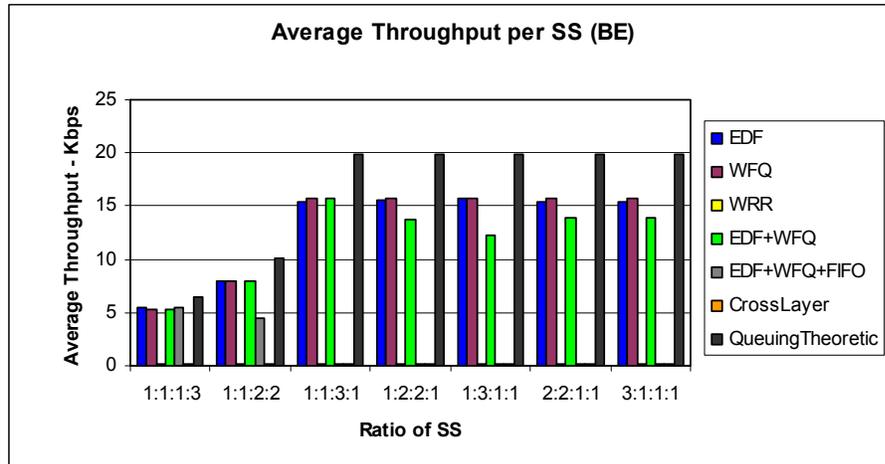
(a) Average Throughput - ertPS



(b) Average Throughput - rtPS



(c) Average Throughput - nrtPS



(d) Average Throughput - BE

Figure 4-1: The effect of SS Ratio – Average Throughput

In general, the average throughput of SSs of a class decreases with increased concentration of SSs of that class (see Figure 4-1). This is due to the total provisioning per class being constant and with more SSs, the load per SS decreases. The cross layer algorithm results in low average throughput of all the SSs. This behaviour is due to the algorithm selecting only one SS in a frame. The algorithm results in a higher average throughput of ertPS SSs than SSs of other classes. Due to the tight delay bound of ertPS SSs, they tend to get selected by the algorithm most of the time. This is an expected behaviour of the cross layer algorithm whereby the lower priority SSs will starve in the presence of large number of higher priority SSs [27]. The WRR algorithm indicates lower average throughput of SSs of the ertPS class compared to other legacy algorithms when the concentration of the SSs in the class is low. This is due to lower weight assigned to the SSs. The WFQ algorithm also assigns weights to the SSs, but since it allocates bandwidth according to the requirements, it results in higher average throughput. When the ratios of SSs are 1:1:3:1, 1:2:2:1 and 1:3:1:1, the residual capacity shared by active SSs is based on their weight, which is low. This results in both WRR and WFQ

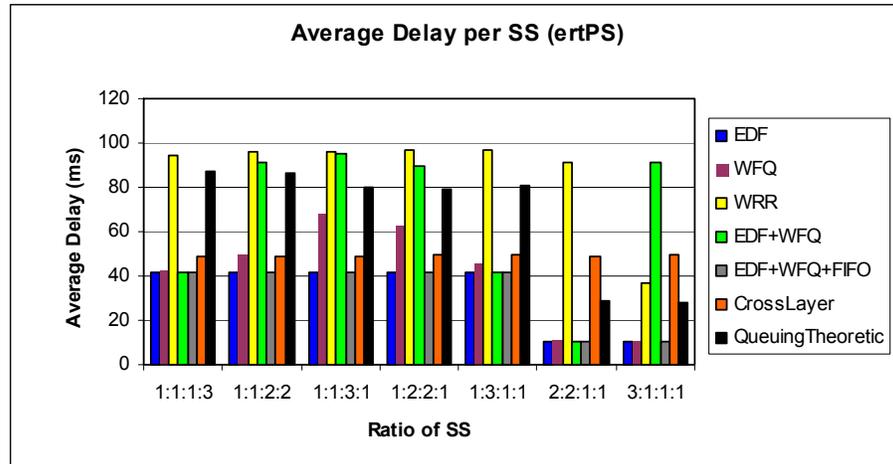
algorithms indicating low average throughput of ertPS SSs, since the difference between MRTR and MSTR is also large.

The WRR algorithm also indicates poor performance when the packet size of the traffic is large. This behaviour is indicated by the low average throughput of rtPS and nrtPS SSs, even under high concentration of rtPS and nrtPS SSs. When the concentration of SSs of the nrtPS class is the highest (1:1:3:1), the hybrid (EDF+WFQ) algorithm indicates lower average throughput for SSs of the ertPS class. Due to high concentration of nrtPS SSs, the overall bandwidth allocation mechanism will allocate a small amount of residual bandwidth to ertPS and rtPS classes. Both ertPS and rtPS SSs will compete for the small amount of bandwidth. Since the EDF algorithm schedules SSs based on their delay requirements only, the average throughput will be low. This will also reflect a low average throughput of rtPS SSs when the ratio is 1:1:3:1. As well, due to the high MSTR of nrtPS SSs, they will consume more bandwidth before reaching the limit (MSTR).

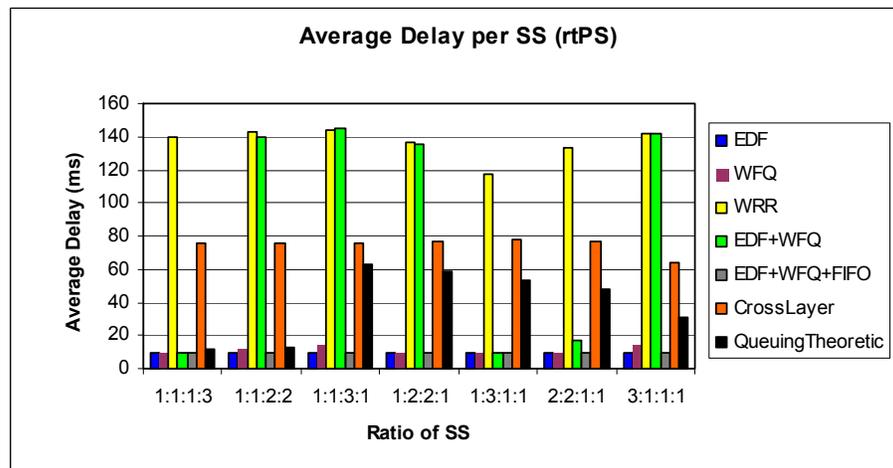
When the concentration of nrtPS SSs is high (1:1:3:1, 1:2:2:1), the Queuing Theoretic algorithm indicates lower average throughput for the ertPS and rtPS classes. This behaviour is due to a lower residual bandwidth assigned to the ertPS and rtPS classes. The average throughput of nrtPS SSs under the Queuing Theoretic algorithm is low when the ratio is 1:1:3:1 because of lower load per nrtPS SSs and higher average delay of ertPS and rtPS SSs that results in a lower utility assigned to them. The Queuing Theoretic algorithm also indicates a higher average throughput for the rtPS class when the concentration of BE and ertPS SSs is the highest (1:1:1:3 and 3:1:1:1). This behaviour is due to the lower MRTR of BE and ertPS SSs that results in a higher threshold assigned to the rtPS class. The EDF algorithm indicates a lower average throughput for the nrtPS

class when the concentration of ertPS or rtPS SSs is the highest since the algorithm provides strict priority to SSs with delay requirements (ertPS and rtPS SSs). Although the hybrid (EDF+WFQ+FIFO) algorithm also provides strict priority to ertPS and rtPS SSs, it results in a higher average throughput for the nrtPS class than the EDF algorithm. This behaviour is due to the hybrid (EDF+WFQ+FIFO) algorithm providing strict priority to nrtPS SSs over BE SSs. On the other hand, under the EDF algorithm, both nrtPS and BE SSs will compete for bandwidth once ertPS and rtPS SSs don't have any data to send.

All the scheduling algorithms, except for hybrid (EDF+WFQ+FIFO), WRR and cross layer algorithms, satisfy the MRTR (1Kbps) of SSs of the BE class. The WRR scheduling algorithm starves the SSs of the BE class due to the large number of SSs of the other three classes (ertPS, rtPS and nrtPS) and the large packet size of BE traffic (100 bytes). The small amount of bandwidth that is allocated to the SSs of the BE class is not enough to transmit one packet and it is therefore wasted. The hybrid (EDF+WFQ+FIFO) algorithm also results in starvation of SSs of the BE class due to the strict priority nature of the algorithm. The Queuing Theoretic algorithm indicates the highest average throughput for BE SSs since the minimum bandwidth allocated to BE SSs is equivalent to one packet size (100 bytes). Due to the large packet size of BE traffic, their average throughput is higher than that provided by the other algorithms.

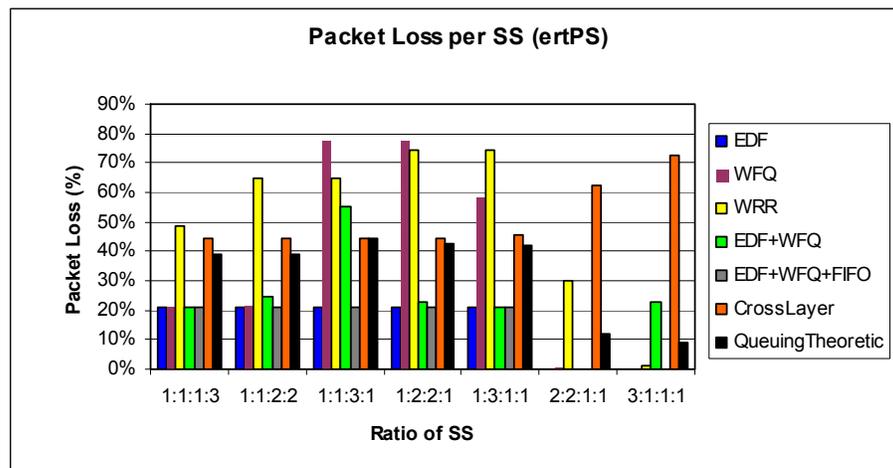


(a) Average Delay - ertPS

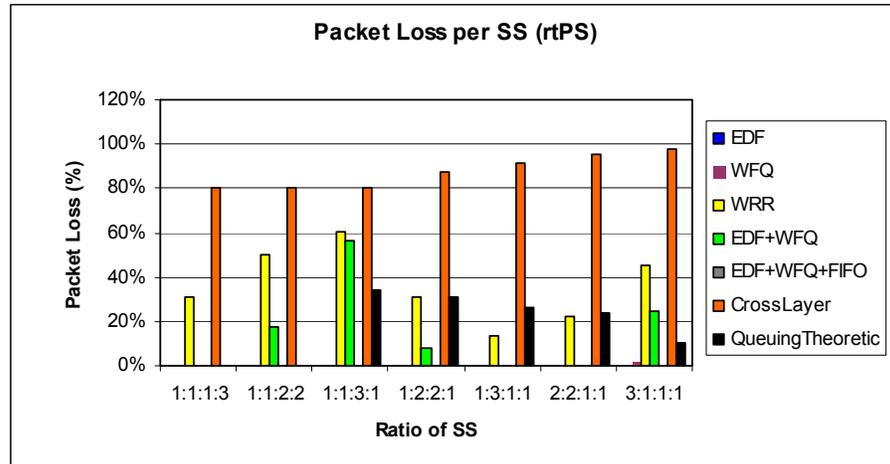


(b) Average Delay - rtPS

Figure 4-2: The effect of SS Ratio – Average delay



(a) Packet Loss - ertPS



(b) Packet Loss - rtPS

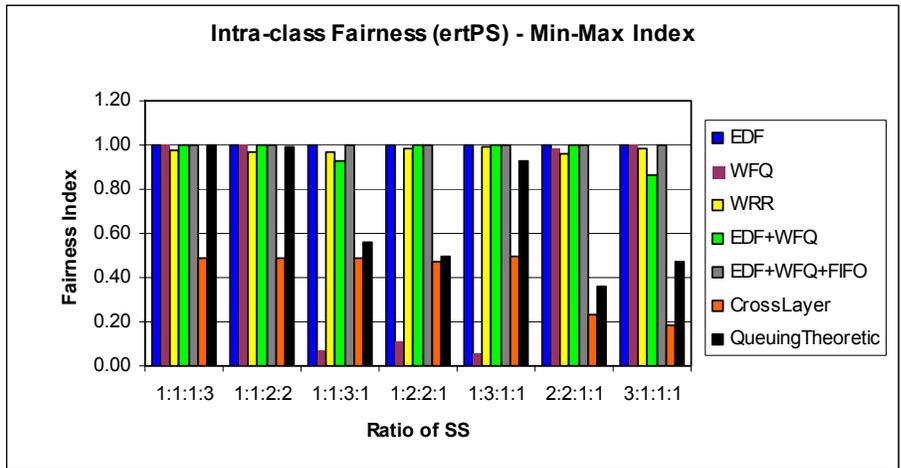
Figure 4-3: The effect of SS Ratio - Packet loss

The WRR algorithm indicates very high average delay for the ertPS class except when the concentration of ertPS SSs is the highest (see Figure 4-2). With a high concentration of ertPS SSs, the weight assigned to them is high resulting in more bandwidth allocated to the SSs. The Queuing Theoretic algorithm indicates a high average delay for the ertPS SSs when their concentration is low. This behaviour is due to a low threshold assigned for the ertPS class. The EDF and WFQ algorithms indicate similar average delay for ertPS SSs except when the concentration of nrtPS SSs is high. WFQ schedules SSs by ensuring they receive at least bandwidth equivalent to their MRTR. Due to the large number of nrtPS SSs, the bandwidth allocated to ertPS SSs will only be enough to satisfy its MRTR, but not enough to keep the average delay low. The hybrid (EDF+WFQ) algorithm indicates a high average delay for the ertPS class when the ratios are 1:1:2:2, 1:1:3:1 and 1:2:2:1. This behaviour is due to the overall bandwidth allocation mechanism of the algorithm that allocates a small amount of bandwidth for the ertPS and rtPS classes. The algorithm indicates a high average delay for ertPS SSs even when their concentration is high as a result of a larger number of ertPS SSs sharing the

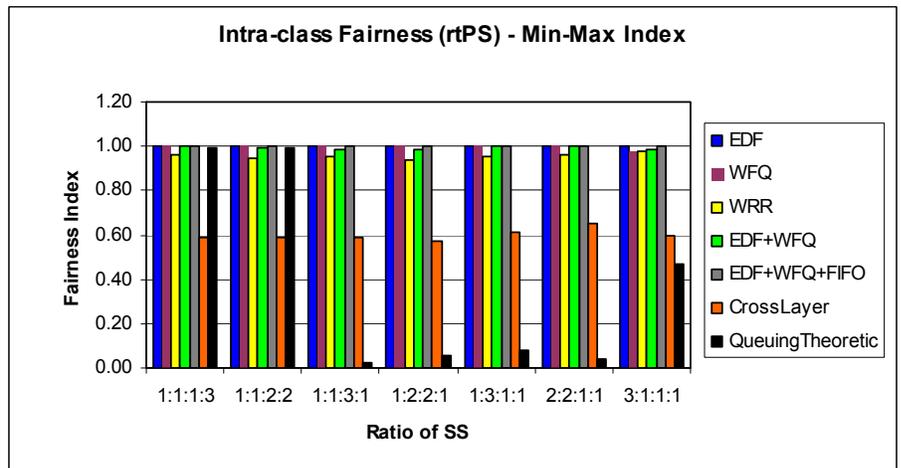
bandwidth with the rtPS SSs. Due to the high load of rtPS SSs, a large number of video packets will arrive before a VoIP packet arrives. This will result, in numerous cases, in a lower deadline assigned to video packets, even though they have a higher maximum latency.

The average delay of rtPS SSs, in most cases, under the EDF, WFQ and hybrid (EDF+WFQ+FIFO) algorithms, is approximately 10ms. The value of 10ms corresponds to the allocation start time, which is the time the allocation according to the UL-MAP message will start. Since the EDF and hybrid (EDF+WFQ+FIFO) algorithms provide high priority to rtPS SSs, all the data of rtPS SSs will be flushed out in a frame. Due to the high MRTR of rtPS SSs (64 Kbps), the weight assigned to them by the WFQ algorithm will also be high. The WRR algorithm indicates a very high average delay of rtPS SSs mainly due to the large packet size of video traffic (150-300 bytes). If the symbols assigned to the rtPS SSs are not enough to transmit a packet due to its large size, the packet will spend a longer time in the queue that will increase the average delay and even packet loss. The Queuing Theoretic algorithm results in a high average delay of the rtPS class when the ratios are 1:1:3:1, 1:2:2:1, 1:3:1:1 and 2:2:1:1. In the algorithm, all the SSs are allocated their MRTR initially. The residual bandwidth is allocated by using a water filling mechanism by allocating bandwidth equivalent to a PDU size to the SS with the lowest utility. Therefore, some of the ertPS and rtPS SSs will not be allocated bandwidth more than their MRTR due to competition from other ertPS and rtPS SSs. This will result in packets spending a longer time in the queue, thus increasing the average delay. The increase in average delay of SSs results in an increase in packet loss, although the relationship between average delay and packet loss is not as explicit i.e.

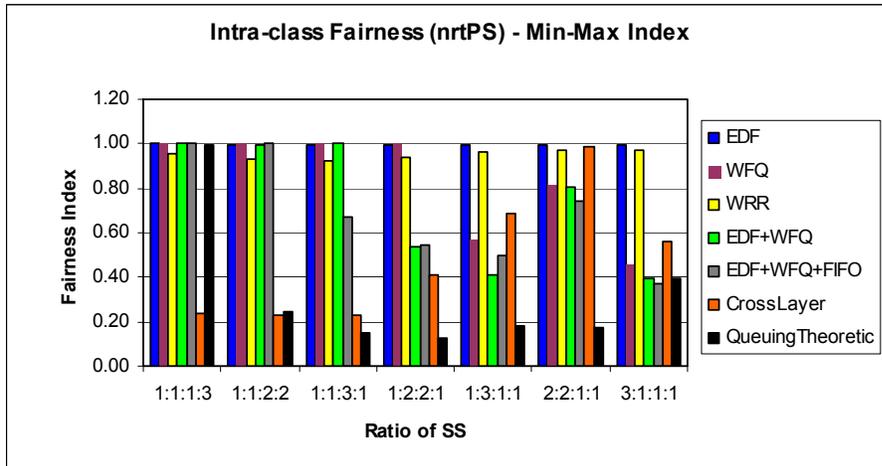
when a large number of packets are dropped from the queue it may result in a decrease in the average delay (see Figure 4-3). This is because the average delay does not include the delay of dropped packets. For instance, when the ratio is 1:3:1:1, the WFQ algorithm has similar average delay as the EDF algorithm, but results in higher packet loss.



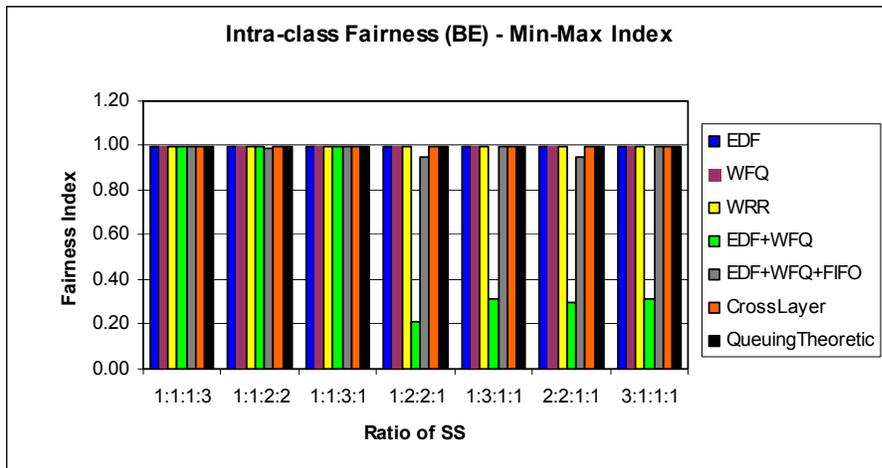
(a) Intra-class fairness - ertPS



(b) Intra-class fairness - rtPS



(c) Intra-class fairness - nrtPS



(d) Intra-class fairness - BE

Figure 4-4: The effect of SS Ratio: Intra-class Fairness – Min-max Index

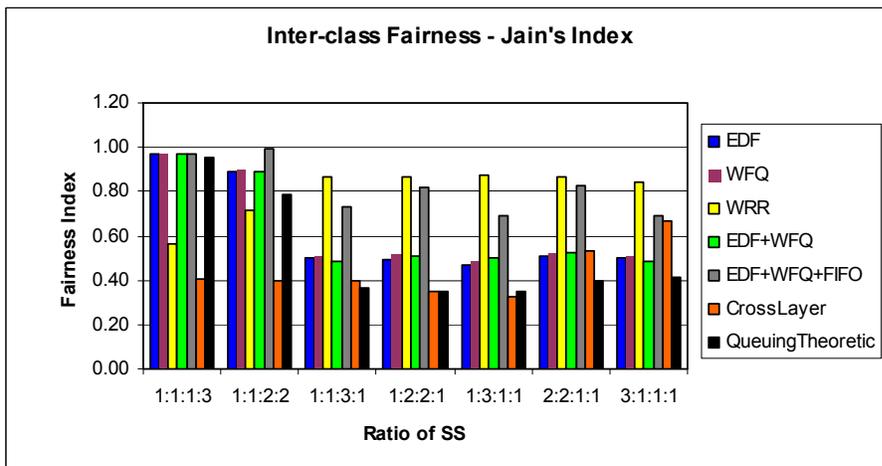


Figure 4-5: The effect of SS Ratio: Inter-class Fairness – Jain's Index

The Min-max index will be used to calculate intra-class fairness due to its high sensitivity towards service unfairness and degradation, since we assume that SSs from the same class should be treated equally. Thus, finding the ratio of the minimum average throughput to the maximum average throughput is an informative indicator of unfairness. Results of intra-class fairness based on Jain's index are provided in Appendix A. When the concentration of SSs of the nrtPS class is high (1:1:3:1 and 1:2:2:1), the fairness among SSs of the ertPS class under the WFQ algorithm is the lowest (see Figure 4-4). WFQ allocates bandwidth to the SSs based on their MRTR. However, any additional available bandwidth is shared among active SSs based on their weights. The weight of the ertPS SSs is low due to their low concentration and low MRTR. Thus, based on the active SSs, packet finish times and residual bandwidth, some SSs may be allocated bandwidth close to their MSTR while others will be allocated bandwidth close to their MRTR. A similar behaviour is noticed among SSs under the Queuing Theoretic algorithm. However, the difference between the minimum and maximum average throughput is smaller due to the fact that the algorithm allocates the residual capacity on a unit of PDU in a round robin fashion. The low fairness can also be attributed to large finish times assigned to the VoIP packets due to the nature of VoIP traffic. This deficiency of the WFQ algorithm has also been discussed in [42] and is further emphasized when studying the effect of uplink burst preamble.

The cross layer algorithm indicates relatively low intra-class fairness for all the SSs (except BE SSs). This behaviour is due to the algorithm allocating bandwidth to only one SS in a frame, resulting in a large difference between the minimum and maximum average throughput in the class. The intra-class fairness of the BE class is high since all

the BE SSs get allocated very little bandwidth. This will result in a small difference between the minimum and maximum average throughput in the class. The Queuing Theoretic algorithm indicates low intra-class fairness among SSs of all the classes (except the BE class) when the concentration of nrtPS SSs is high (ratios 1:1:3:1 and 1:2:2:1). The intra-class fairness of ertPS and rtPS classes is low because of the low residual bandwidth and the allocation mechanism as discussed previously. The intra-class fairness of the nrtPS class is low due to the high average delay of ertPS and rtPS SSs. The high average delay of ertPS and rtPS SSs results in a lower utility assigned to them compared to the utility assigned to nrtPS SSs.

When the ratios are 2:2:1:1 and 3:1:1:1, the intra-class fairness for the ertPS class is low. This behaviour is due to the large number of ertPS SSs competing for bandwidth, resulting in a larger difference in minimum and maximum average throughput. When the concentration of BE SSs is low, the intra-class fairness of the nrtPS class is low under the hybrid (EDF+WFQ+FIFO) algorithm. This behaviour is due to the strict priority of ertPS and rtPS SSs and that a larger number of nrtPS SSs compete for a small amount of residual bandwidth. The hybrid (EDF+WFQ) algorithm indicates a low intra-class fairness for the BE class when the concentration of nrtPS or BE class is low. This behavior is due to the competition of BE and nrtPS SS for the small amount of bandwidth allocated to them as a result of the overall bandwidth allocation mechanism of the algorithm.

The EDF algorithm shows a higher inter-class fairness when the concentration of SSs of the BE class is high but the fairness decreases with increased concentration of SSs of the ertPS, rtPS and nrtPS classes (see Figure 4-5). With large number of SSs of the

nrtPS class, the ertPS and rtPS SSs will be allocated large amount of bandwidth whereas the nrtPS SSs will compete with BE SSs for bandwidth. This will result in large difference in bandwidth allocated between the different classes. When the ratios are 1:3:1:1, 2:2:1:1 and 3:1:1:1, the EDF algorithm shows low inter-class fairness because of the large number of ertPS and rtPS SSs receiving majority of the bandwidth. The hybrid (EDF+WFQ+FIFO) algorithm indicates a high inter-class fairness even when the concentration of ertPS and rtPS SSs is high. This behaviour of the algorithm is because it provides strict priority to nrtPS SSs over BE SSs. Since the inter-class fairness is calculated using average throughput normalized with respect to MRTR of the SSs, the WRR algorithm will indicate high inter-class fairness. This is because the WRR algorithm assigns weight to the SSs according to their MRTR and serves all the classes in rounds. The Queuing Theoretic algorithm indicates low inter-class fairness except when the concentration of BE SSs is high. This is because the utility function of the Queuing Theoretic algorithm is based on the average delay for rtPS and ertPS SSs and the average throughput for nrtPS SSs. We calculate the inter-class fairness based on the average throughput. However, the utility function for rtPS SSs does not take the average throughput into consideration which is an important QoS parameter for the rtPS class, but not so important for the ertPS class. Therefore, the unfairness of rtPS SSs due to the average throughput is not detected by the utility function, although the purpose of the utility function is to provide fairness.

4.3.2 The effect of uplink burst preamble

According to the IEEE 802.16-2004 standard [1], each uplink burst is associated with a preamble whose length is defined in the Uplink Channel Descriptor (UCD)

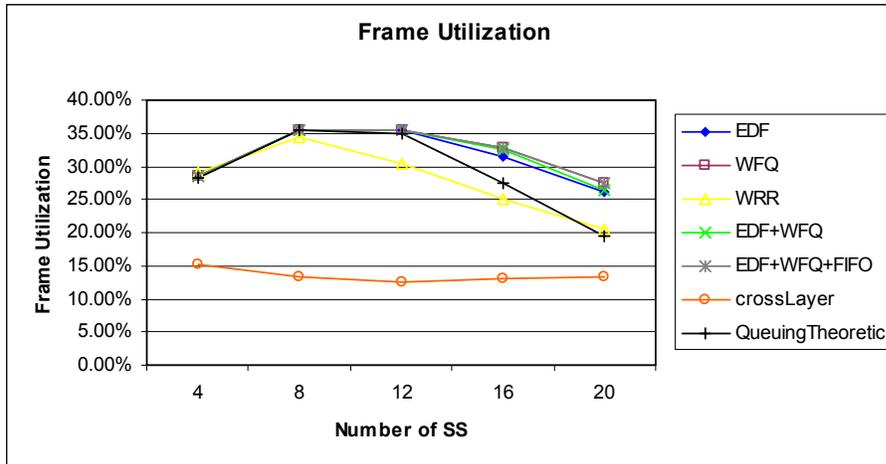
message. The length of the preamble can be either 16 symbols or 32 symbols. In our experiments, we use 16 symbols for uplink preamble. The purpose of the uplink burst preamble is to allow the BS to synchronize to each SS as to when they will begin to transmit a burst of data. This can represent a significant overhead when the number of SS is large, as we will observe from the results of this experiment. We will study the effect of the preamble by increasing the number of SSs.

4.3.2.1 Effect of uplink burst preamble on frame utilization

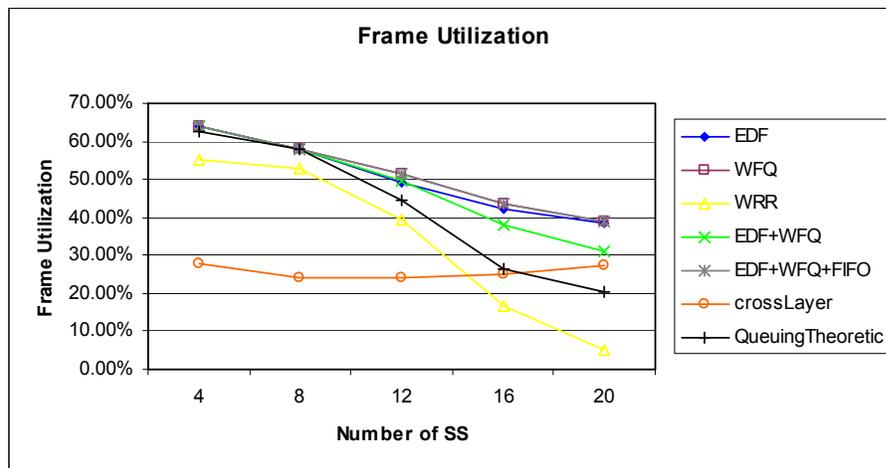
In this experiment, we will study the behavior of the algorithms with respect to frame utilization when the number of SSs is increased. To be able to distinguish between the algorithms, a large amount of traffic was generated. Since VoIP and BE traffic have relatively low rates, we used Video (rtPS) and FTP (nrtPS) traffic only. The experiment was conducted under light load of 4Mbps and a heavy load of 8Mbps. Table 4-7 lists the parameters used for the experiment.

Table 4-7: The effect of uplink burst preamble – Parameters (frame utilization)

Parameter	Value
Number of SSs	4-20
Ratio of SS (rtPS:nrtPS)	1:1
Frame Length	10 ms



(a) Frame utilization under light load



(b) Frame utilization under heavy load

Figure 4-6: The effect of uplink burst preamble – Frame Utilization

Under light load, as the number of SSs increases, increase in frame utilization is indicated by all the algorithms except the cross layer algorithm (see Figure 4-6). This behavior is observed until the number of SSs is eight. With more than twelve SSs, the frame utilization starts to decrease due to the increased overhead of the uplink burst preamble. The decline in the frame utilization is more severe for the Queuing Theoretic and WRR algorithms as these algorithms tend to select the maximum number of SSs in a frame, thus resulting in maximum overhead.

The cross layer algorithm results in the lowest frame utilization regardless of the load. This is because the algorithm selects only one SS per frame. Under heavy load, the cross layer algorithm still provides a stable, although higher (~28%) frame utilization, compared to that under light load (~14%).

As the number of SSs increase, the frame utilization decreases due to the increased overhead of uplink burst preamble. When the number of SSs is large (greater than 16), the cross layer algorithm indicates a higher frame utilization than WRR and Queuing Theoretic algorithms. Due to the heavy load, the single user selected by the cross layer algorithm occupies a significant portion of the frame whereas in the WRR and Queuing Theoretic algorithms a large portion of the frame is wasted by uplink burst preambles. The hybrid (EDF+WFQ) algorithm indicates lower frame utilization than other legacy algorithms for a large number of SSs. This is because the algorithm allocates bandwidth to each traffic class according to the MRTR of the SSs in that class. The bandwidth assigned to a class can remain unused if SSs of that class don't have any data to send, resulting in lower frame utilization.

4.3.2.2 The effect of uplink burst preamble on user performance and fairness

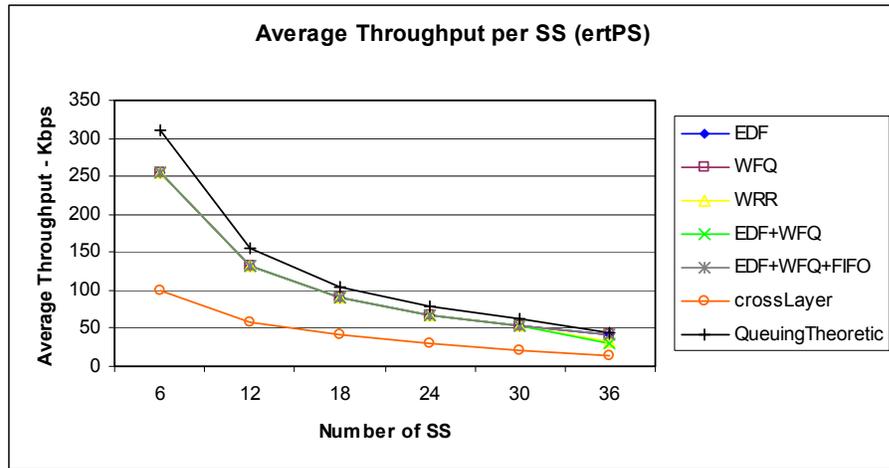
In this experiment, we will study the effect of uplink burst preamble on the scheduling algorithms with respect to average throughput, average delay, packet loss and fairness. We will study the performance of the algorithms both under light and a heavy load. We use MRTR of 10Kbps for ertPS SSs as it allows us to choose appropriate values for the parameters. A light load of 3400 Kbps is supplied with 500 Kbps reserved for ertPS class, 1500 Kbps reserved for rtPS class, 1250 Kbps reserved for nrtPS class and 150 Kbps reserved for BE class. A heavy load of 6800 Kbps is supplied with 1000 Kbps

reserved for ertPS class, 3000 Kbps reserved for rtPS class, 2500 Kbps reserved for nrtPS class and 300 Kbps reserved for BE class. The remaining parameters for the experiment are listed in Table 4-8.

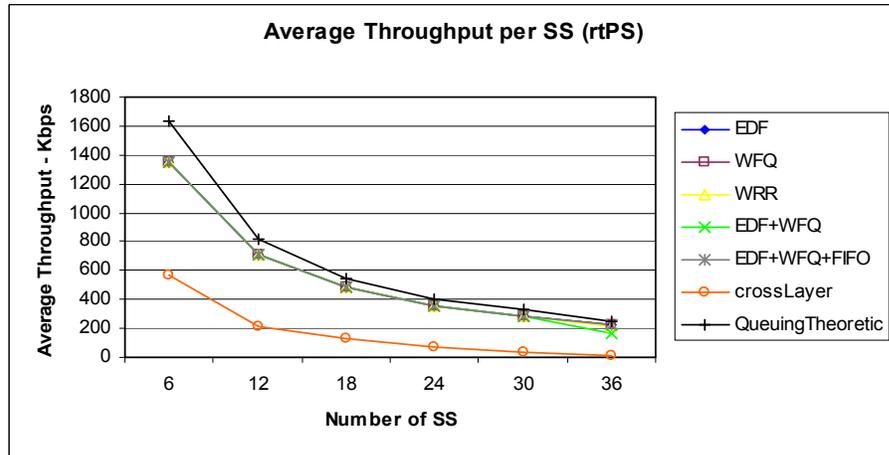
Table 4-8: The effect of uplink burst preamble – Parameters

Parameter	Value
Number of SSs	6-36
Ratio of SS (ertPS:rtPS:nrtPS:BE)	3:1:1:1
Frame Length	20 ms

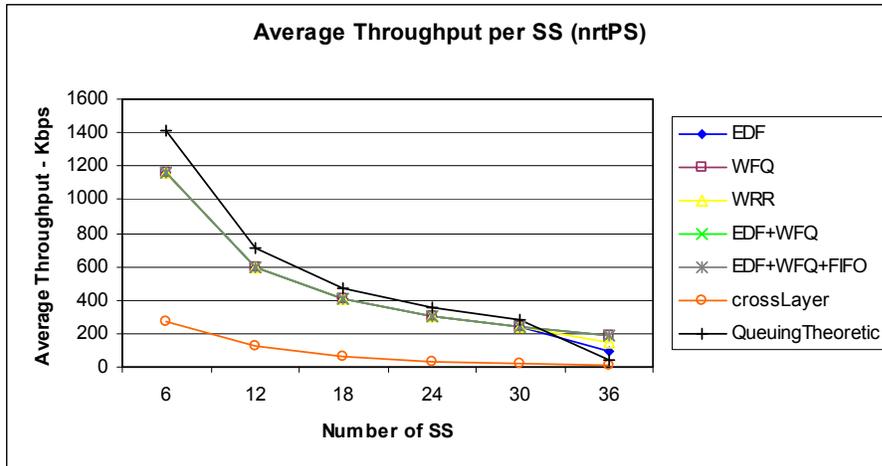
Light load:



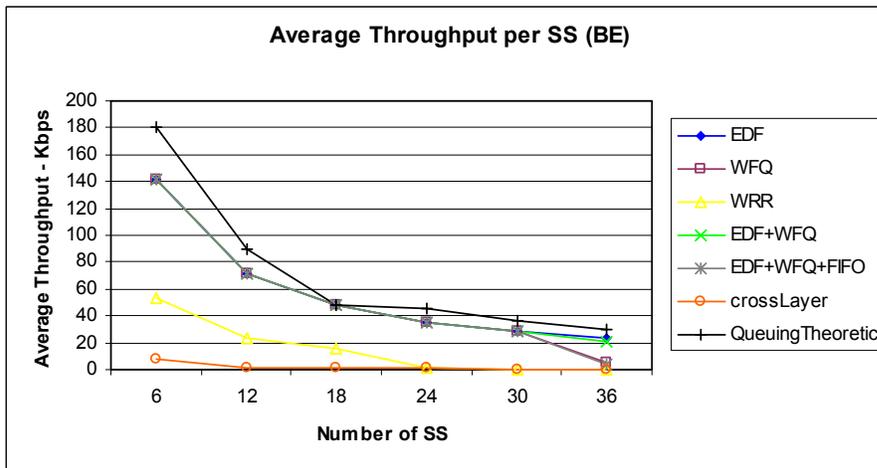
(a) Average Throughput - ertPS



(b) Average Throughput - rtPS

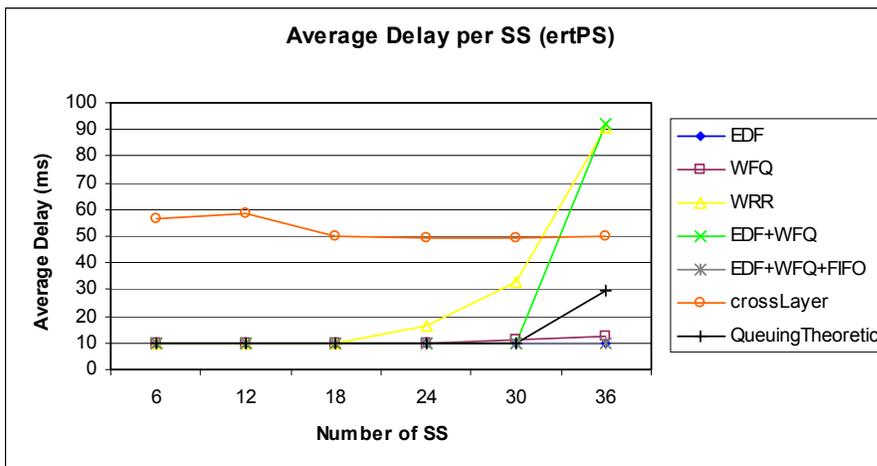


(c) Average Throughput - nrtPS

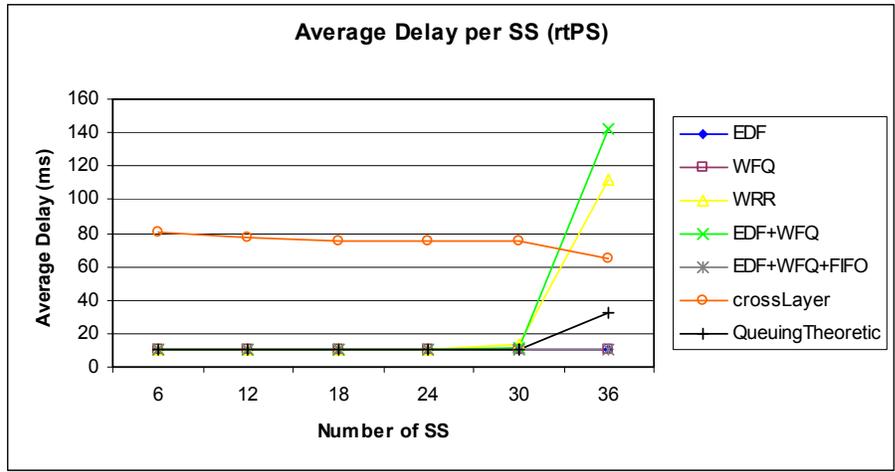


(d) Average Throughput - BE

Figure 4-7: The effect of uplink burst preamble – Average throughput (Light load)

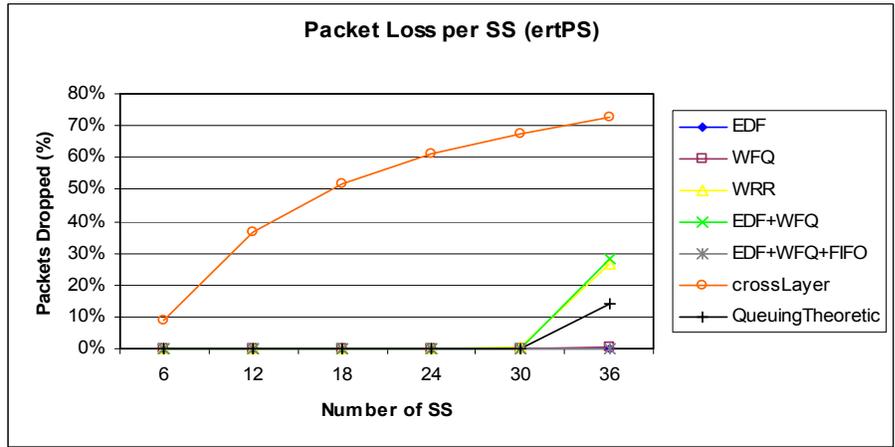


(a) Average Delay - ertPS

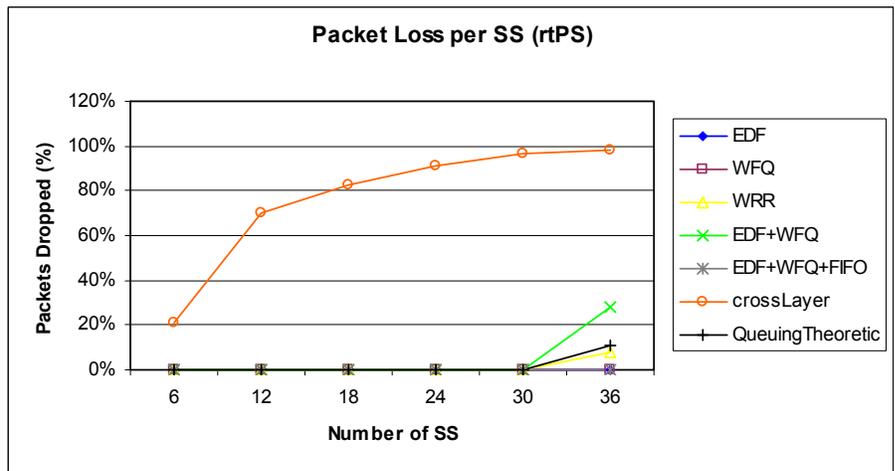


(b) Average Delay - rtPS

Figure 4-8: The effect of uplink burst preamble – Average delay (light load)



(a) Packet Loss - ertPS

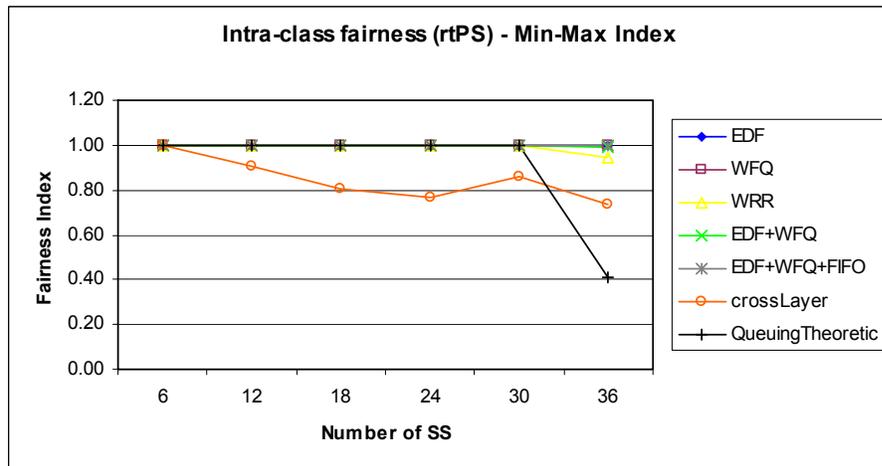


(b) Packet Loss - rtPS

Figure 4-9: The effect of uplink burst preamble – Packet loss (light load)



(a) Intra-class fairness - ertPS



(b) Intra-class fairness - rtPS



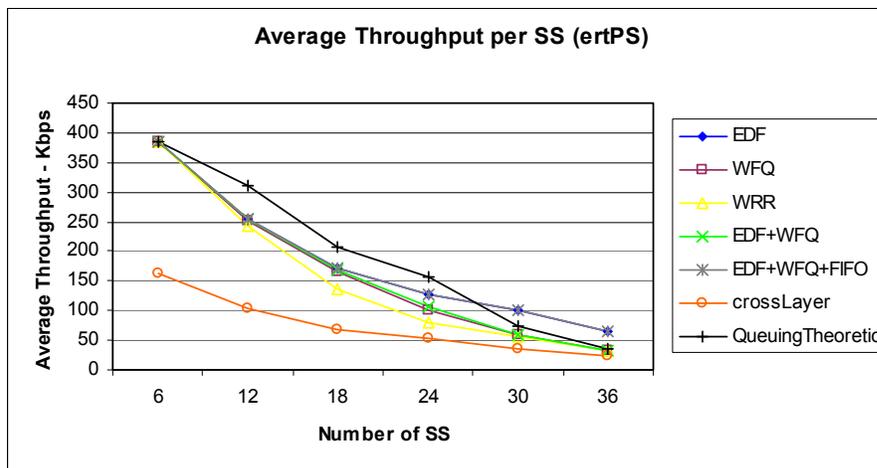
(c) Intra-class fairness - nrtPS



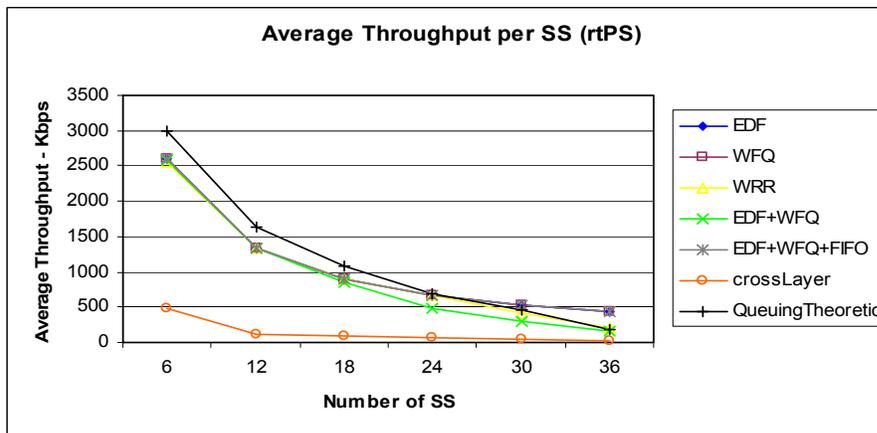
(d) Intra-class fairness - BE

Figure 4-10: The effect of uplink burst preamble: Intra-class fairness (Light load)

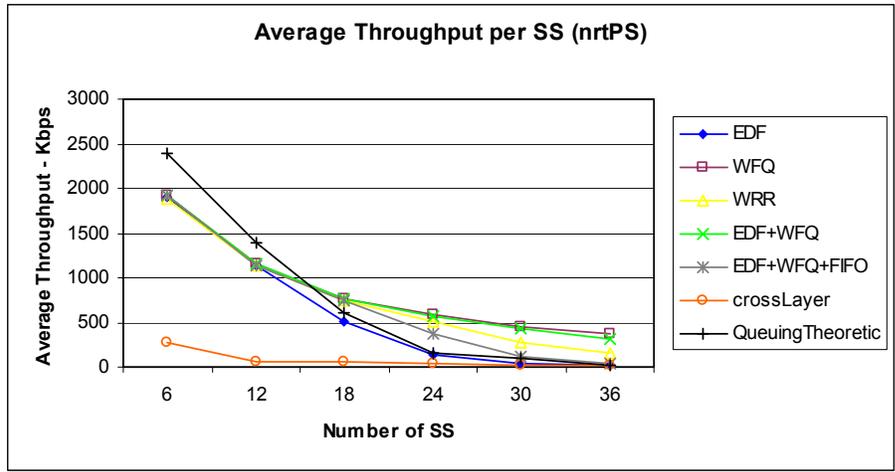
Heavy load:



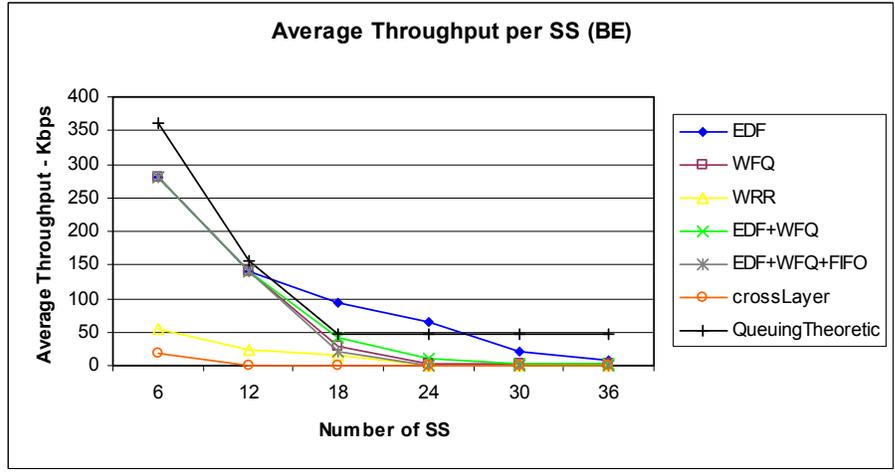
(a) Average Throughput - ertPS



(b) Average Throughput - rtPS

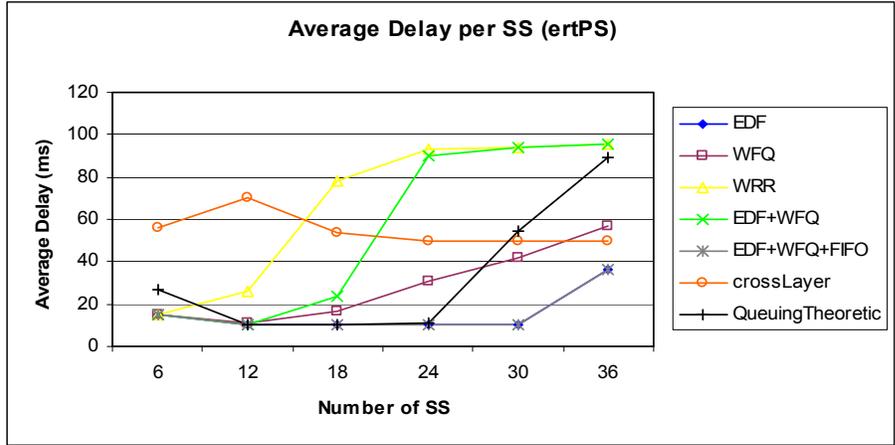


(c) Average Throughput - nrtPS

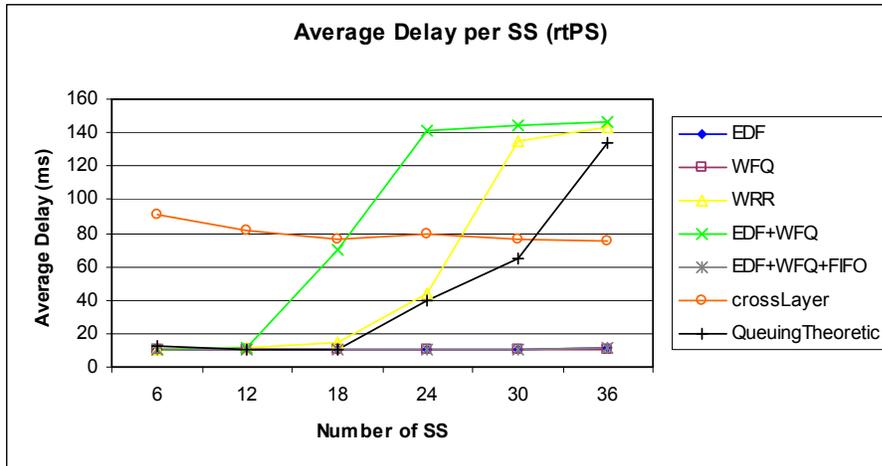


(d) Average Throughput - BE

Figure 4-11: The effect of uplink burst preamble: Average throughput (Heavy load)

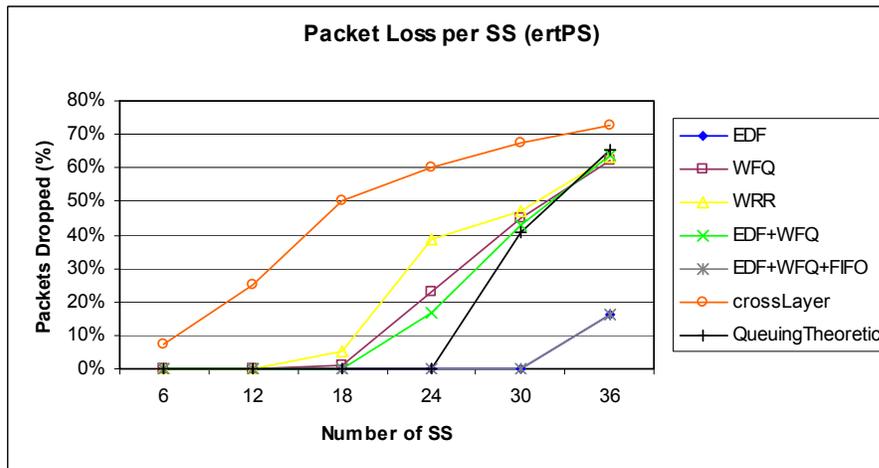


(a) Average Delay - ertPS

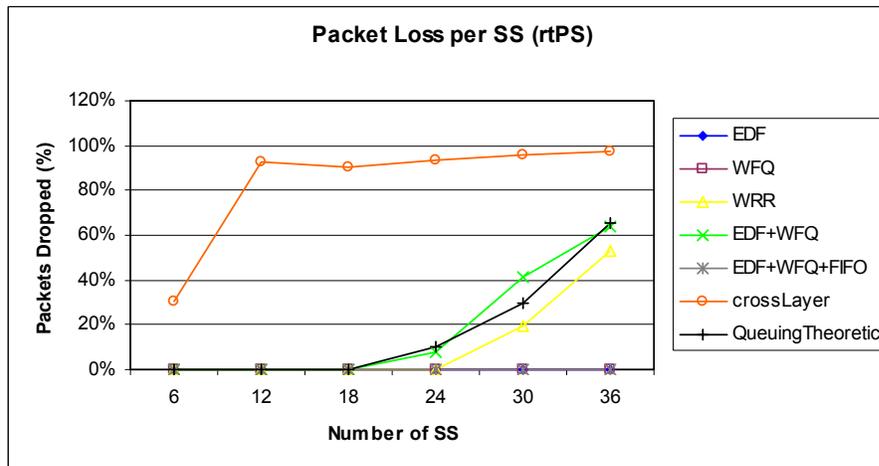


(b) Average Delay – rtPS

Figure 4-12: The effect of uplink burst preamble: Average delay (Heavy load)

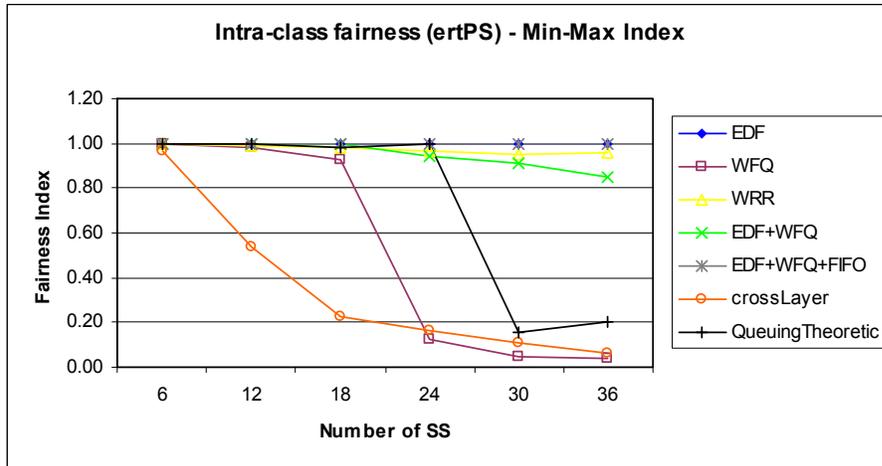


(a) Packet Loss - ertPS



(b) Packet Loss - rtPS

Figure 4-13: The effect of uplink burst preamble: Packet loss (Heavy load)



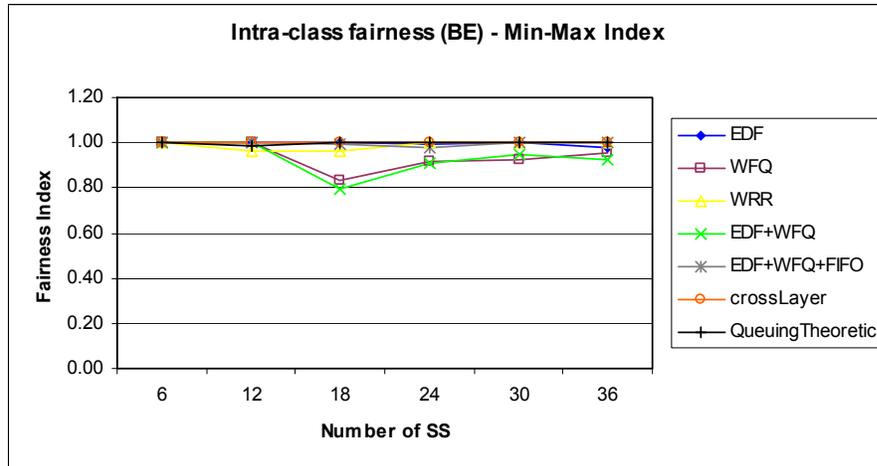
(a) Intra-class fairness - ertPS



(b) Intra-class fairness - rtPS



(c) Intra-class fairness - nrtPS



(d) Intra-class fairness - BE

Figure 4-14: The effect of uplink burst preamble: Intra-class fairness (Heavy load)

The average throughput, under both light and heavy loads, decreases with an increasing number of SSs due to decreasing load per SS and increase in bandwidth wasted by uplink burst preambles. The Queuing Theoretic algorithm, for fewer SSs, shows a higher average throughput for SSs of all traffic classes since it allocates at least the MRTR in every frame for them (see Figure 4-7, Figure 4-11). The algorithm indicates the highest average throughput for the ertPS class. However, under the heavy load and with a large number of SSs (greater than 24), the Queuing Theoretic algorithm indicates lower average throughput than the hybrid (EDF+WFQ+FIFO) algorithm. This behaviour is due to the large overhead incurred by the Queuing Theoretic algorithm as it selects the maximum number of SSs in a frame. SSs of the BE class get assigned bandwidth equivalent to the size of one packet (100 bytes). Therefore, due to the large packet size, the average throughput of BE SSs, under the heavy load and large number of SSs (greater than eighteen), stays at 50 Kbps. The average throughput of SSs of the nrtPS class is low with a large number of SSs and under heavier load (see Figure 4-11). This is because the minimum allocated bandwidth per frame by the Queuing Theoretic algorithm is smaller

than the packet size of FTP traffic. With a packet size of 150 bytes and MRTR of 45 Kbps, at least 112.5 bytes every frame is allocated for each nrtPS SS. This minimum allocation is insufficient to transmit one packet and is thus wasted. With large number of SSs, the ertPS and rtPS SSs will be assigned higher priority by the Queuing Theoretic algorithm due to their high average delay. This will result in very few transmission opportunities for the nrtPS SSs and therefore their MRTR will not be satisfied. Under lighter load, SSs of the nrtPS class get allocated more bandwidth that is large enough to transmit at least one packet (see Figure 4-7). Due to the light load, the nrtPS SSs will get more transmission opportunities resulting in their average throughput being higher than the MRTR. The cross layer algorithm indicates the lowest average throughput for the SSs out of all the algorithms as it selects only one SS in a frame. When the number of SSs is high, the lower priority SSs (nrtPS and BE classes) are allocated very little bandwidth, indicated by a very low average throughput. This is an expected behaviour of the cross layer scheme as discussed in [27]. The difference in average throughput between the legacy algorithms is more noticeable under the heavy load scenario. The EDF and hybrid (EDF+WFQ+FIFO) algorithms indicate a higher average throughput for SSs of ertPS and rtPS classes due to their strict priority nature towards real-time SSs. The WFQ algorithm indicates almost identical average throughput for SSs of the rtPS class when compared with EDF and hybrid (EDF+WFQ+FIFO) algorithms. This behaviour is due to the high MRTR of SSs of the rtPS class, allowing the WFQ algorithm to allocate a large amount of bandwidth for them. The WRR algorithm results in very low average throughput for the BE class due to the low MRTR of SSs of the class and the large packet size of the BE traffic (100 bytes).

The average delay and packet loss increase with increasing number of SSs due to increasing overhead of uplink burst preamble and increasing number of SSs (see Figure 4-8, Figure 4-9, Figure 4-12 and Figure 4-13). The cross layer algorithm shows no significant difference in average delay under both light and heavy loads (see Figure 4-8 and Figure 4-12). Since the cross layer algorithm selects only one SS in a frame, it will result in a large backlog of data. The backlogged packets will miss their deadline and will therefore be dropped. This behaviour is reflected by the increase in packet loss with increasing number of SSs (see Figure 4-9 and Figure 4-13). The WRR and hybrid (EDF+WFQ) algorithms indicate the highest average delay with a large number of SSs (greater than 18 for heavy load and greater than 30 for light load). Since these algorithms partition bandwidth according to the MRTR and number of SSs, and their nature of selecting large number of SSs every frame, the amount of bandwidth allocated for SSs of ertPS and rtPS classes will be less. Under the hybrid (EDF+WFQ) algorithm, both ertPS and rtPS SSs will compete for the bandwidth. This will result in an increase in average delay with increasing number of SSs. The lowest average delay under the light load, as indicated by the EDF and hybrid (EDF+WFQ+FIFO) algorithms is 10ms, which is the value of the allocation start time. The minimum amount of time a packet will spend in the queue is the start of the uplink allocation according to the UL-MAP message, which is the value of allocation start time.

Under light load, the Queuing Theoretic algorithm results in an increase in average delay with maximum number of SSs (see Figure 4-8). This behaviour is due to more SSs competing for the same amount of bandwidth in a class. The increased average delay is also due to the large overhead of the uplink burst preamble. The increase in

overhead will result in a reduction of allocated bandwidth of each class. The increase in average delay is more severe under the heavy load (see Figure 4-12). Although the Queuing Theoretic, WRR and hybrid (EDF+WFQ) algorithms indicate higher average delay than the cross layer algorithm, they result in a lower packet loss (see Figures 4-9 and 4-13).

The fairness among SSs of the ertPS class (see Figure 4-10 and Figure 4-14) decreases with an increasing number of SSs under the cross layer, WFQ and Queuing Theoretic algorithms. With an increased number of SSs, the cross layer algorithm will give the highest priority to SSs of the ertPS class. Since only one SS is selected by the cross layer algorithm, the difference in minimum and maximum average throughput in the class will be high. Under the Queuing Theoretic algorithm, since a larger number of SSs compete for the same amount of bandwidth (low provisioned bandwidth per class due to large overhead), some SSs will receive a large portion of the bandwidth (more than their MRTR) and some will not. This is the same reason for the decrease in fairness among SSs of the rtPS class. The WFQ algorithm indicates a low fairness among ertPS SSs because of the bursty nature of VoIP traffic. Evaluation of time-stamp based schedulers in [42] shows that WFQ indicates a low fairness among users when the traffic is bursty, such as the VoIP traffic used for ertPS class in our model.

Under the Queuing Theoretic algorithm, some SSs will receive a large portion of bandwidth (more than their MRTR) and some will not. This is due to the utility function of rtPS SSs that does not take the average throughput into account, therefore resulting in fluctuation of intra-class fairness. The intra-class fairness of the rtPS class under the

Cross layer algorithm depends on the amount of traffic transmitted by the selected SS e.g. when the number of SSs is 12 and 18.

The intra-class fairness for the nrtPS class, as indicated by the cross layer algorithm, dips when the number of SSs is 12. This behaviour is due to the cross layer algorithm selecting only one SS in a frame and with only two nrtPS SSs it indicates a large difference between the minimum and maximum average throughput in the class. Under heavy load, the intra-class fairness for the nrtPS class fluctuates in the case of the Queuing Theoretic algorithm when the number of SSs is greater than eighteen (see Figure 4-14). For a small number of nrtPS SSs, a large amount of residual bandwidth remains (after allocating the MRTR to each SS). Consequently, the difference between the minimum and maximum average throughput among the SSs is more perceptible, which is translated as lower fairness. When the number of SSs increases to 24, this difference in throughput decreases resulting in higher fairness. Although the residual bandwidth is less, it is enough to satisfy most nrtPS SSs, thus resulting in the decrease in difference between the minimum and maximum average throughput. When number of SSs is greater than twenty four, the fairness decreases because more nrtPS SSs compete for the smaller amount of residual bandwidth which is not sufficient to satisfy them all. The intra-class fairness for the BE class is very high due to very little bandwidth allocated to these SSs, resulting in very little difference between the minimum and maximum average throughput in the class.

4.3.3 The effect of frame length

According to the IEEE 802.16-2004 standard [1], the supported frame lengths for the WirelessMAN-OFDM PHY layer are 2.5ms, 4ms, 5ms, 8ms, 10ms, 12.5ms and

20ms. With a larger frame, Subscriber Stations (SSs) can send more data due to more symbols available in the frame. For example, with the largest frame size of 20ms and symbol duration of 12.5 μ s, the total number of symbols available for the uplink sub-frame will be 800. On the other hand, with the smallest frame size of 2.5ms, only 100 symbols will be available for the uplink sub-frame.

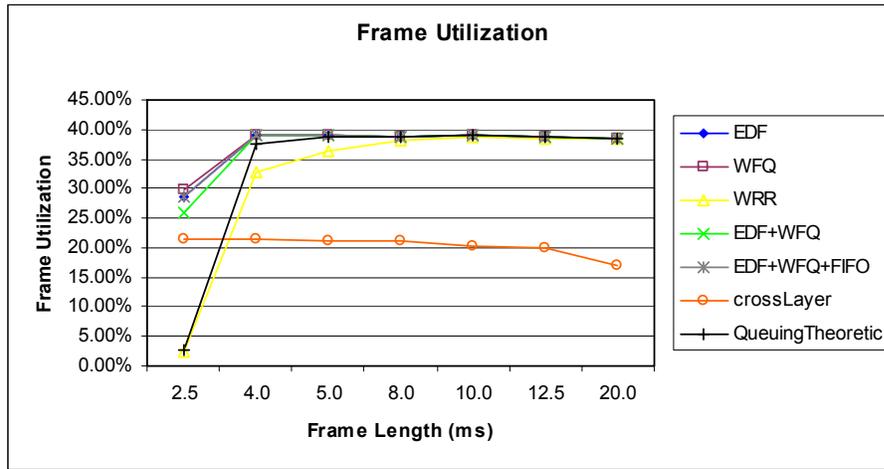
4.3.3.1 The effect of frame length on frame utilization

In this experiment, we will study the frame utilization offered by the various algorithms under light load of 4.5Mbps and a heavy load of 9Mbps. We use video (rtPS) and FTP (nrtPS) traffic only for this experiment. Under the light load, each rtPS SS will send traffic at a rate of 1000 Kbps and the nrtPS SS will send traffic at a rate of 500 Kbps. Under the heavy load, each rtPS SS will send traffic at a rate of 2000Kbps and the nrtPS SS will send traffic at a rate of 1000 Kbps. Table 4-9 lists the parameters used for the experiment.

Table 4-9: The effect of frame length – Parameters (frame utilization)

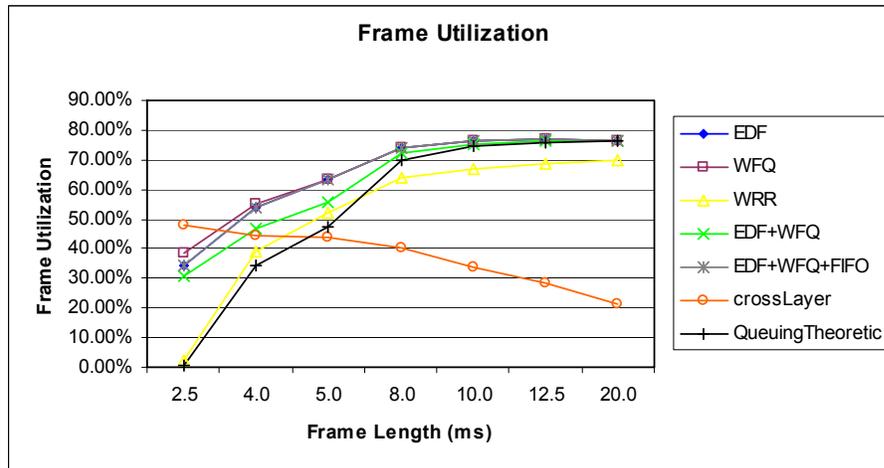
Parameter	Value
Number of SSs	5
Ratio of SS (rtPS:nrtPS)	4:1
Frame Length	2.5ms, 4ms, 5ms, 8ms, 10ms, 12.5ms and 20ms

Light load:



(a) Frame utilization under light load

Heavy load:



(b) Frame utilization under heavy load

Figure 4-15: The Effect of Frame Length – Frame utilization

With a smaller frame size and light load, the cross layer algorithm indicates higher frame utilization than the WRR and Queuing Theoretic algorithms (see Figure 4-15A). This behavior is due to the WRR and Queuing Theoretic algorithms selecting the maximum number of SSs in a frame, thus resulting in the largest overhead. As the frame size increases, the frame utilization increases sharply under the WRR and Queuing Theoretic algorithms, reaching the same level as that of other legacy scheduling

algorithms. This is because with larger frames, the effect of the overhead due to preamble symbols for the five SSs is negligible.

With a smaller frame and under heavy load, the cross layer algorithm indicates the highest frame utilization as the overhead from selecting multiple SSs by the other algorithms is significant (see Figure 4-15B). As the frame size increases, the frame utilization of the legacy and the Queuing Theoretic algorithms increases while that of the cross layer algorithm decreases. The amount of data a single SS has to send remains the same and with a larger frame, the cross layer algorithm indicates lower frame utilization. The frame utilization indicated by the WRR algorithm is lower due to the large packet size of video traffic (150-300 bytes) and FTP traffic (150 bytes). The large packet size will result in more symbols wasted under the WRR algorithm.

4.3.3.2 The effect of frame length on average delay and packet loss

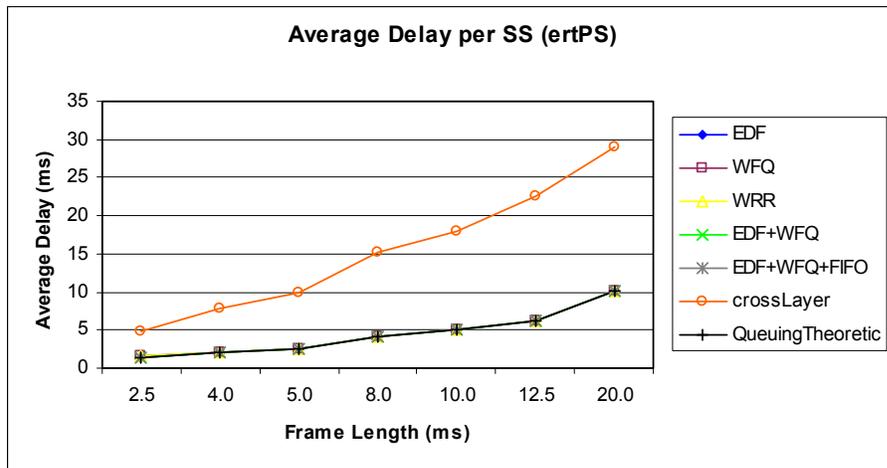
The frame length mostly affects the average delay and packet loss with respect to user performance. Since the traffic load is limited by the smallest frame size, with a large frame length the data to be transmitted will be insignificant compared to the symbols available in the frame. This will result in the scheduling algorithms flushing out all the backlogged data, thus indicating very similar performance between them with respect to the average throughput. The FTP (nrtPS) and HTTP (BE) traffic in this experiment will essentially be treated as background traffic. A heavy load of 4280 Kbps is supplied with 180 Kbps reserved for ertPS class, 3000 Kbps reserved for rtPS class, 1000 Kbps reserved for nrtPS class and 100 Kbps reserved for BE class. A light load of 2140 Kbps is supplied with 90 Kbps reserved for ertPS class, 1500 Kbps reserved for rtPS class, 500

Kbps reserved for nrtPS class and 50 Kbps reserved for BE class. The remaining parameters for the experiment are listed in Table 4-10.

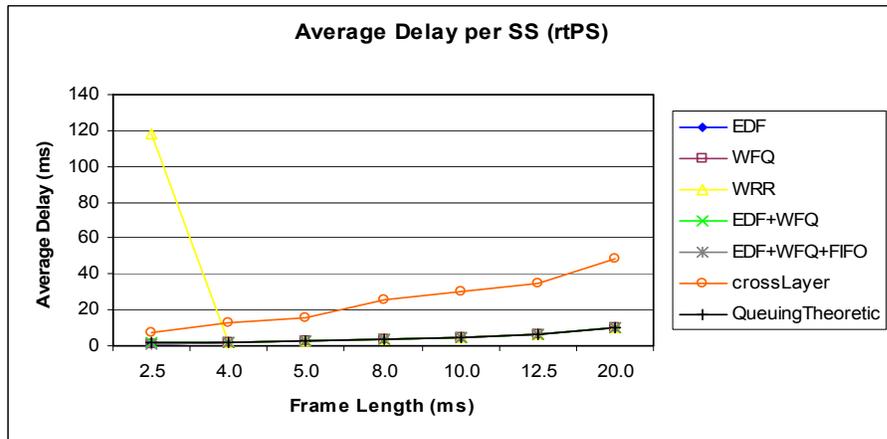
Table 4-10: The effect of frame length – Parameters

Parameter	Value
Number of SSs	5
Ratio of SS (ertPS:rtPS:nrtPS:BE)	1:2:1:1
Frame Length	2.5ms, 4ms, 5ms, 8ms, 10ms, 12.5ms and 20ms

Light load:

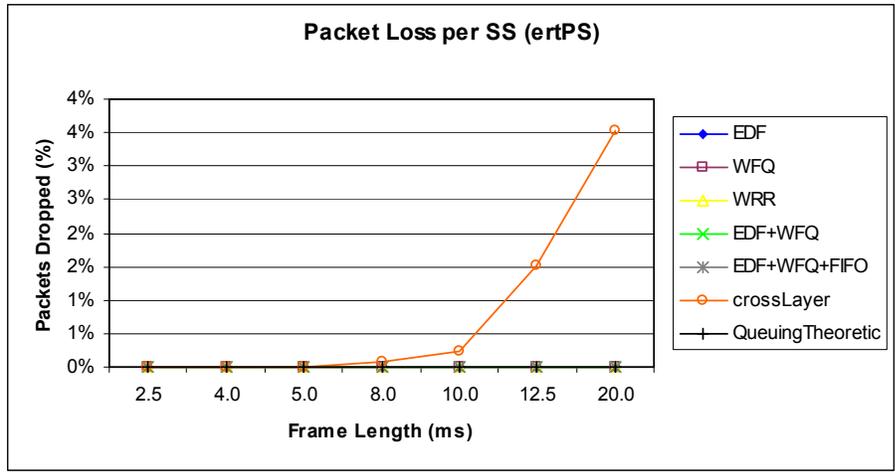


(a) Average Delay - ertPS

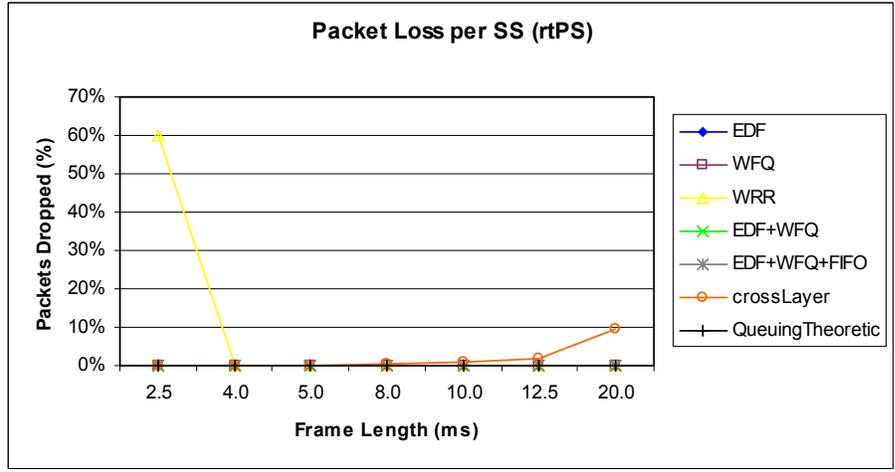


(b) Average Delay - rtPS

Figure 4-16: The effect of frame length: Average delay (Light load)



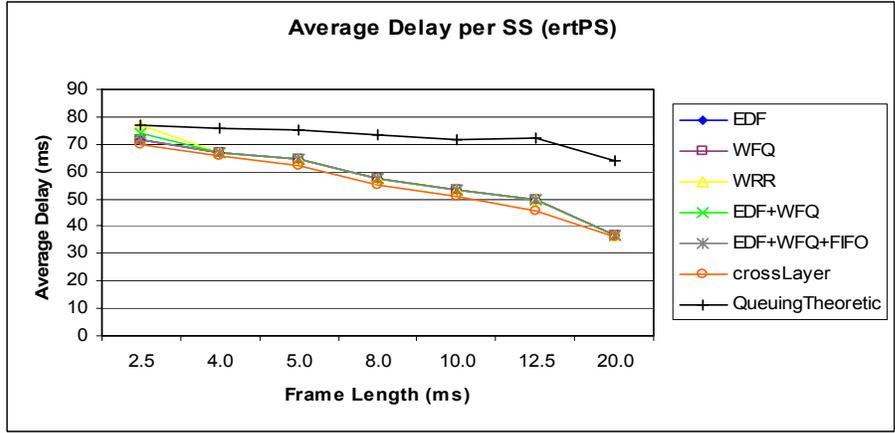
(a) Packet Loss - ertPS



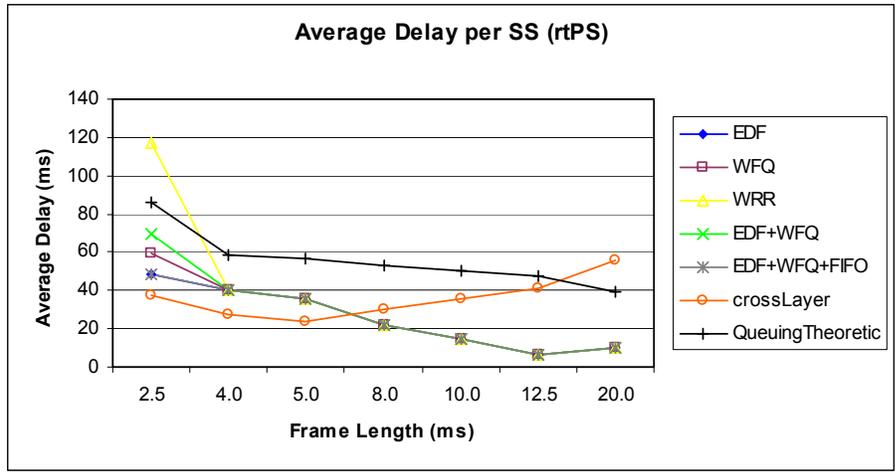
(b) Packet Loss - rtPS

Figure 4-17: The effect of frame length: Packet loss (Light load)

Heavy load:

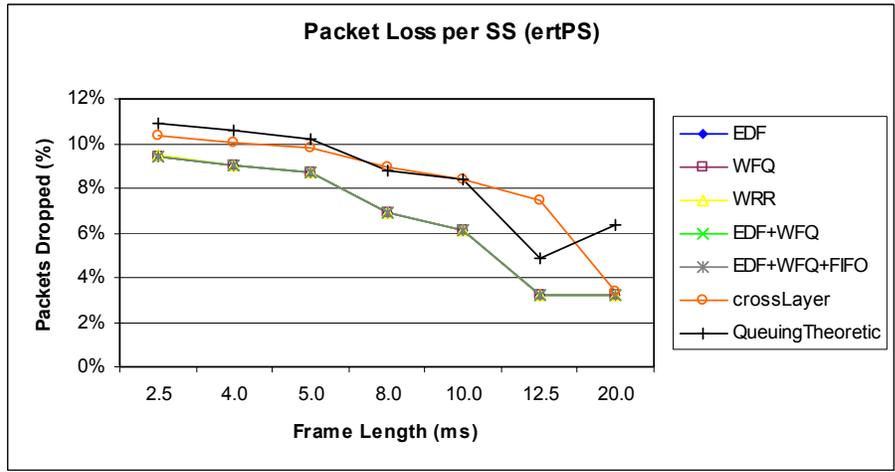


(a) Average Delay - ertPS

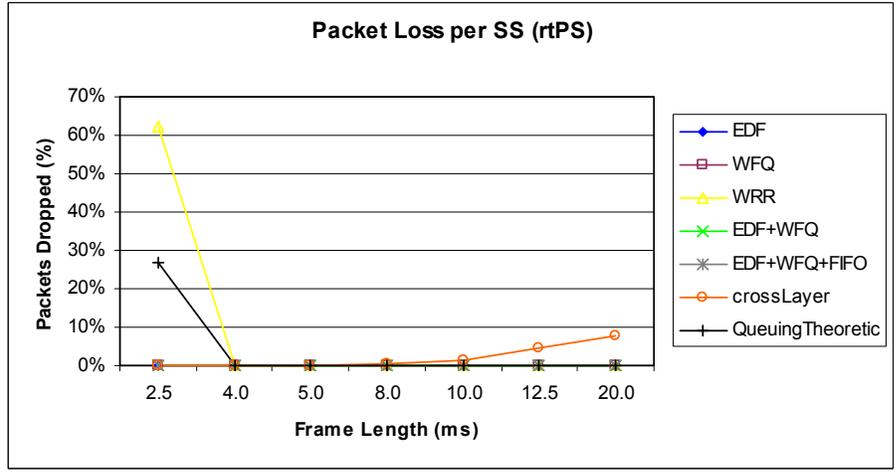


(b) Average Delay - rtPS

Figure 4-18: The effect of frame length: Average delay (Heavy load)



(a) Packet Loss - ertPS



(b) Packet Loss - rtPS

Figure 4-19: The effect of frame length: Packet loss (Heavy load)

Under light load, the average delay of SSs from both the ertPS and rtPS class increases with increasing frame length (see Figure 4-16). This behaviour can be attributed to the packets spending a longer time in the queue with a larger frame size. Even though the average delay increases, most scheduling algorithms indicate no packet loss since abundant bandwidth is available for all the data to be flushed out of the queue (see Figure 4-17). The cross layer algorithm shows an increase in packet loss for frame size greater than 8ms. Since the algorithm selects only one SS in a frame, with a larger frame, packets will spend longer time in the queue than the maximum delay bound, resulting in packet loss.

Under heavy load, the average delay of SSs from both ertPS and rtPS classes decrease with increasing frame length (see Figure 4-18). The decrease in average delay is due to more packets being flushed out of the queue of the SSs. The cross layer algorithm indicates an increase in average delay for the rtPS class as the packets wait a longer time in the queue due to a larger frame size. The average delay of SSs of the ertPS class under the Queuing Theoretic algorithm is higher than that indicated by other algorithms. This is due to the fact that the Queuing Theoretic algorithm tends to satisfy all the SS's MRTR requirements and allocate the residual bandwidth according to the utility of the SSs. Therefore, depending on the channel quality, some SSs will not be allocated bandwidth any further, thus their packets spend a longer time in the queue. The Queuing Theoretic algorithm indicates packet loss for ertPS class but not for the rtPS class (see Figure 4-19). The utility of ertPS SSs is calculated using the same function as that for the rtPS SSs but with a tighter delay bound. Thus, after satisfying the MRTR of each SS, if the residual bandwidth is not enough, packets of ertPS SSs will be dropped first than those of rtPS

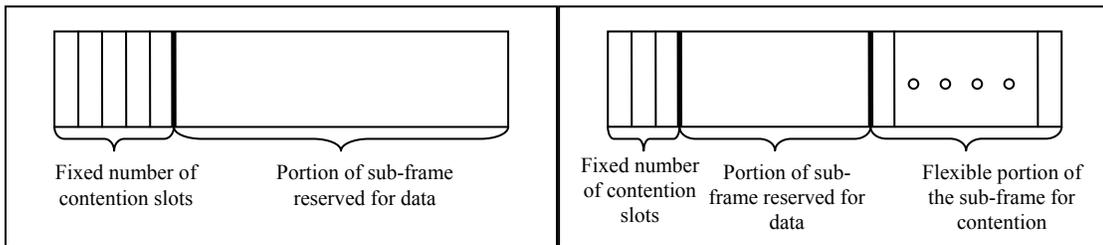
SSs. The packet loss for the rtPS class increases under the cross layer algorithm for frame sizes greater than 8ms. This behaviour is mainly due to the longer frame length that results in the packets waiting a longer period of time in the queue.

4.3.4 Bandwidth Request Analysis

In this experiment, we compare the piggyback and contention request mechanisms, as specified in the IEEE 802.16-2004 standard [1]. In the piggyback mechanism, a SS can attach its bandwidth request onto the data packets whereas in the contention mechanism a SS will compete for a slot to send its bandwidth request. Since piggyback requests do not have a type field, they will always be incremental requests. When a BS receives an incremental bandwidth request, it will add the quantity of the bandwidth requested to its current perception of the bandwidth needs of the SS. Due to the possibility of collisions, bandwidth requests under the contention mechanisms will always be aggregate requests. When a BS receives an aggregate bandwidth request, it will replace its perception of the bandwidth needs of the SS with the quantity of bandwidth requested.

The experiment will be carried out with SSs of the nrtPS and BE classes as contention for bandwidth is allowed for these classes only. The IEEE 802.16-2004 standard [1] specifies that the block of contention slots can occur in any order and in any quantity within the frame (limited by the number of available symbols in the frame) at the discretion of the BS uplink scheduler as indicated by the UL-MAP message. We compare the piggyback mechanism with the contention request mechanism under two different ways of allocating slots for contention. In the first scheme, a fixed number of slots, equal to the number of SSs, will be reserved for contention. The second scheme is an enhancement to the first whereby a fixed number of slots, equal to half the number of

SSs, are reserved and the SSs are also allowed to contend in the unused portion of the uplink sub-frame. The unused portion of the sub-frame is the part of the sub-frame that is not reserved for data symbols. From the UL-MAP message, each SS can determine the unused portion of the uplink sub-frame. We compare the piggyback and contention mechanisms under heavy load of 6Mbps and a light load of 1.5Mbps. For the heavy load, the total traffic arrival rate of nrtPS SSs is 5Mbps and that of BE SSs is 1Mbps. For the light load, the total traffic arrival rate of nrtPS SSs is 1.2Mbps and that of BE SSs is 0.3Mbps. Figure 4-20 provides an overview of the two contention request mechanisms evaluated and Table 4-11 lists the parameters of the experiment.



(a) Contention Mechanism: Fixed number of slots

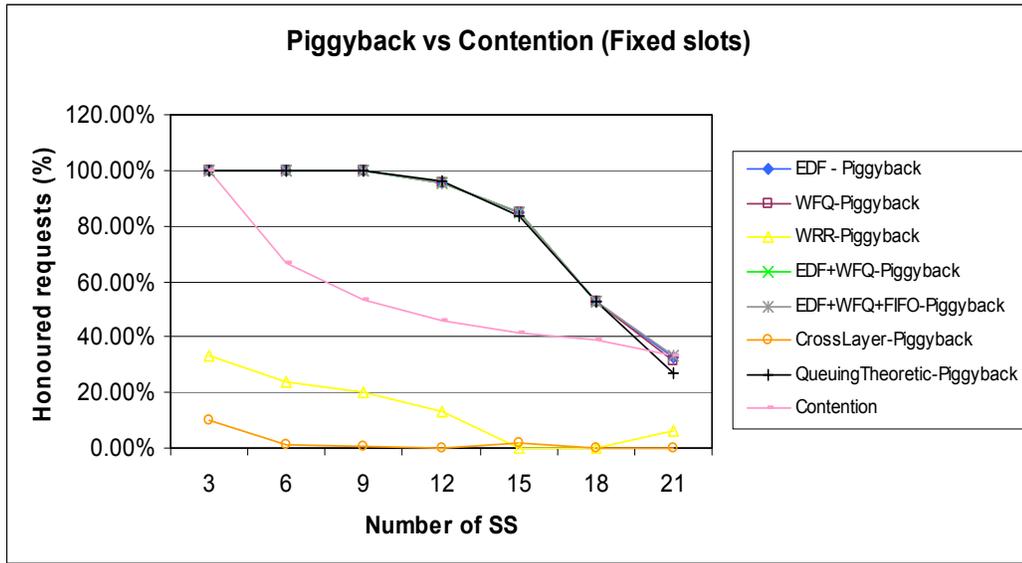
(b) Contention Mechanism: Variable number of slots

Figure 4-20: Uplink sub-frame structure: Contention Request Mechanism

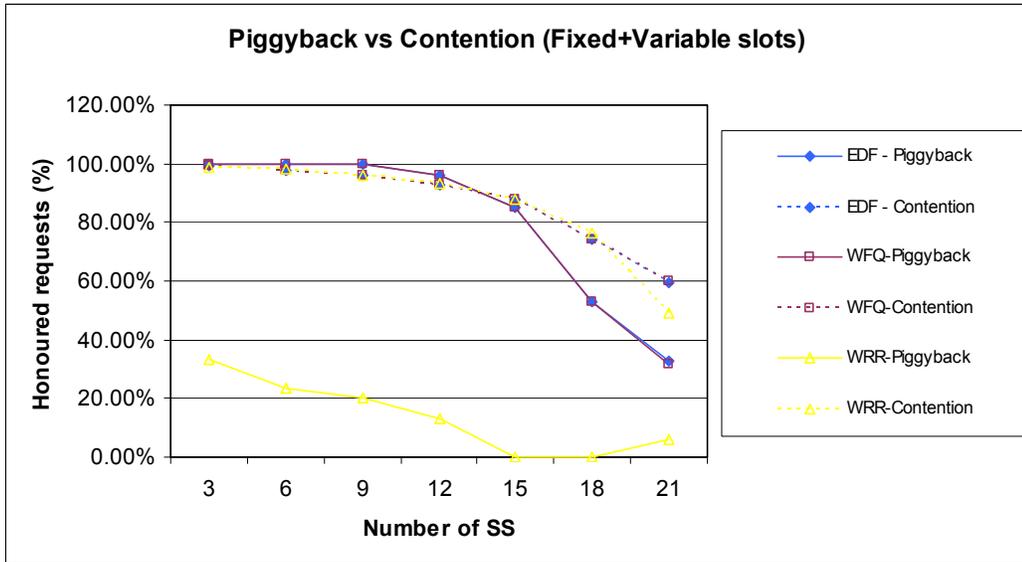
Table 4-11: Bandwidth request analysis - Parameters

Parameter	Value
Number of SSs	3-21
Ratio of SS (nrtPS:BE)	2:1
Frame Length	10ms

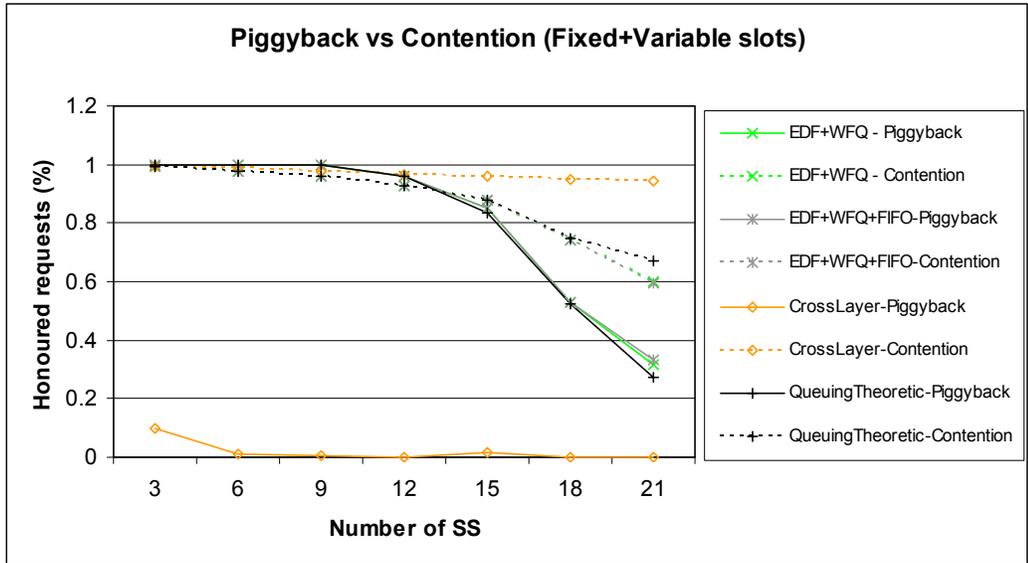
Light load:



(a) Piggyback vs Contention (Fixed slots)



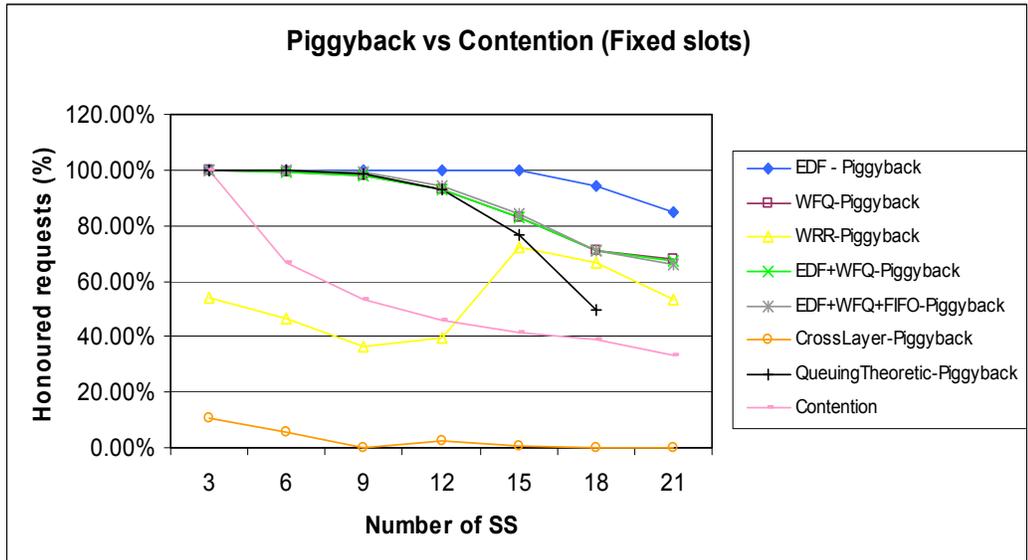
(b) Piggyback vs Contention (Variable slots) – Homogenous algorithms



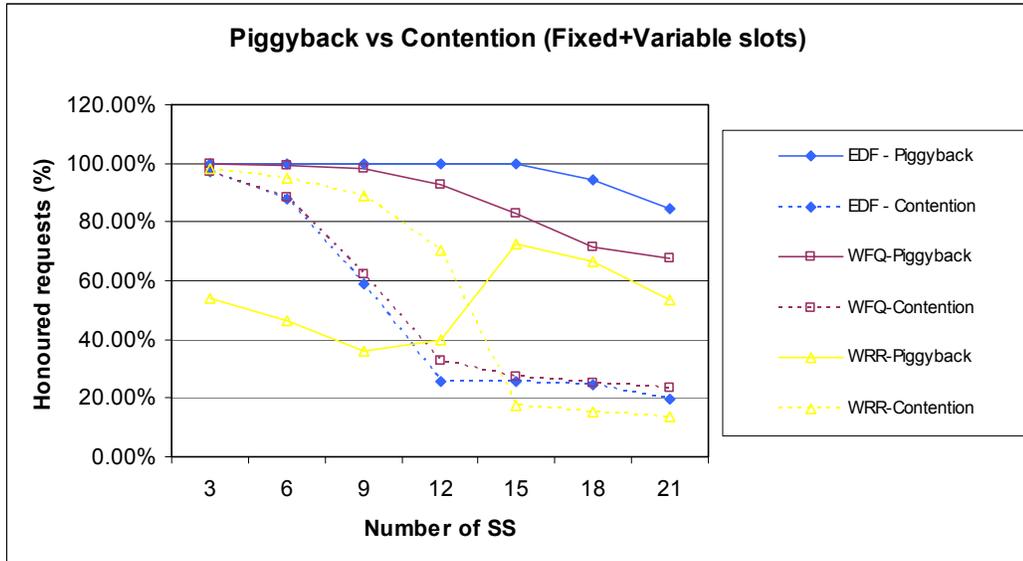
(c) Piggyback vs Contention (Variable slots) – Hybrid and Opportunistic algorithms

Figure 4-21: Bandwidth request analysis – Light load

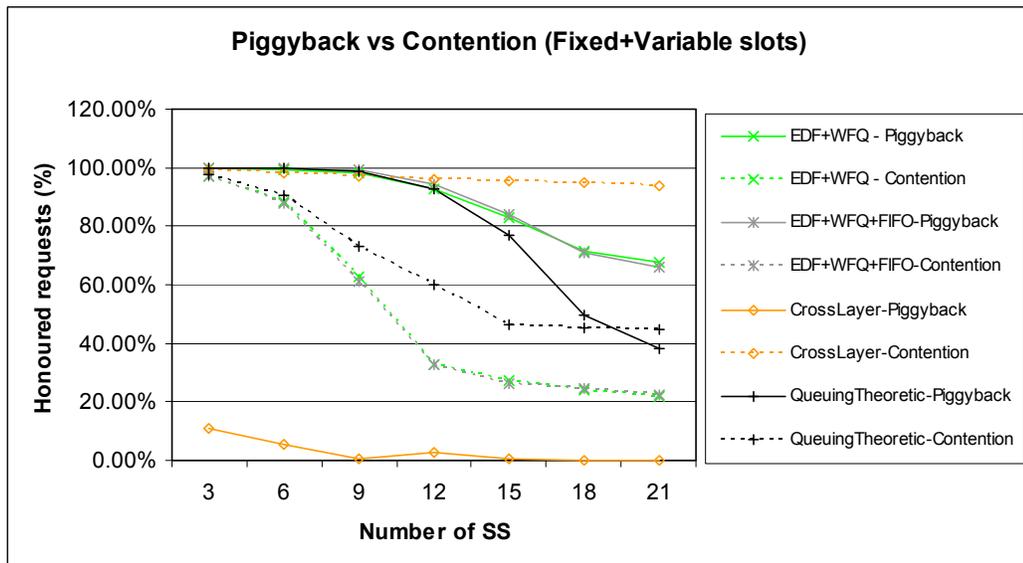
Heavy load:



(a) Piggyback vs Contention (Fixed slots)



(b) Piggyback vs Contention (Variable slots) – Homogenous algorithms



(c) Piggyback vs Contention (Variable slots) – Hybrid and Opportunistic algorithms

Figure 4-22: Bandwidth request analysis – Heavy load

We can observe that the contention mechanism is more suitable for the cross layer algorithm than the piggyback mechanism, under both light and heavy load (see Figure 4-21 and Figure 4-22). This behaviour is due to the cross layer algorithm selecting only one SS in a frame that will result in only one SS getting a chance to piggyback its bandwidth request each frame. The cross layer algorithm indicates very good

performance under the contention mechanism since a major portion of the uplink sub-frame will be unused, also indicated by low frame utilization (see Figure 4-6). The unused portion will be used for contention, resulting in a high success rate of reserving a slot for bandwidth request.

Under light load, both the contention mechanisms (see Figure 4-21) indicate a better chance for the SSs to reserve a slot for bandwidth request than the piggyback mechanism, under the Queuing Theoretic and legacy algorithms. This behaviour is due to most of the data from the queues of the SSs being flushed out, resulting in few backlogged packets to piggyback bandwidth requests. Due to the light load, abundant symbols will be available for the contention mechanism resulting in very few collisions. With fewer SSs, the piggyback mechanism indicates a slightly higher percentage of honoured requests, due to larger load per SS. As the number of SSs increase, the load per SS decreases, which means some SSs will not have any data backlog to piggyback a bandwidth request, resulting in the contention mechanism showing superior performance than the piggyback mechanism. Under the piggyback mechanism, the WRR algorithm indicates a low percentage of honoured requests compared to other algorithms. Initially the percentage of honoured requests decrease since the bandwidth allocated based on the weight is enough to transmit all the packets from the queues of the SSs. When the number of SSs is 21, the bandwidth allocated to each SS is not enough, thus preventing a SS from sending its backlogged packets. This is because of the large number of SSs resulting in a large overhead and thus less bandwidth available for each SS.

Under heavy load, the piggyback mechanism indicates a higher percentage of honoured requests than both contention mechanisms (see Figure 4-22). The EDF

algorithm results in the highest percentage of honoured requests under the piggyback mechanism than all the other algorithms. This is due to both nrtPS and BE SSs competing for bandwidth that results in a large backlog of data in the frame and allows the SS to piggyback bandwidth request. The hybrid (EDF+WFQ+FIFO) algorithm results in a lower percentage of honoured requests under the piggyback mechanism since it gives strict priority to nrtPS SSs. This will result in smaller backlog of data for nrtPS SSs and due to their high concentration (2:1 ratio) it will indicate a lower percentage of honoured requests. The WFQ and hybrid (EDF+WFQ) algorithms also indicate a lower percentage of honoured requests under the piggyback mechanism. This behaviour is due to a higher weight assigned to nrtPS SSs that results in smaller backlog of data in their queues.

For fewer SSs (less than 12), the WRR algorithm indicates a higher percentage of honoured requests under the contention mechanism than under the piggyback mechanism. Due to fewer SSs, the bandwidth allocated to each SS is enough to flush out all the data resulting in fewer packets remaining to piggyback the bandwidth request. With a large number of SSs (greater than 12), the bandwidth allocated to each SS is not enough to transmit all the packets from the queue resulting in the piggyback mechanism performing better than the contention mechanism. When the number of SSs is greater than 15, the piggyback mechanism indicates a decrease in honoured requests. This behaviour is due to less bandwidth allocated to each SS and the large packet size of BE and nrtPS traffic. The small amount of bandwidth allocated is not enough to transmit one packet, and therefore a large number of SSs don't get selected. Even if a SS has large backlog of data, if the SS is not selected in the current frame, it cannot piggyback on the bandwidth request.

The Queuing Theoretic algorithm indicates a higher percentage of honoured requests under the contention mechanism (fixed+variable slots) than the legacy algorithms (see Figure 4-22). This behaviour is due to the limitation on the bandwidth allocation per class under the Queuing Theoretic algorithm. Due to a lower traffic arrival rate of BE SSs, some of the bandwidth allocated to the BE class will be unused resulting in more symbols available for contention.

4.4 Summary

In this chapter we evaluated the performance of a number of representative WiMAX scheduling schemes. We presented the simulation environment, including the traffic model and values for the MAC and PHY layer simulation parameters. We conducted a series of experiments to observe the performance of scheduling algorithms under the context of IEEE 802.16 MAC layer. Table 4-12 provides a summary of our findings.

The experiments reveal that the overhead due to the uplink burst preamble can significantly affect the bandwidth available to the SSs. Scheduling algorithms that select maximum number of SSs (WRR, Queuing Theoretic) result in maximum overhead compared to algorithms (cross layer) that select the least number of SSs. Even though the cross layer algorithm selects only one SS in a frame, it provides superior performance than the other algorithms in some select cases. When the load per SS is high, with a smaller frame size the cross layer algorithm will result in higher frame utilization. As well, when the number of SSs is high, the cross layer algorithm will result in higher frame utilization. The advantages of the cross layer algorithm are limited to frame

utilization only as it indicates poor performance with respect to average throughput, average delay and fairness under all the scenarios we studied.

Table 4-12: Comparison of uplink scheduling algorithms in WiMAX

Scheme	Fairness		Suitable bandwidth request mechanism	Frame utilization	Average Throughput (ertPS/ rtPS/ nrtPS/ BE)	Average Delay/ Packet loss
	Intra-class	Inter-class	(Heavy load/ Light load)			
EDF	High	Low	Piggyback / Contention	High	High/ High/ Low/ Low	Low/ Low
WFQ	Low	Low-Medium	Piggyback / Contention	High	Medium/ High/ High/ Low-Medium	Low/ Low
WRR	High	High	Piggyback/ Contention	Low, Medium ¹	Low/ Medium/ Medium/ Low-Medium	Medium-High/ Medium-High
EDF+WFQ	Medium	Medium	Piggyback/ Contention	Medium, High ²	High/ Medium-High/ High/ Low-Medium	Medium-High/ Medium-High
EDF+WFQ +FIFO	Medium	Low-Medium	Piggyback/ Contention	High	High/ High/ High/ Medium/ Low	Low/ Low
Queuing Theoretic	Low-Medium	Low-Medium	Piggyback/ Contention	Medium, Low ³	High/ High/ High/ Medium-High	Low- Medium/ Low-Medium
Cross layer	Low	Low	Contention/ Contention	Low, High ⁴	Low/ Low/ Low/ Low	High/ High

¹ Low – when number of SSs is large or frame size is small, Medium – when number of SSs is small or frame size is large.

² Medium – when number of SSs is large or frame size is small, High – when number of SSs is small or frame size is large.

³ Low – when number of SSs is large or frame size is small, Medium – when number of SSs is small.

⁴ Low – when number of SSs is small, High – when number of SSs is large or frame size is small.

The scheduling algorithms show interesting results when studied under different mix of traffic. Algorithms such as EDF and hybrid (EDF+WFQ+FIFO) indicate superior performance for SSs of ertPS and rtPS classes with respect to average throughput, average delay and packet loss when the concentration of real-time traffic is high. These algorithms will also result in starvation of SSs of nrtPS and BE classes. The difference between EDF and hybrid (EDF+WFQ+FIFO) algorithms is that under the EDF algorithm SSs of both nrtPS and BE classes will compete for bandwidth whereas in the hybrid (EDF+WFQ+FIFO) algorithm, SSs of nrtPS class have strict priority over SSs of the BE class. The performance of the Queuing Theoretic algorithm is limited by the class thresholds and that of the hybrid (EDF+WFQ) algorithm is limited by the allocation of bandwidth among the traffic classes. Under the Queuing Theoretic algorithm, if the bandwidth allocated to SSs of a class is equal to the threshold of that class, the SSs of this class will not be allocated bandwidth any further in the current frame even if these SSs have a large backlog of data. The WRR algorithm indicates poor performance under variable packet size since bandwidth is allocated to the SSs based solely on their weight.

The frame length also has a significant impact on the performance of the scheduling algorithms. The WRR and Queuing Theoretic algorithms result in very poor performance with small frame sizes since most of the available resources will be wasted for the uplink burst preamble. This is because the algorithms tend to select the maximum number of SSs, resulting in maximum overhead. Due to the larger frame duration, the average delay and packet loss experienced by the SSs will also increase.

The performance of the scheduling algorithms is also influenced by the bandwidth request mechanism adopted by the SSs. The cross layer algorithm will result in a very

high success rate for reserving a slot for bandwidth request under the contention mechanism than under the piggyback mechanism. On the other hand, under the Queuing Theoretic and legacy algorithms (homogenous and hybrid), the SSs have a better chance of reserving a bandwidth request slot when using the contention mechanism under light load and the piggyback mechanism under heavy load. The number of slots reserved for contention has a significant impact on the success rate of reserving a bandwidth request slot.

Chapter 5. Conclusions and Future Work

In recent years, the need for broadband wireless access has increased due to applications such as video conferencing, VoIP, online gaming and streaming audio/video demanding high bandwidth and tight delay bounds. The IEEE 802.16 standard specifies a means of broadband internet access for fixed and mobile stations and promises to provide last mile internet access at an affordable rate. The IEEE 802.16-2004 standard specifies the MAC and PHY layer functionalities for “Fixed” WiMAX for both PMP and mesh mode operations. The standard states a QoS framework that specifies four traffic classes, but the scheduling mechanism for these classes is left open for vendor implementation. The choice of a scheduling algorithm for the multi-class traffic can have a significant impact on the satisfaction of the users. In this thesis, we investigated several proposals for uplink scheduling algorithms aimed at satisfying QoS requirements of the multi-class traffic. We categorized the uplink scheduling algorithms into three classes, namely, homogenous algorithms, hybrid algorithms and opportunistic algorithms. Representative algorithms from each category were selected for evaluation. The algorithms were evaluated under different mix of traffic and with respect to the major characteristics of IEEE 802.16 MAC layer such as bandwidth request mechanisms, frame size and the uplink burst preamble.

The performance metrics used to evaluate the scheduling algorithms are average throughput, average delay, packet loss, frame utilization and fairness. Fairness was studied at two levels: intra-class fairness and inter-class fairness. Jain’s fairness index is used to measure inter-class fairness where Min-max index is used for intra-class fairness.

The Min-max fairness index is more sensitive to service degradation and unfairness between users, allowing us to better distinguish between the scheduling algorithms performance within the same class. To effectively study the bandwidth request mechanisms, we introduced a metric, honoured requests, which measures the percentage of successful bandwidth requests.

The EDF and hybrid (EDF+WFQ+FIFO) algorithms result in the lowest average delay for ertPS and rtPS SSs. Both these algorithms provide strict priority to ertPS and rtPS SSs. They also do not consider the MRTR of the SSs in deciding the transmission schedule and result in starvation of lower priority SSs. The difference between the EDF and hybrid (EDF+WFQ+FIFO) algorithm is that in the former, the nrtPS and BE SSs will compete for bandwidth whereas in the latter the nrtPS SSs have strict priority over BE SSs. Therefore, scheduling algorithms that employ a strict priority mechanism are not a good choice to satisfy the QoS requirements of the multi-class traffic in WiMAX.

The WRR, WFQ and hybrid (EDF+WFQ) algorithms provide a more fair distribution of bandwidth among the SSs. The WFQ and WRR algorithms attempt to satisfy the MRTR of the SSs by assigning weights to the SSs based on their MRTR. The worst case delay bound guaranteed by the WFQ algorithm can be sufficient for the UGS SSs but not for ertPS and rtPS SSs. The average delay under the WFQ algorithm is greatly affected by the density of the traffic utilizing the residual capacity i.e. under bursty traffic, such as VoIP, the average delay experienced by the SSs will be high. Therefore, modifications to the algorithm are required to satisfy the delay requirement of the ertPS and rtPS SSs and ensure the average delay is not considerably affected by the traffic density. The WRR algorithm does not provide a bound on the delay, even under the worst case, and it does

not work well in the presence of variable packet size. The WRR algorithm also results in the largest overhead since it selects all the SSs in each frame. The packet size and the queue length can be incorporated in the WRR algorithm to provide a delay bound and ensure the algorithm operates well in the presence of variable packet sizes. This might require sacrificing the simplicity of the algorithm which is one of its major advantages. The hybrid (EDF+WFQ) algorithm allocates bandwidth among the traffic classes in a fairer manner than the hybrid (EDF+WFQ+FIFO) algorithm. This algorithm is also very adaptable to changing concentration of traffic since the bandwidth among the traffic classes is distributed according to the number of SSs in a traffic class and their MRTR. The use of a computationally expensive algorithm such as WFQ for BE SSs is an overwhelming solution. Since BE SSs do not have QoS requirements, simple algorithms such as FIFO or RR would be sufficient.

The priority functions of the cross layer algorithm take into account all the QoS requirements of the multi-class traffic in WiMAX such as the average delay, average throughput and the channel quality. Even though the algorithm selects only one SS in a frame, it shows superior performance compared to the other algorithms in select cases. One of these cases is when the number of SSs is large and the load per SS is high, the cross layer algorithm results in the least overhead due to uplink burst preamble compared to other algorithms that select multiple SSs in a frame. The algorithm results in dissatisfaction of the QoS requirements of the SSs and starvation of lower priority SSs, since it selects only one SS in each frame. The algorithm is also very sensitive to the coefficients assigned to each traffic class, which is one of the reasons for starvation of lower priority SSs in the presence of large number of higher priority SSs.

The Queuing Theoretic algorithm considers the queue size of the SSs, which is an important factor in the presence of bursty traffic and variable packet size. The allocation within a traffic class is limited by the class threshold. Therefore, it is critical that an appropriate threshold be assigned to each class to ensure the QoS requirements of the SSs in the class are met. The algorithm ensures that BE SSs are not starved even in the presence of large number of ertPS, rtPS and nrtPS SSs. A drawback of the algorithm is that the utility function for rtPS SSs attempts to satisfy the delay requirement only. MRTR is also an important QoS parameter of the rtPS class, but is ignored in the utility function. The algorithm also results in large overhead due to uplink burst preamble since it selects all the SSs in each frame.

Legacy scheduling algorithms (homogenous and hybrid) do not explicitly consider all the required QoS parameters of the traffic classes in WiMAX. The algorithms consider only some of the parameters which are not sufficient since the scheduling classes have multiple QoS parameters such as the rtPS class that requires delay, packet loss and throughput guarantee. The algorithms implicitly attempt to meet the QoS requirements such as the EDF algorithm satisfying the delay requirement of ertPS and rtPS SSs. On the other hand, the Cross Layer and Queuing Theoretic algorithms include the maximum latency, MRTR and the channel quality in the priority functions. Therefore, the legacy scheduling algorithms are not the most suitable for the multi-class traffic in WiMAX, as they do not exploit the characteristics of WiMAX and the requirements of the various traffic classes. Although, the Cross Layer and Queuing Theoretic algorithms include all the QoS parameters in their priority/utility functions, they have certain drawbacks as well.

Since the WFQ algorithm does not perform well under traffic that is bursty such as VoIP, more suitable algorithms for VoIP traffic would be EDF and Queuing Theoretic. Both EDF and Queuing Theoretic algorithms would also be suitable for other real-time applications such as video conferencing and streaming media. If the number of SSs is very high, then the Queuing Theoretic algorithm would not provide desirable performance as it results in the maximum preamble overhead. Queuing Theoretic algorithm can be used in networks with high traffic density but few SSs, whereby many hosts are connected to each SS. The WFQ algorithm would be the most suitable if the type of traffic is predominantly non real-time such as FTP traffic. Based on the simulation results, we observed that the WFQ algorithm tends to provide superior performance with respect to average throughput, which is the most important QoS requirement for delay tolerant applications such as FTP.

Next, we propose enhancements to the scheduling algorithms to address some of these drawbacks and discuss some of the open problems:

- **Multiple SSs:** A major drawback of the cross layer scheduling algorithm is its inability to select multiple SSs in a frame. To incorporate multiple SSs in the algorithm while maintaining its promise of satisfying the QoS requirements of the multi-class traffic in WiMAX, bandwidth in a frame can be allocated according to the normalized priority of the SSs. The normalized priority of a SS is the priority of the SS relative to the sum of priority of all the SSs (equation 5.1). More specifically, if the bandwidth available in a frame is C and the priority of connection i is ϕ_i , then the bandwidth allocated in one frame to the connection is:

$$b_i = C * \left(\frac{\phi_i}{\sum_{j=1}^n \phi_j} \right) \quad (5.1)$$

where $\sum_{j=1}^n \phi_j$ is the sum of priority of all the SSs.

The above formulation should allow for a more fair allocation of bandwidth among all the SSs. It will also ensure that the lower priority SSs (SSs of nrtPS and BE classes) do not starve in the presence of large number of SSs of ertPS and rtPS classes, as in original algorithm. To ensure that the MRTR of the SSs is satisfied, bandwidth equivalent to the MRTR can first be allocated to all the SSs. Any residual bandwidth can be allocated according to equation 5.1.

- Packet size:** Based on the simulation results, we noticed that the WRR scheduling algorithm indicates poor performance when the traffic contains packets of variable size or when the packet size is too large. To resolve this issue, the packet size of the traffic needs to be included in calculating the weight of the SSs. A variation of WRR called Variably Weighted Round Robin (VWRR) is proposed in [43] that adaptively changes the weight of a SS based on the mean packet size. In this algorithm, if the average packet length at any instance in time is smaller than the maximum average packet length, then the weight of the connection at this instance increases. It has been shown analytically that the fairness of VWRR in the best case equals that of Deficit Round Robin (DRR) and in the worst case equals that of WRR. This scheme would be more suitable for the heterogeneous traffic in WiMAX.
- Queue length:** The queue length has a significant impact on the average delay and the packet loss. Therefore, the scheduling algorithm needs to incorporate the queue

size to minimize the delay and packet loss. Additionally, allocating resource for a SS with an empty or near empty queue will waste resources. A queue aware uplink bandwidth allocation and rate control mechanism is proposed in [44]. The bandwidth allocation mechanism adaptively allocates bandwidth for polling service in the presence of higher priority UGS service by exploiting the queue status information. Several scheduling algorithms exist that exploit the queue size in wired networks [45], [46], [47]. H.Wang et al. [47] propose a scheduling algorithm to support premium service in Differentiated Services (DiffServ) architecture [48]. This algorithm assigns a weight to each connection based on the average queue size, minimum and maximum thresholds. The minimum threshold represents the desired queuing delay and the maximum threshold represents the acceptable queuing delay. These algorithms or variations of them can be used to ensure a reasonable delay can be provided for the real-time traffic while ensuring the QoS requirements of other types of traffic are satisfied.

- **Bandwidth request mechanism:** The choice of bandwidth request mechanism significantly affects the performance of the scheduling algorithms. We studied the performance of the algorithms under the piggyback and contention request mechanisms. A critical part of the contention request mechanism is the number of slots reserved for contention. We evaluated two ways of reserving slots for contention; the first whereby fixed number of slots are reserved and the second scheme that initially reserves fixed number of slots and utilizes any further available slots. The number of fixed slots to be reserved for contention, under both contention slot reservation schemes, requires further evaluation under different traffic loads. If

large number of slots is reserved, the slots available for data will be less resulting in lower throughput. If fewer slots are reserved, the success of reserving a slot for contention will be low but the throughput will increase due to more slots available for data.

- **Call Admission Control (CAC):** The performance of an uplink scheduling algorithm is highly dependant on the Call Admission Control (CAC) scheme adopted. Hence, we applied a basic CAC scheme for all the scheduling algorithms. The Queuing Theoretic algorithm proposed in [29] employs a CAC algorithm based on class thresholds. The class thresholds are calculated based on the connection blocking probability while maximizing the average system revenue. The purpose of the thresholds is to limit the amount of bandwidth allocated to each traffic class. Further investigation of all the algorithms needs to be carried out under various common CAC algorithms. Based on the results, we can select a CAC algorithm that can provide reasonable performance for all the scheduling algorithms or identify the shortcomings, if any.
- **Inter-class bandwidth allocation:** The most critical part of hybrid algorithms is distribution of bandwidth among the traffic classes. We discussed the merits of hybrid (EDF+WFQ+FIFO) algorithm that uses a strict priority mechanism and of the hybrid (EDF+WFQ) algorithm that distributes bandwidth according to the MRTR and number of Ss within the same traffic class. The bandwidth can also be distributed among the traffic classes to maximize fairness or user satisfaction. The issue of distribution of bandwidth among traffic classes in hybrid algorithms requires further study.

Bibliography

- [1] Y.Cao and V.Li, "Scheduling Algorithms in Broadband Wireless Networks", Proceedings of the IEEE, pp.76-87, January 2001.
- [2] B.Skrikar, "Packet Scheduling Algorithms to Support QoS in Networks", Masters Thesis, Indian Institute of Technology, 71 pp., October 1999.
- [3] IEEE 802.16-2004, "IEEE Standard for Local and Metropolitan Area Networks – Part 16: Air Interface for Fixed Broadband Wireless Access Systems", October 2004.
- [4] IEEE 802.16a-2003, "IEEE standard for Local and Metropolitan Area Networks – Part 16: Air Interface for Fixed Broadband Wireless Access Systems – Medium Access Control Modifications and Additional Physical Layer Specifications for 2-11GHz", January 2003.
- [5] IEEE 802.16c-2002, "IEEE Standard for Local and Metropolitan Area Networks – Part16: Air Interface for Fixed Broadband Wireless Access Systems – Amendment1: Detailed System Profiles for 10-66 GHz", December 2002.
- [6] IEEE 802.16e-2005, "IEEE Standard for Local and Metropolitan Area Networks – Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems Amendment 2: Physical and Medium Access Control Layers for Combined Fixed and Mobile Operation in Licensed Bands", February 2006.
- [7] IEEE 802.16f-2005, "IEEE Standard for Local and Metropolitan Area Networks – Part 16: Air Interface for Fixed Broadband Wireless Access Systems – Amendment1: Management Information Base", December 2005.
- [8] IEEE 802.16g, "Unapproved Draft IEEE Standard for Local and Metropolitan Area Networks – Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems – Amendment 3: Management Plane Procedures and Services", February 2007.
- [9] J.Wolnicki, "The IEEE 802.16 WiMAX Broadband Wireless Access; Physical Layer (PHY), Medium Access Control (MAC) layer, Radio Resource Management", *Seminar on Topics in Communications Engineering*, January 2005.
- [10] C.Eklund, R.Marks, K.Stanwood and S.Wang, "IEEE standard 802.16: a technical overview of the WirelessMANTM air interface for broadband wireless access", *IEEE Communications Magazine*, pp.98-107, June 2002.

- [11] K.Lee, J.Hahm and Y.Kim, "QoS Application Method in Portable Internet", *Proceedings of Asia-Pacific Conference on Communications*, pp.237-239, October 2005.
- [12] C.Cicconetti, A.Erta, L.Lenzini and E.Mingozzi, "Performance Evaluation of the IEEE 802.16 MAC for QoS Support", *IEEE Transactions on Mobile Computing*, vol.6, no.1, pp.26-38, January 2007.
- [13] M.Katevenis, S.Sidiropoulos and C.Courcoubetis, "Weighted Round-Robin Cell Multiplexing in a General-Purpose ATM Switch Chip", *IEEE Journal on Selected Areas in Communications*, vol.9, pp.1265-1279, October 1991.
- [14] M.Shreedhar and G.Varghese, "Efficient Fair Queuing using Deficit Round Robin", *IEEE/ACM Transactions on Networking*, vol.1, no.3, pp.375-385, June 1996.
- [15] N.Ruangchaijatupon, L.Wang and Y.Ji, "A Study on the Performance of Scheduling Schemes for Broadband Wireless Access Networks", *Proceedings of International Symposium on Communications and Information Technology*, pp. 1008-1012, October 2006.
- [16] D.Ferrari and D.Verma, "A scheme for real-time channel establishment in wide-area networks", *IEEE Journal on Selected Areas in Communications*, vol.8, no.3, pp.368-379, April 1990.
- [17] T.Tsai, C.Jiang and C.Wang, "CAC and Packet Scheduling Using Token Bucket for IEEE 802.16 Networks", *Journal of Communications*, vol.1, no.2., pp.30-37, May 2006.
- [18] A.Parekh, R.Gallager, "A Generalized Processor Sharing Approach to Flow Control in Integrated Services Networks: The Single Node Case", *IEEE/ACM Transactions on Networking*, vol.1, pp.344-357, June 1993.
- [19] Y.Shang and S.Cheng, "An Enhanced Packet Scheduling Algorithm for QoS Support in IEEE 802.16 Wireless Network", *Proceedings of 3rd International Conference on Networking and Mobile Computing*, pp.652-661, August 2005.
- [20] J.Bennett and H.Zhang, "Hierarchical packet fair queuing algorithms", *IEEE/ACM Transactions on Networking*, vol.5, pp.675-689, October 1997.
- [21] J.Bennett and H.Zhang, "WF²Q: Worst-case fair weighted fair queuing", *Proceedings of INFOCOM '96*, pp.120-128, March 1996.
- [22] O.Yang and J.Lu, "Call Admission Control and Scheduling Schemes with QoS Support for Real-time Video Applications in IEEE 802.16 Networks", *Journal of Multimedia*, vol.1, no.2, May 2006.

- [23] K. Wongthavarawat, and A. Ganz, "Packet scheduling for QoS support in IEEE 802.16 broadband wireless access systems", *International Journal of Communication Systems*, vol. 16, issue 1, pp. 81-96, February 2003.
- [24] K.Vinay, N.Sreenivasulu, D.Jayaram and D.Das, "Performance evaluation of end-to-end delay by hybrid scheduling algorithm for QoS in IEEE 802.16 network", *Proceedings of International Conference on Wireless and Optical Communication Networks*, 5 pp., April 2006.
- [25] M.Settembre, M.Puleri, S.Garritano, P.Testa, R.Albanese, M.Mancini and V.Lo Curto, "Performance analysis of an efficient packet-based IEEE 802.16 MAC supporting adaptive modulation and coding", *Proceedings of International Symposium on Computer Networks*, pp.11-16, June 2006.
- [26] J.Lin and H.Sirisena, "Quality of Service Scheduling in IEEE 802.16 Broadband Wireless Networks", *Proceedings of First International Conference on Industrial and Information Systems*, pp.396-401, August 2006.
- [27] Q.Liu, X.Wang and G.Giannakis, "Cross-layer scheduler design with QoS support for wireless access networks", *Proceedings of International Conference on Quality of Service in Heterogeneous Wired/Wireless Networks*, 8 pp., August 2005.
- [28] H.Rath, A.Bhorkar and V.Sharma, "An Opportunistic Uplink Scheduling Scheme to Achieve Bandwidth Fairness and Delay for Multiclass Traffic in Wi-Max (IEEE 802.16) Broadband Wireless Networks", *Proceedings of IEEE Global Telecommunications Conference*, pp.1-5, November 2006.
- [29] D.Niyato and E.Hossain, "A Queuing-Theoretic Optimization-Based Model for Radio Resource Management in IEEE 802.16 Broadband Wireless Networks", *IEEE Transactions on Computers*, vol.55, no.11, pp. 1473-1488, November 2006.
- [30] V.Singh and V.Sharma, "Efficient and fair scheduling of uplink and downlink in IEEE 802.16 OFDMA networks", *Proceedings of IEEE Wireless Communications and Networking Conference*, pp.984-990, September 2006.
- [31] S.Kim and I.Yeom, "TCP-aware Uplink Scheduling for IEEE 802.16", *IEEE Communications Letters*, pp.146-148, February 2007.
- [32] C.Mckillen, S.Sezer and X.Xang, "High performance service-time-stamp computation for WFQ IP packet scheduling", *Proceedings of IEEE Annual Symposium on Emerging VLSI Technologies and Architectures*, pp.65-70, March 2006.
- [33] J.Xu and R.Lipton, "On Fundamental Tradeoffs between Delay Bounds and Computational Complexity in Packet Scheduling Algorithms", *Proceedings of the 2002 SIGCOMM Conference*, pp.279-292, October 2002.

- [34] H.Fattah and C.Leung, "An overview of scheduling algorithms in wireless multimedia networks", *Proceedings of IEEE Wireless Communications*, pp.76-83, October 2002.
- [35] K.Chen, A.Liu and L.Lee, "A multiprocessor real-time process scheduling method", *Proceedings of the fifth International Symposium on Multimedia Software Engineering*, pp.29-36, December 2003.
- [36] The Network Simulator-NS-2, <http://www.isi.edu/nsnam/ns>.
- [37] J.Chen, C.Wang, F.Tsai, C.Chang, S.Liu, J.Guo, W.Lien, J.Sum, and C.Hung, "The design and implementation of WiMAX module for ns-2 simulator" *Proceedings of the 2006 Workshop on Ns-2: the IP Network Simulator*, 5 pp., October 2006.
- [38] B.Kim, Y.Hur, "Application Traffic Model for WiMAX Simulation", *POSDATA, Ltd*, April 2007.
- [39] A.Tee and F.Khan, "Comments and suggestions on open issues in the IEEE 802.20 evaluation criteria document", *IEEE 802.20 Working Group on Mobile Broadband Wireless Access*, November 2004.
- [40] M.Dianati, X.Shen and S.Naik, "A new fairness index for radio resource allocation in wireless networks", *Proceedings of IEEE Wireless Communications and Networking Conference*, pp.712-717, March 2005.
- [41] D.Altman, D.Machin, T.Bryant and M.Gardner, "Statistics With Confidence: Confidence Intervals and Statistical Guidelines", *British Medical Journal*, 140 pp., June 2000.
- [42] H.Shi and H.Sethu, "An Evaluation of Timestamp-Based Packet Schedulers Using a Novel Measure of Instantaneous Fairness", *Proceedings of International Performance, Computing and Communications Conference*, pp.443-450, April 2003.
- [43] Y.Ito, S.Tasaka and Y.Ishibashi, "Variably Weighted Round Robin Queuing for Core IP Routers", *Proceedings of International Performance, Computing and Communications Conference*, pp.159-166, April 2002.
- [44] D.Niyato and E.Hossain, "Queue-Aware Uplink Bandwidth Allocation and Rate Control for Polling Service in IEEE 802.16 Broadband Wireless Networks", *IEEE Transactions on Mobile Computing*, vol.5, no.6, pp.668-679, June 2006.
- [45] H.Cho, M.Fadali and L.Hyunjeong, "Dynamic queue scheduling using fuzzy systems for internet routing", *Proceedings of 14th International Conference on Fuzzy Systems*, pp.471-476, May 2005.

- [46] M.Horng, W.Lee, K.Lee and Y.Kuo, "An adaptive approach to weighted fair queue with QoS enhanced on IP network", *Proceedings of IEEE Region 10 International Conference on Electrical and Electronic Technology*, pp.181-186, August 2001.
- [47] H.Wang, C.Shen and K.Shin, "Adaptive-Weighted Packet Scheduling for Premium Service", *Proceedings of IEEE International Conference on Communications*, pp.1846-1850, June 2001.
- [48] S. Blake, "An architecture for Differentiated Services", RFC No. 2475, *Internet Engineering Task Force*, December 1998.

Appendix A – Intra-class fairness (Jain’s index)

Following are the results obtained from measuring intra-class fairness using Jain’s index.

Our discussion in chapter 4 uses Min-max index as it is more sensitive to unfairness and service degradation.

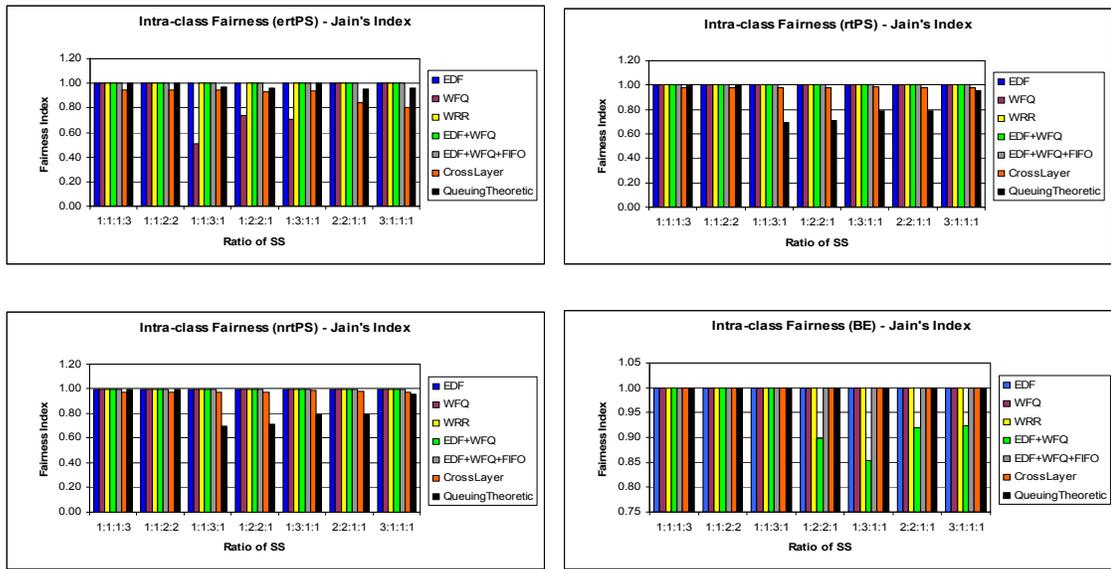


Figure A.1: The effect of SS ratio: Intra-class fairness

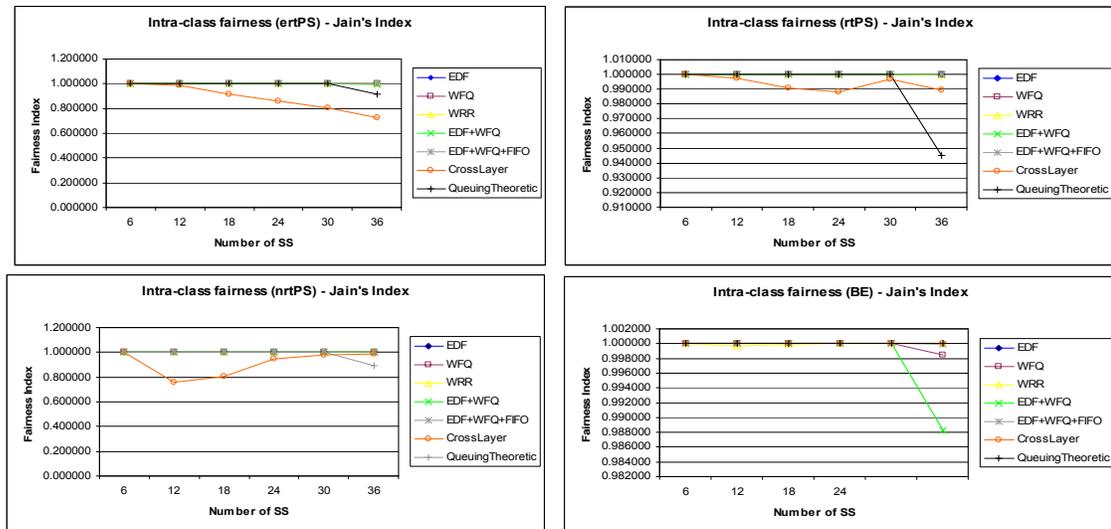


Figure A.2: The effect of uplink burst preamble: Intra-class fairness (Light load)

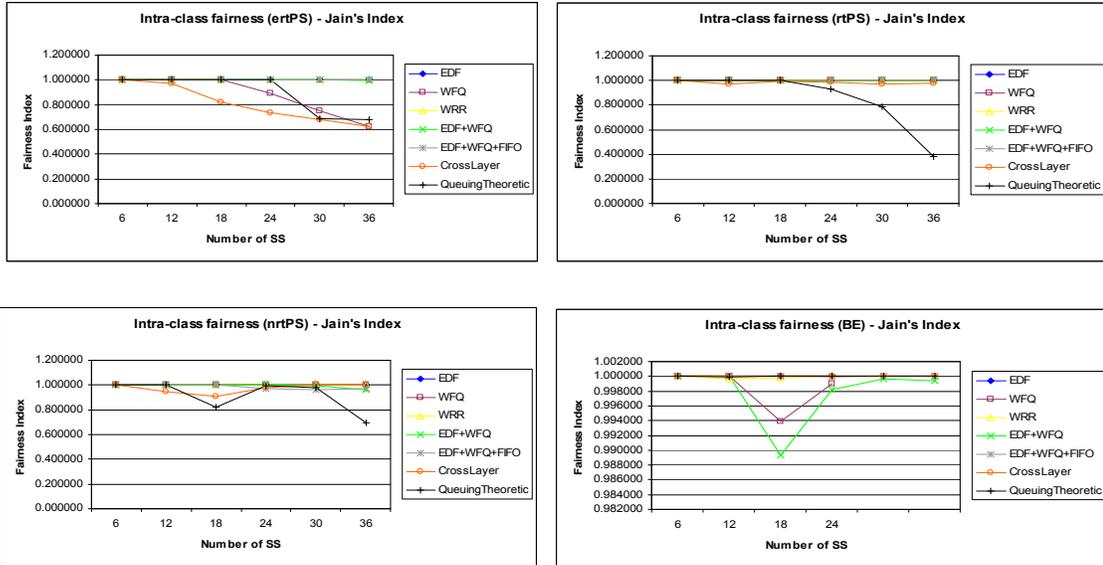


Figure A.3: The effect of uplink burst preamble: Intra-class fairness (Heavy load)