A Framework for Cross-Layer QoS-Aware Radio Resource Management in Mobile WiMAX Systems

PhD Thesis

By

IHSAN SHAHWAN

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Prof. Mohamed A. Ali

Date

Chair of Examining Committee

Dean Mumtaz Kassir

Date

Executive Officer

Prof. Samir Ahmed, EE Dept, CCNY, CUNY

Prof. Roger Dorsinville, EE Dept, CCNY, CUNY

Dr. Andrew Wallace, AT & T

Prof. Tarek N. Saadawi, EE Dept, CCNY, CUNY

Supervisory Committee

THE CITY UNIVERSITY OF NEW YORK

Abstract

A Framework for Cross-Layer: QoS-Aware Radio Resource Management in Mobile WiMAX Systems.

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IHSAN SHAHWAN

Adviser: Professor Mohamed Ali

The growing demand for mobile Internet and advanced wireless multimedia services and applications has accelerated the development and deployment of new broadband wireless access technologies including fourth-generation (4G) mobile WiMAX and cellular Long-Term Evolution (LTE). These technologies are capable of delivering up to 100 Mb/s speed. In this thesis, mobile WiMAX is chosen as representative candidate for 4G technology. WiMAX (Worldwide Interoperability for Microwave Access) is a rapidly growing broadband wireless access (BWA) technology based on IEEE 802.16 suite of standards. WiMAX is generally available in two versions: fixed and mobile. Fixed WiMAX, which is based on the IEEE 802.16-2004 standard, is ideally suited for delivering wireless, last-mile access for fixed broadband services.

Mobile WiMAX, which is based on the IEEE 802.16-2005 standard, supports both fixed and mobile applications while offering users improved performance, capacity, and mobility. The Mobile WiMAX air interface supports several new key features that distinguish it from other

metropolitan area wireless access technologies including utilizing Orthogonal Frequency Division Multiple Access (OFDMA) as the radio access solution to mitigate the effects of multipath fading, the use of multiple-input multiple-output (MIMO) antenna techniques, and the support of several adaptive modulation and coding schemes (MCSs) such as Binary Phase Shift Keying (BPSK), Quadrature Phase Shift Keying (QPSK) and Quadrature Amplitude Modulation (QAM). Radio Resource Management (RRM) techniques such as Admission Control (AC), Packet Scheduler (PS), Dynamic Bandwidth Allocation (DBA), and subcarrier allocation and mapping (SAM), are essential for supporting differentiated and guaranteed Quality of Service (QoS) as well as to ensure that scarce radio resources are efficiently utilized in wireless networks. Though the Mobile WiMAX standard provides the preliminary specifications for RRM techniques including QoS requirements and signaling mechanisms, however, the algorithms/schemes that manage the radio resources including AC, PS, DBA, and SAM, are not defined in the standards. These are still open research topics and are left for vendor implementation. The emerging 4G IP-based multimedia services and applications require efficient RRM schemes that can collectively support prioritization, high throughput, fairness, and, above all, ensure reliable end-to-end differentiated QoS delivery. Most of the RRM algorithms reported in the literature to date, however, fall short of providing a comprehensive and scalable RRM framework that enables carrier-class QoS support for both the existing and emerging wide range of multimedia services and applications required by the 4Gmobile users. Consequently, new generations of RRM techniques that can holistically address all of these concerns are required. This is the focus of this thesis.

Preface and Acknowledgments

This work would not have been possible without the support of Allah first and then the guidance of my supervisor professor M. Ali and his patience with me.

I would like to thank my mother for her love, support and encouragement. She has been always the inspirer throughout my life; words alone can never express my appreciation to my great mother.

I would like acknowledge professor Dorsinville who is the chairman of our EE Department for his continuous support during my study period. I would like to thank the committee who gave their time and effort in reading and providing me with their comments and feedback which helped me to improve the quality of this.

The inspiration from my little daughter and the support of my colleagues, whom I had technical discussion in various subjects, helped me to overcome many challenges that I have faced.

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To my mother Sayda and the soul of my father Mohamed Borhom and My lovely and beautiful daughter Samar

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Chapter 1

1.1 Introduction

WiMAX and LTE [1], [2], [3], [4], [5], [6] and [7] are two different technologies that will eventually be used to achieve data speeds of up to 1 Gb/s. These technologies have the potential to replace wired broadband connections with wireless, and enable services such as mobile TV, HDTV, and video conferencing without the need for a fixed-line or dish in the home. These technologies will also be capable of supporting a wide range of emerging and unforeseen applications currently regarded as too bandwidth-intensive to be delivered using existing mobile technologies. In this thesis, we have chosen mobile WiMAX as representative candidate for 4G technology.

WiMAX (Worldwide Interoperability for Microwave Access) is a rapidly growing broadband wireless access (BWA) technology based on IEEE 802.16 suite of standards [5-7]. WiMAX is generally available in two versions: fixed and mobile. Fixed WiMAX, which is based on the IEEE 802.16-2004 standard, is ideally suited for delivering wireless, last-mile access for fixed broadband services. It is similar to DSL or cable modem service. Theoretically, a WiMAX base station (BS) can provide broadband wireless access in range up to 30 miles (50 km’s) for fixed stations with a maximum data rate of up to 70 Mbps. Mobile WiMAX, which is based on the IEEE 802.16-2005 standard, supports both fixed and mobile applications while offering users improved performance, capacity, and mobility.

To support mobility and enhanced quality of service (QoS), the IEEE 802.16e-2005 standard (Mobile WiMAX) was approved as amendment to the 802.16-2004 standards. Mobile WiMAX was the first BWA solution that enabled convergence of mobile and fixed broadband networks through a common wide area broadband radio access technology and flexible network architecture.

Adaptive modulation and coding (AMC) is one of the advanced PHY layer features that was introduced with mobile WiMAX in order to cope with both the time and location varying nature of the wireless channel characteristic. To support a reliable transmission, the mobile WiMAX standard supports several modulation and coding schemes (MCSs) including Binary Phase Shift Keying (BPSK), Quadrature Phase Shift Keying (QPSK) and Quadrature Amplitude Modulation (QAM). The MCS is selected based on the channel condition, i.e., the SNR (Signal to Noise Ratio) reported by the MSs to the BS receiver via the UL-channel quality indictor (CQI) channel.

AMC enables the WiMAX system to select the most appropriate MCS depending on the channel condition. For instance, an MS close to the BS, which typically has a high SINR, is assigned an efficient higher order modulation scheme with low coding redundancy, e.g., 64-QAM (6 bits/symbol). On the contrary, an MS located at the cell boundary (far away from the BS), which typically has much lower SNR, is assigned a more robust lower order modulation scheme (less efficient), e., g., BPSK (1 bit/symbol). Thus, users with higher channel quality receive higher throughput in bits per second compared to those with lower channel quality. While AMC ensures maintaining reliable communications with users that have bad channel quality, this is accomplished, however, at the expense of reducing their throughput.

Since January 2007, the IEEE 802.16 working group has been developing a new amendment of the IEEE 802.16 standard (IEEE 802.16m) as an advanced air interface to meet the requirements of the International Telecommunication Union-Radio communication/International mobile Telecommunication (ITU-R/IMT)-advanced for 4G systems, as well as for the next-generation (NG) mobile network operators [8]. The NG mobile WiMAX will be capable of over-the-air data transfer rates in excess of 1Gb/s and of supporting a wide range of existing and emerging IP-based multimedia services and applications, while still maintaining full backward compatibility with the existing mobile WiMAX systems.

LTE and WiMAX have many features and functionalities in common, e., g., both are designed to move data rather than voice, and both are all IP-based networks based on OFDMA technology. Thus, most of the work presented in this thesis including the overall proposed framework for Radio Resource Management (RRM) in Mobile WiMAX is also applicable to LTE, provided that the detailed differences in the two standards are taking into account. While both 4G technologies share numerous of the salient features, however, they also still have plenty of differences. The crucial difference is that, unlike WiMAX, which requires a new network to be built, LTE runs on an evolution of the existing UMTS infrastructure already used by over 80 per cent of mobile subscribers globally [9].This means that even though development and deployment of the LTE standard has lagged Mobile WiMAX, it has a crucial incumbent advantage.

So which technology will ultimately prevail? It is arguable that LTE is more ‘risk-free' than WiMAX because it will run on an evolution of existing mobile infrastructure. However, if WiMAX is interworked with the already well-established 3G cellular networks such as UMTS or CDMA2000, it is almost certain that WiMAX will form an important part of the future 4G networks. This has already been addressed by the evolving WiMAX network specifications, which has added interworking with 3G systems and IP multimedia subsystems (IMS). Despite their differences in origin and current availability, the two competing 4G technologies may grow closer with time, especially as newer iterations on the standard emerge.

1.2 Thesis Motivation

The fundamental premise of Mobile WiMAX MAC architecture is the support of a wide range of multimedia services and applications (data, voice, and video) along with their diverse QoS requirements. The IEEE 802.16e standard supports five classes of service (CoSs) including Unsolicited Grant Service (UGS), extended real-time Polling Service (ertPS), real-time Polling Service (rtPS), non real-time Polling Service (nrtPS), and Best Effort (BE). Each of these services has a mandatory set of QoS parameters that must be included in the definition such as maximum sustained traffic rate, minimum reserved traffic rate, and maximum allowable delay. Radio Resource Management (RRM) techniques such as Admission Control (AC), Packet Scheduler (PS), Dynamic Bandwidth Allocation (DBA), and subcarrier allocation and mapping(SAM), are essential for supporting differentiated and guaranteed QoS as well as to ensure that scarce radio resources are efficiently utilized.

topics and are left for vendor implementation. Most of the RRM algorithms reported in the literature to date fall short of providing a comprehensive and scalable RRM framework that enables QoS support for Different CoSs.

The problem is further exacerbated particularly in the cases where QoS requirement of constant bit rate (CBR) applications such as streaming video and audio mandates maintaining a fixed data rate per connection in the network, regardless of the mobile user’s location and channel conditions.

Considerable research efforts have focused on RRM and QoS support for WiMAX networks; however, several key critical issues that may significantly impact the performance of the RRM schemes, and hence the overall performance of WiMAX systems, have still not been fully explored and/or addressed, including:

Most of the existing RRM schemes including PS and DBA algorithms utilize the simple OFDM-based scheduler, in which multiple subscribers use a time division multiple access (TDMA) to share the media. The scheduler in this case is fairly simple because the resource allocation unit is available only in the time domain (time Slots), and the entire frequency channel is given to each user. The scheduling decision is simple to decide what time Slots should be allocated to each subscriber. On the other hand, both time and frequency domains need to be considered for OFDMA-based mobile WiMAX. In this case, the resource allocation unit is a two-dimension time-frequency Slot. The OFDMA scheduler is the most complex one because each user can receive some portions of the allocation for the combination of time and frequency so that the channel capacity is efficiently utilized [10].

Most of the relatively few reported RRM algorithms that utilized the OFDMA-based two-dimension Slot scheduling have assumed resource allocation strategy that is based on non-adaptive static subcarrier assignment (SSA), in which a group of subcarriers forming different Slots are assigned to each user regardless of channel frequency responses. In this case, each user effectively becomes a single user who is independent of all other users. On the other hand, channel frequency responses are different at different frequencies or for different users (channel characteristics for different users are almost mutually independent in multiuser environments) [11] [12]. The subcarriers experiencing deep fading for one user may not be in a deep fade for other users; therefore, each subcarrier could be in a good condition for some users in a multiuser OFDMA-based network. Thus, frequency-selective scheduling or dynamic subcarrier assignment (DSA) schemes dynamically assign subcarriers to users having the best instantaneous channel conditions (i.e., frequency response) by exploiting the multiuser diversity. The frequency-selective scheduling can significantly enhance system capacity.

Most of the work reported in the literature to examine the impact of AMC schemes on the overall performance of mobile WiMAX systems have assessed the performance impact of AMC schemes in a static approach, where users are divided into few fixed subgroups, each is pre-assigned to one of the available MCS zones that the system support. Thus, user mobility in the context of crossing multiple MCS zones within the same cell, which can significantly impact the AMC-based WiMAX system performance and stability, is not taken into account. As a mobile user moves away from the closeness of a BS towards the cell boundary, he experiences different gradually decreasing SNR values; each is associated with proportional gradual decrease in the throughput, as he traverses from one MCS zone to another. In this case, the WiMAX system must dynamically keep shifting to a lower order and more robust MCS to maintain the link stability. As the mobile user approaches the cell boundary and enters the BPSK zone, his throughput falls drastically. Thus, AMC by itself is not appropriate for constant bit rate (CBR) applications such as streaming video and audio. In other words, the concept of SNR-bandwidth trade-off invoked via utilizing AMC in WiMAX systems is not sufficient enough to maintain both the CBR connection quality and link stability. Channel quality is always compromised with channel capacity. The simplest approach to address this problem is to pre-assign a fixed robust lower order MCS, e. g., BPSK/QPSK, that ensures providing acceptable BER as well as nearly fixed data rate throughout the entire journey. This approach, however, leads to significant wastage of scarce network capacity as these MCSs typically have poor spectral efficiency (1 b/s/Hz).

Most of the recent WiMAX RRM scheduling algorithms reported in the literature typically target one or two of the classic performance metric, e.g., fairness, differentiated QoS, or system throughput maximization. Fairness and throughput are two crucial issues in resource allocation for wireless networks and, typically, there is always an effective tradeoff between throughput and fairness. The fundamental premise of algorithms that balance fairness and throughput is to achieve a good overall throughput while avoiding starvation of the subscribers that are not using spectrally efficient modulations. Proportional Fairness is a representative example of this category. Even though these algorithms perform reasonably well at balancing throughput versus fairness over the air-link, prioritization is not embedded in these algorithms [13]. These typical RRM scheduling algorithms often manage to improve performance of the networks via optimizing one of these performance metrics [14]. However, the emerging 4G IP-based multimedia services and applications require efficient RRM schemes that can collectively support prioritization, high throughput, fairness, and, above all, ensure reliable end-to-end differentiated QoS delivery. Consequently, a new generation of RRM schemes that can holistically address all of the aforementioned shortcomings and concerns are required. This is the focus of this thesis.

1.3 Thesis Statement & Contribution

This thesis addresses the important problem of Radio Resource Management (RRM) and Quality of Service (QoS) support for the emerging 4G broadband wireless access technologies. Specifically, this thesis examines the technical feasibility and assesses the performance analysis for implementing an integrated framework for uplink (UL) RRM techniques in mobile WiMAX systems that holistically address the aforementioned shortcomings and concerns. The ultimate objective of the proposed framework is to provide optimum or close to optimum UL scheduling and resource allocation strategies that meet the typical mobile WiMAX mandatory set of QoS parameters and requirements associated with each CoS including CBR streaming applications; while concurrently striking a balance between maintaining nearly a constant throughput per every CBR streaming connection in the network, fairness among all mobile users irrespective of their channel conditions and locations (throughput fairness), and the overall system capacity.

To achieve this objective, several novel ULRRM functional modules including Admission Control (AC), Packet Scheduler (PS), Dynamic Bandwidth Allocation (DBA), and subcarrier allocation and mapping (SAM) are developed for mobile WiMAX. The notion of an integrated RRM framework is realized via introducing two critical mechanisms. First, the functionalities of all RRM modules (AC, PS, and DBA) at the MAC layer are first integrated. The QoS-aware scheduling decision (obtained from the unified RRM scheduling modules) at the MAC layer is then integrated with the channel state information and corresponding opportunistic AMC schemes as well as the SAM at the PHY layer, to get the final optimum (or near optimum) scheduling and resource allocation decisions.

This is a complex and challenging optimization task that requires fast and periodic intra-communications among the RRM scheduling modules (AC, PS, and DBA) at the MAC layer as well as inter-communications between these scheduling modules at the MAC layer and all relevant transmission modules at the PHY layer. This task also requires frequent and periodic updates (each cycle) of numerous system parameters that collectively impact the overall system performance. This is implemented with a moderate increase in the UL CQI (channel quality indicator) overhead. Thus, the proposed integrated RRM frame work supports a fully integrated PHY-MAC layer strategy, which is the key for obtaining optimum (or near optimum) scheduling and resource allocation decisions each cycle.

The implementation of the proposed RRM framework proceeded in three sequential phases. Each succeeding phase builds upon previous phase(s) in order to address its shortcomings and enhance its performance. All RRM PS and DBA algorithms and schemes presented in the three phases utilize the OFDMA-based two-dimension time-frequency Slot assignment. The resource allocation strategy utilized by RRM PS and DBA algorithms presented in the first and second phases is based on non-adaptive static subcarrier assignment. On the other hand, the resource allocation strategy utilized in the third phase is based on dynamic subcarrier assignment.

In the first phase, a set of integrated QoS-aware AC, PS, and DBA algorithms and schemes are developed. Specifically, we present and devise a 3-level hierarchical algorithm (3LHA) for UL scheduling in Mobile WiMAX that targets specific requirements of each class of service while enforcing opportunistic cross-layer approach in order to meet QoS requirements and maximize the overall system throughput. In contrast to mainstream hierarchical scheduling algorithms that utilize the typical strict priority algorithm at the first level of the hierarchy to schedule between different classes of services, the proposed algorithm utilizes, for the first time to the best of our knowledge, MCS zones as the first level priority metric of the hierarchy. Typical strict priority is then used at the second level in the hierarchy to schedule between different classes of service within each MCS zone. Several different intra-class scheduling algorithms are then used at the third level to schedule within each class of service for each MCS zone; Earliest Deadline First (EDF) is used for rtPS, Weighted Fair Queuing (WFQ) for nrtPS, and Round Robin(RR) for BE.

Since certain classes of service have higher priority than others, starvation control has to be considered, specifically for hierarchical scheduling algorithms. To address this issue, we also devise a QoS-aware AC mechanism at the BS along with token bucket-based traffic policing at the SSs to avoid starvation of lower priority nrtPS and BE services. The AC maintains the QoS of in-progress WiMAX connections by admitting a new connection only if all the existing and new connections can be guaranteed their QoS requirements. The proposed PS and DBA algorithms are then integrated with the AC mechanism to support effective QoS provisioning.

The set of integrated RRM AC, PS, and DBA algorithms and schemes presented in the first phase adequately target two main performance metrics: efficient differentiated QoS support and maximizing the system throughput. This is achieved, however, at the expense of sacrificing fairness among all mobile users. In the second phase, we develop a set of integrated RRM PS and DBA algorithms that not only adequately address the shortcoming (fairness problem) of the first phase but also address one of the important problems that has received little/no attention in the literature (problem 3 listed in section I.1 above); “the impact of users mobility (in the context of crossing multiple different MCS zones within the same cell) on the overall performance of AMC-based mobile WiMAX systems that specifically support CBR streaming applications.

Specifically, taking into account the speed of mobile users, their varying channel conditions, and the AMC schemes at the PHY layer, we propose and devise a cross-layer UL scheduling algorithm in OFDMA-based mobile system that is specifically tailored to maintain not only almost a fixed data rate but also a limited maximum delay variance per every connection in the network. To keep the main objective of this phase in focus, we only consider users mobility within one cell as they get to cross all different possible MCS zones within the cell. To consider the most stringent QoS requirements scenario, the algorithm generalizes the concept of CBR applications by assuming that all CoSs supported by the IEEE 802.16e standard strive to maintain a fixed data rate throughout the entire life of the connection. We assume that this is a requirement included in the service level agreement between the customers and the service provider. Thus, the algorithm strives to maintain almost constant throughput per every WiMAX connection in the network (including rtPS, nrtPS, and BE connections) while still meeting the typical mobile WiMAX mandatory set of QoS parameters and requirements associated with each CoS.

The proposed algorithm strikes a balance between maintaining nearly a constant throughput per every connection in the network, fairness among all mobile users irrespective of their channel conditions and locations (throughput fairness), and the overall system capacity, while still meeting differentiated QoS requirements for each CoS. To maintain nearly a constant throughput per every connection in the network with the least possible resources, a joint cross-layer optimization of the bandwidth resources (Slots) and MCS assigned periodically to the connection is performed on a frame by frame basis such that the product of the spectral efficiency of the assigned MCS and the bandwidth always yields the required fixed throughput. This is implemented, as will be shown later, via introducing the notion of Zone Balance Ratio (ZBR).

Finally, the work presented in the third phase addresses one common shortcoming that is exhibited by both the RRM PS and DBA algorithms and schemes presented in the first and second phases; namely, the resource allocation strategy utilized in both the first and second phases is based on non-adaptive static subcarrier assignment (SSA). The resource allocation strategy utilized in the third phase, however, is based on adaptive dynamic subcarrier assignment (DSA). Thus, the same RRM PS and DBA algorithms that have been presented in the first and second phases are reused again in the third phase, however, using the developed DSA resource allocation strategy rather than SSA. Therefore, the results presented in the third phase reflect the complete envisioned integrated RRM framework for Mobile WiMAX.

The results presented in this thesis are obtained through extensive computer simulations and modeling using substantial system simulator development, which is carried out during the course of this work. This includes both mathematical modeling considerations as well as software design, implementation, and testing. The system parameters used here to assess the overall performance of the proposed RRM framework are identical to the performance evaluation parameters specified in Mobile WiMAX system evaluation documents [15] [16] and WiMAX profiles.

1.4 Novelty and Publications

The results of this study have been published in the following articles:

- **Ihsan Shahwan**, M. Ali, Muath Obaidat, R. Dorsinville , "Stable Move Algorithm (SMA) for BW Allocation in Wireless Mobile WiMAX Networks802.16e", IEEE Wireless Telecommunications Symposium(WTS),New York city, New York ,April 2011.
- **Ihsan Shahwan**, M. Ali, Muath Obaidat, R. Dorsinville, "WiMAX: Cross Layer Bandwidth Allocation Strict Priority Based Adaptive modulation and Coding", IEEE International Conference on Communications and Information Technology (ICCIT), Aqaba, Jordan, March 2011.

In addition, the collaborative work on resource allocation in wireless mobile ad hoc networks which have been published in the following articles.

- Muath Obaidat, M. Ali**, Ihsan Shahwan**" QoS-Aware Multipath Routing Protocol for Delay Sensitive Applications in MANETs: A Cross-Layer Approach" ICETE 2011 Wireless Information Networks and Systems, Seville, Spain, July 2011.
- Muath Obaidat, M. Ali, **Ihsan Shahwan**" A Novel Multipath Routing Protocol for MANETs" Wireless Communications Networking and Mobile Computing, WICOM, Wuhan, China Sept. 2011.

1.5 Thesis Outline

This thesis is organized as follows:

- **Chapter 2:** ***Background and Literature Review*** **-** This chapter gives the general WiMAX back ground technology, deployment and importance of WiMAX as a long term broad band access provider and to solve the last mile problem. It also spots the light on MAC and PHY layer in WiMAX and the time frequency grid resource allocation, different CoS and QoS supported by WiMAX. In addition related work and the previous proposed work to solve scheduling in WiMAX have been presented.

- **Chapter 3:** ***Framework: Modeling and Mobility Scenarios*** **-** This chapter list all simulation parameters used; it also proposes a novel wireless mobility model used to test and implement the proposed work.

- **Chapter 4 :** ***A Novel Cross layer Three Level Hierarchal Algorithm QoS Provisioning*** **(3LHA) –** The chapter introduces the three level hierarchal algorithm which takes into consideration the connection class of service and its QoS requirements at the same time the user channel condition that this connection belongs to.

- **Chapter 5:** ***A Novel Cross layer Modified by Zone Balance Ratio QoS Provisioning Algorithm (ZBR-3LHA)***

The chapter introduces and implements the zone balance ratio function (ZBR) to maintain fairness and to adapt to different AMC schemes in different modulation zones. Connection maximum threshold is defined in which the connection maximum number of granted Slots is determined.

- **Chapter 6:** ***A Novel Combined Cross Layer with Dynamic Slot Allocation (Fully integrated PHY-MAC)*** **-** This chapter presents a fully integrated scheduler for QoS provisioning, it combines the aforementioned algorithm introduced in chapter 5 at the PHY layer which was introduced in chapter 4 with the algorithm located in the MAC layer to exploit multi users diversity and to achieve the best system performance. Moreover, modeling noise, wireless channel, calculating the channel impulse response, implementing the transmission guide matrix to guide upper layers for resource allocation have been discussed. A comprehensive description of a novel dynamic Slot allocation with low complexity channel impulse response estimation for WiMAX 802.16e has been presented.

- **Chapter 7:** ***Conclusions and Future Work*** **–** Conclusions of the proposed work and enhancements through future work has been laid in this chapter

- **Appendix A:** Practical Example for Calculating the Average SNR in DSSA.
- **Appendix B:** Practical Example shows DSSA and Frame Sequence Number.
- **Appendix C:** Pseudo Code for Cross Layer Hierarchal Scheduling.

Chapter 2

Back Ground and Literature Review

2.1 Overview of WiMAX Technology

In WiMAX standards Base station grant bandwidth in two modes, the first is Grant Per Subscriber (GPSS), in which the BS grants all the connections that belong to the same SS, and then the SS distribute the granted BW among its connections, and the second is Grant Per Connection (GPC), in this mode the SS receives the granted BW for a certain connection and consequently each connection has to request its own BW.

GPC is adopted in this work. The BS is responsible to dynamically maintains different QoS requirements for different connections belong to various users. In WiMAX Connection had to be established by the network through the call admission control (CAC) in which QoS is achieved.

data are transmitted via Orthogonal Frequency Division Multiplexing (OFDMA) [17] wherein data is mapped into Slots, each Slot is formed by one OFDM symbol and a certain number of frequency carriers named sub channel, before data sent to Physical layer of the wireless medium (PHY), data is encapsulated into a certain number of Slots on frame bases, WiMAX frame consists of limited number of those Ss for Uplink and downlink with variable rates, the rate of each Slot is a function in the adaptive modulation and coding (AMC) mechanism which will vary according to the physical medium conditions ,several of the AMC flavors are provided to maintain the tradeoff between the data rate and the robustness.

There are three modes of operations for WiMAX, the first is point to multi point (PMP), the second is point to point (PP) and the third is Mesh mode. In the first, BS connects to all SS, in the second two BS could connect to each other to extend the communication range while in the third mode each SS could work as an agent for his BS by forwarding the message to the next destination.

In WiMAX networks the decision of assigning resources for a certain SS is taking based on his location and the channel status. If the received signal strength dropped below a certain threshold due to wireless channel fluctuating or changing the user location from AMC zone to another the scheduler has to adapt to the change by assigning the proper AMC to defeat the new situation, changing the AMC while the call in session will have a massive impact on the connection rate. The reason for that is the (S)-bit rate assigned to SS who is located in BFSK zone can support just a fraction compared to the one who assigned to the SS located in the QAM-64 zone.

2.2 WiMAX Deployment and Basic Structure

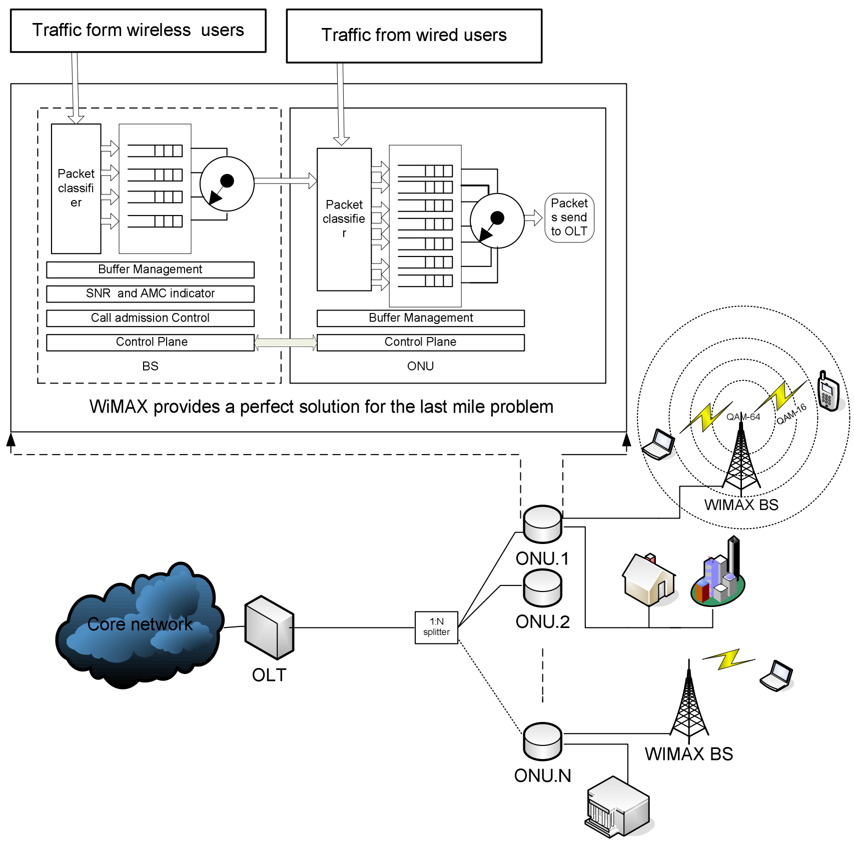

Figure 2-1 : IP-Based WiMAX Network Architecture

WiMAX provides a perfect solution for the last mile problem in which it can be deployed in areas without infrastructure, like country sides and suburban areas, WiMAX network consist of the BS located on elevated position connected to a group of subscribers (SS) on cellular structure principle in which each BS is responsible for SS located within the cell borders only. The technology has replaced some of existing telecommunications infrastructures. For example; in fixed wireless configuration it has replaced the traditional telephone copper wire networks, the cable TV's coaxial cable infrastructure. As mobile a mobile, WiMAX aims to replace cellular networks.

2.2.1 Cell structure and Grant per connection mode (GPC)

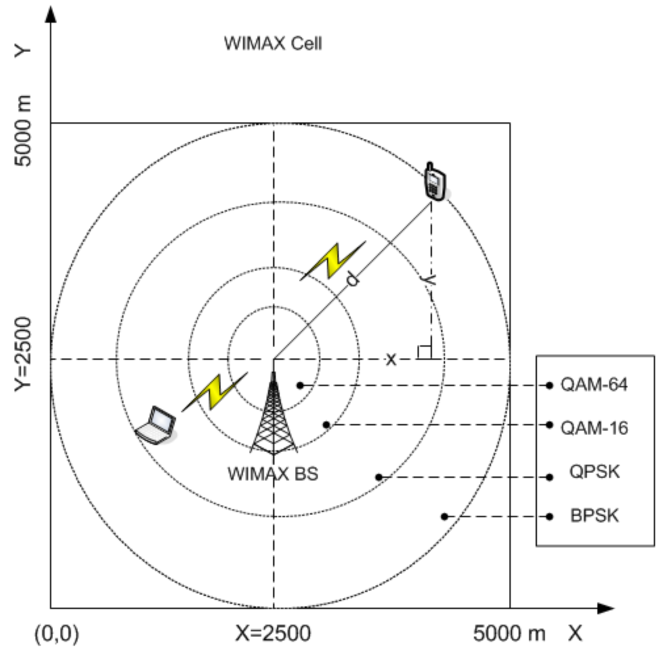

Figure 2-2 : BS structure and different QAM modulation Zones

Base station is represented by a tower located in elevated place at the center of circle shape which represents the total coverage of the cell. The main cycle is divided into four concentric cycles of coverage. The transmission range of radius (R = 2.5 km) is considered. The cell is divided into 4 QAM regions; each is represented by concentric circles with different radii. The coverage area for each AMC is decided up to minimum required SNR for each modulation scheme in the standard. As given in Table 3.2, the distance between the SS and the BS, SNR for each user, and the path loss are calculated as given in chapter 3.

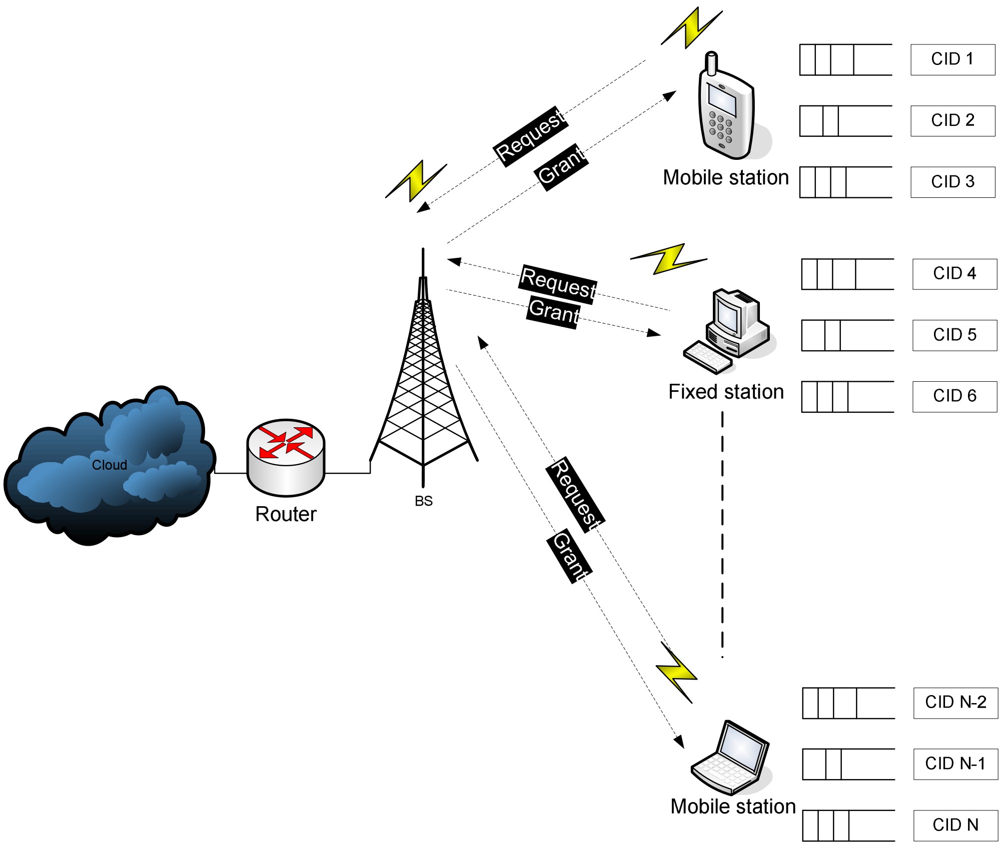

Figure 2-3 : Request-grant mechanism adopted is the Contention free (Polling) mode.

All connections generates traffic for each of the four different class of services rtPS, nrtPS and BE as shown in Table 3.1 sends their request to the BS and the BS grant radio resources taking into consideration to maintain QoS according to the implemented RRM.

2.3 Description of MAC layer in WiMAX

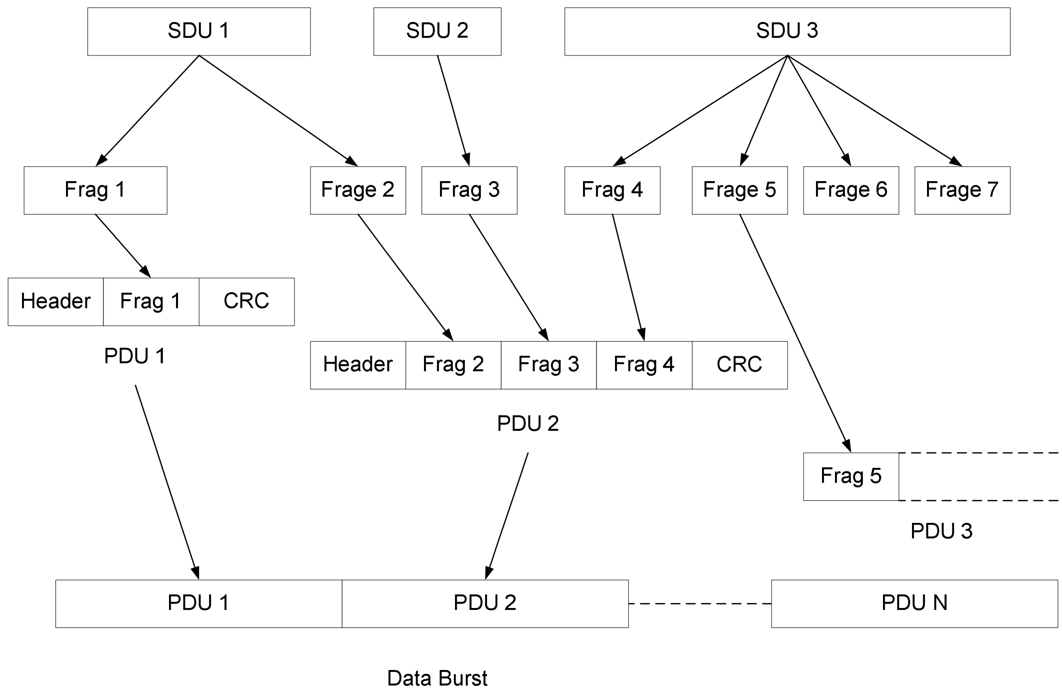

Figure 2-4 : SDUs Concatenation and Fragmentation

The IEEE 802.16E MAC-Layer is a connection oriented Network which means that each subscriber/connection must establish connection with the BS in order to get a band width. The MAC Layer can be classified into three sub-Layers, The First is Convergence Sub layer (CS) which interact with the upper routing Layer like ATM and IP, the second is the Common part Sub-Layer (CPS) which includes the MAC characteristics, finally the security Sub-layer (Sec-S) which responsible for the Authentication and encryption.

The MAC Layer takes data from upper Layer as MAC Service Data Units (M-SDU's) Protocol and convert them into MAC Protocol Data Units (M-PDU's) ready for transmission over the air.

WiMAX MAC supports variable length PDU's, multiple SDU's from higher layer may concatenated into a single PDU to save MAC header overhead, on the other hand a single huge SDU could be fragmented into smaller PDU's and sent over multiple frames.

2.4 Description of PHY Layer in WiMAX (OFDMA)

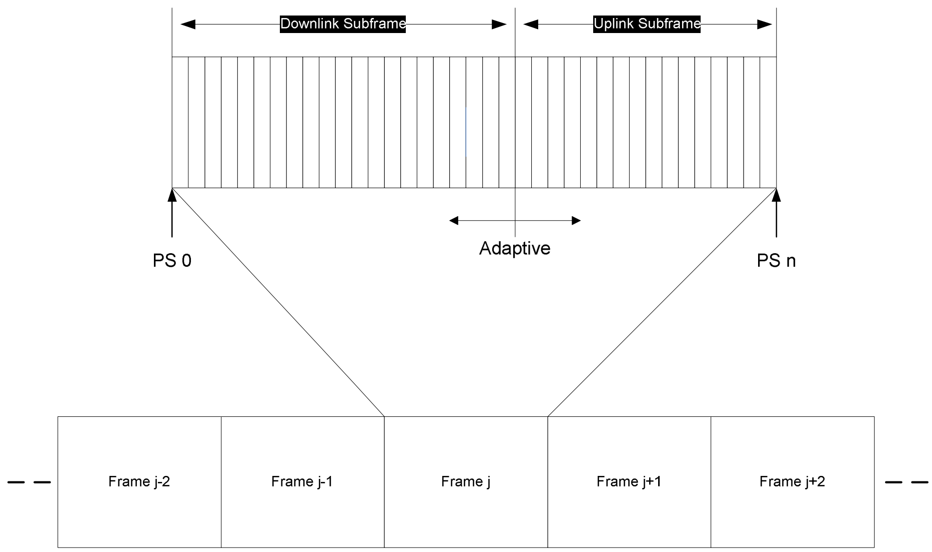

Figure 2-5 : TDD WiMAX Frame

The mobile WiMAX air interface utilizes Orthogonal Frequency Division Multiple Access (OFDMA), [18] as the radio access method to improve multipath performance in non-line-of-sight environments. OFDMA exploits the frequency diversity of the multipath channel by coding and interleaving the information across the sub-carriers prior to transmissions. The IEEE 802.16e-2005 air-interface supports both Time Division Duplex (TDD) and Frequency Division Duplex (FDD) modes; however, only TDD mode of operation is considered in this work. The TDD mode is preferred because it enables dynamic allocation of DL and UL resources to efficiently support DL/UL band width allocation and requires only one channel for both downlink and uplink. The frame structure under TDD mode includes the control messages UL-

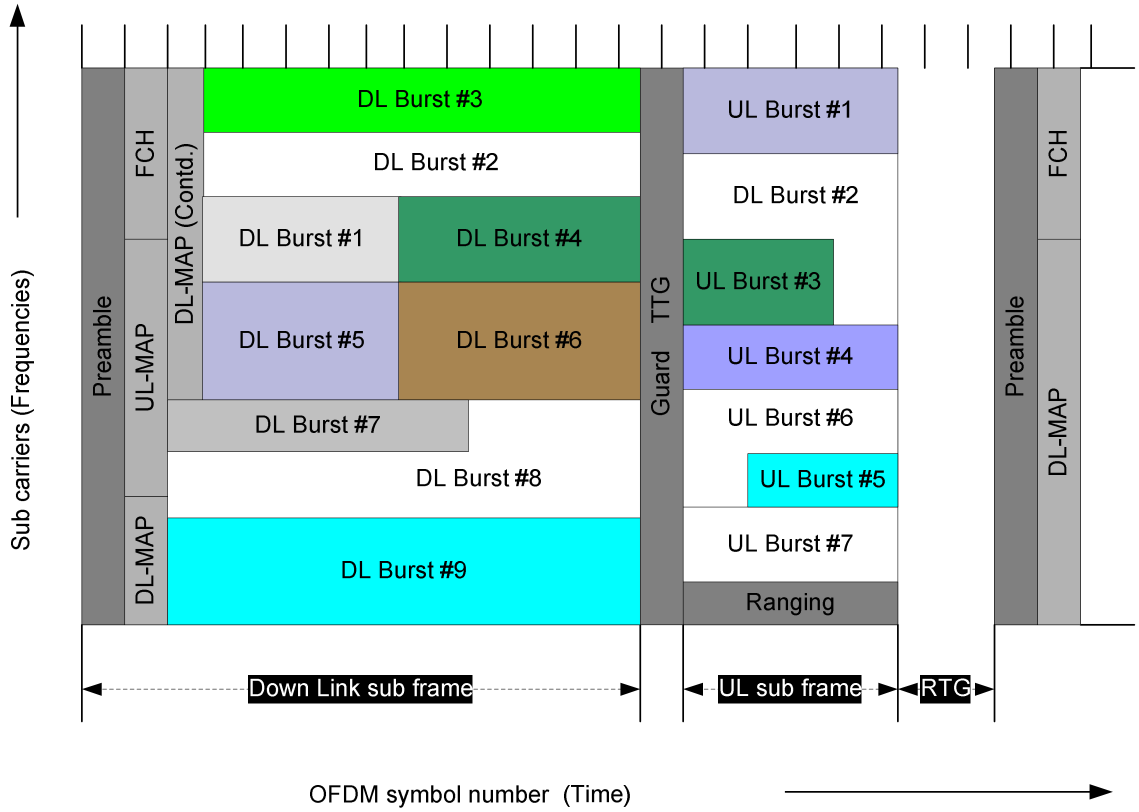

Figure 2-6 : A sample TDD frame structure for mobile WiMAX

MAP, DL-MAP and also includes the downlink and uplink data bursts whose scheduled time and frequency are determined by the bandwidth allocation algorithm and is obtained in the MAP messages. In an OFDM system, resources are available in the time domain by means of OFDM symbols and in the frequency domain by means of sub-carriers. The time and frequency resources can be organized into sub-channels for allocation to individual users. Active (data and pilot) sub-carriers are grouped into subsets of sub-carriers called sub- channels. The WIMAX OFDMA PHY [m8] supports sub-channelization in both DL and UL.

2.5 Radio Resource Elements and Time/Frequency Grid in WiMAX

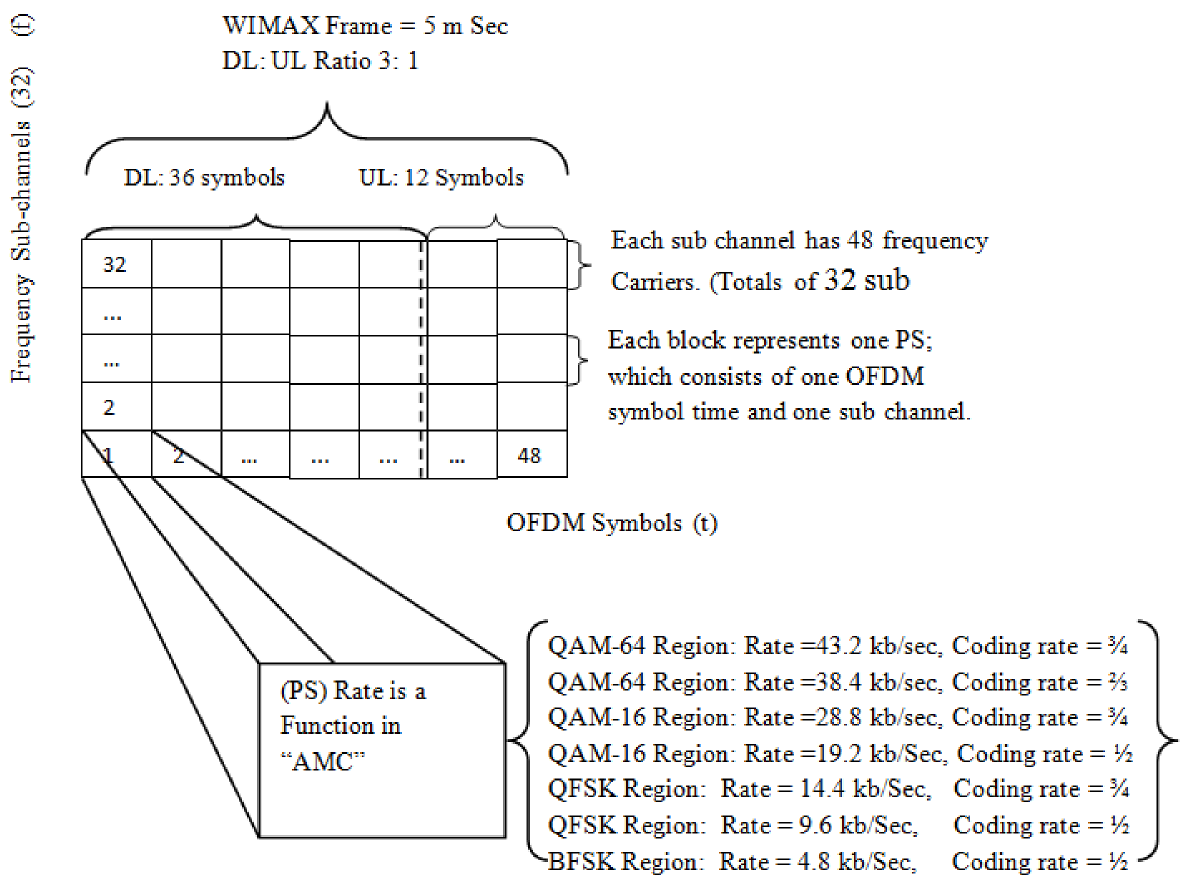

Figure 2-7 : Slot-Rate Vary according to user SNR, AMC and coding rate

In OFDMA the total channel band width is divided into many orthogonal narrowband sub channels. The serial high-rate data stream is converted into several parallel low-rate streams, each modulated on a different subcarriers, a group of those subcarriers will form the Slot which is the basic band width resource element could be granted for the connection.

The minimum frequency-time resource unit of sub-channelization is one Slot, which is equal to 48 data tones (sub-carriers) in the frequency domain and one OFDM symbol time in the time domain. In a system with 20 MHZ band width there are a total of 32 sub channels in the frequency domain and 48 OFDM symbols in the time domain, considering that each OFDM symbol time is 103µs. (Figure 2-7) The bit rate supported by each Slot when its granted to a subscriber (SS) varies based on his wireless channel condition, Signal to noise ratio and the adaptive modulation and coding scheme.

2.6 QoS Support (Service Classes) in WiMAX

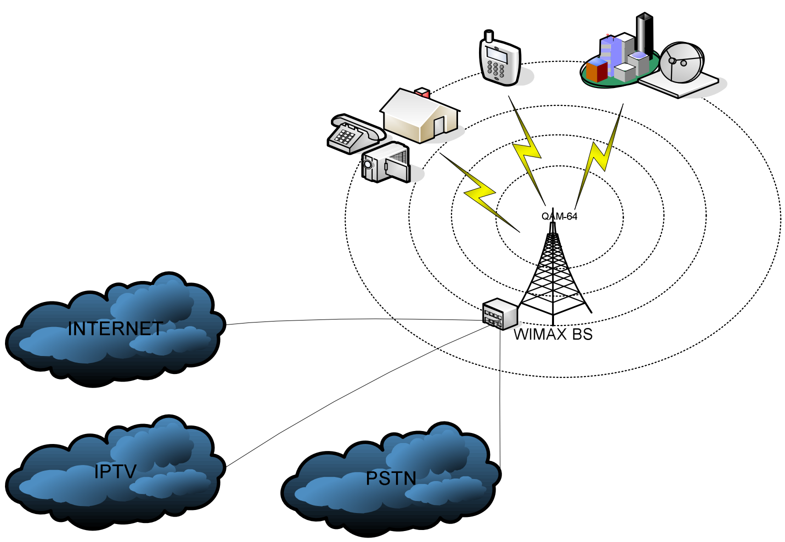

Figure 2-8 : WiMAX maintains QoS and Broad band wireless access in the Last mile

802.16e supports five different service classes, namely the Unsolicited Grant Service (UGS), Real-time Polling Service (rtPS), Non-Real-time Polling Service (nrtPS), Extended Real time polling Service (ErtPS) and the Best Effort (BE). Each service class classifies different data handling mechanisms to achieve service differentiation. A fixed amount of Slots is reserved for UGS class of service in each cycle for every connection because it has the highest priority among all classes, the rest of classes depend on the scheduler to gain transmission opportunity. For simplicity work in this thesis was performed under the assumption that the scheduling algorithm supports three classes of service: rtPS, nrtPS, and BE, but it can be easily extended to support any number of services. Each Mobile Station (MS) has three queues, one for each class of service. At each cycle (frame), MSs report their current queue sizes for each rtPS, nrtPS, and BE connection (admitted flow) in either bytes or the instantaneous data rate to the BS for bandwidth allocation for the next cycle. For rtPS services, the bandwidth request message also includes the arrival time of the head of the line (HOL) packet.

2.7 Overview of Uplink (BWA) and Radio Resource Management

Wireless Schedulers for WiMAX has been classified into two categories; channel-unaware schedulers and channel-aware schedulers.

2.7.1 Channel-unaware schedulers

Channel-unaware schedulers are not practical since the channel is considered an error free and the user channel state condition is not a part of the scheduling decision. In reality wireless links characteristics are subject change due to signal attenuation, fading, interference and noise. On the other hand Channel-unaware schedulers operate in two modes, Intra-class Scheduling: which is responsible for allocating the resources within the same class according to required QoS parameters; many examples for the afro mentioned types were proposed in the literature, like Weighted Round Robin (WRR) [19] and Deficit Round Robin (DRR) [20] are fair and simple despite the poor delay which fairly grant resources (certain number of bytes or packets) to all connections on one by one base. While Weighted Fair Queuing (WFQ) [21] and the variations of it, such as Worst Case Fair Queuing (WF2Q) [22], Self-Clock Faire Queuing (SCFQ) [23] have achieved better delay performance at the cost of high complexity and unfairness; Proportional fair (PF) scheduling algorithm [24] have been proposed to provide fairness among users but, it failed to maintain quality of service (QoS) between their connections [25]. In addition to the a fro mentioned algorithms; priority-based [26], [27], and [28], in which queues are assigned different priorities the highest is UGS queue ,next is rtPS, then nrtPS and finally is the best effort queue (BE) have been proposed for interclass scheduling in WiMAX networks.

2.7.2 Channel-Aware Schedulers

Lately, special attention has been given to cross layer optimization in which the BW allocation operates in both layers; typically PHY-level (“sub-channel allocation”) jointly with the MAC-layer (QoS and service differentiation). Those algorithms intend to exploit the multiuser diversity and channel fading. Bandwidth allocation tactics follows different strategies; maximizing the overall throughout, fairness, in addition to QoS grantee.

WiMAX is a connection oriented network so the schedule has to grantee QoS. To maximize the network throughput, schedule will prefer the user with the best channel condition and avoid allocating resources with user has a bad channel because the packet will be dropped any way.

Optimization of fairness criteria was introduced in the literature [29], [30]. Maximizing the overall network throughput has been proposed in [31], [32]. However wasn't applied for OFDMA as PHY Link layer while some has aimed to fined balance between efficiency and total system throughput [33], [34], [35] and [36]. The problem of a fairness scheduling for multicarrier systems was proposed in [37], [38], [39], [40], [41], [42] and [43]. The former assigns equal data rate for all connections by providing users with bad channel or located in low order modulation zone by more resources, As a consequence the recourses capacity for a users with good channel condition or have a high modulation order is not exploited. However the resource element used was frequency sub-carriers not OFDMA Slots.

None of the previous mentioned algorithms have considered the wireless channel state information (CSI) represented by the user SNR, Adaptive modulation and coding techniques (AMC) , connection class of service (CoS), the amount of requested band width and packet delay concurrently in the scheduling decision, and that's where our new proposed Radio resource management scheme comes to fill the missing piece in the puzzle by considering the main parameters in the decision of scheduling, this has shown significant improvement of the overall throughput of the whole network and in the average packet delay of different classes of service

Chapter 3

Framework: Modeling and Mobility Scenarios

3.1 Introduction

A simulation model using MATLAB and C++ has been developed. In order to have a realistic simulation environment similar to WiMAX deployments, the WiMAX system parameters used here are selected according to the recommended default values from the WiMAX Forum Mobile System Profile [44], [45] and [46] these parameters are listed in Table 3.3, In order to compare the proposed scheme to the bench mark scheme, a simulation model for both; fixed and the mobile environment were built. The simulator consists of four fundamental models: the system model, channel model, fixed model and mobility model.

3.2 System Model

A single Base station (BS) with a transmission range R = 2.5 km is considered. The cell is divided into 4 QAM regions; each is represented by concentric circles with different radii. The coverage area for each AMC is given in Table 3.2, the distance between the SS and the BS is calculated on frame bases using parameters from the mobility model as in section 3.5 and 3.6. The path loss and SNR are calculated for different users by the propagation model as in 3.3.

Initially five users are located in each of the four QAM zones. Every user generates traffic for each of the four different QoS connections UGS, rtPS, nrtPS and BE as shown in Table 3.1

3.3 Propagation Model

Cost 231 Hata [47] urban propagation model is used to predict the path loss in the cell, the main equation for path loss in dB is:

$$\mathrm{PL} = 46.3 + 33.9\ \log_{10}(\mathrm{f})\ - 13.82\log_{10}(\mathrm{h_b}) - \mathrm{ah_m} +\ (44.9 - 6.55\ \log_{10}(\mathrm{h_b}))\log_{10}\mathrm{d} + \mathrm{c_m} \qquad (3\text{-}1)$$

Where f is the frequency in MHz, d is the distance between SS and BS in km, and h_b is the SS antenna height above ground level in meters. Urban environment is considered for our simulation in which, c_m equal to 3 dB, $ah_m$is defined as [48] to be:$ah_m = 3.20\ (\log_{10}(11.75h_r)^2 - 4.97)$. Knowing the path loss, the power received can be determined based on the user's location as

$$P_{\text{received}} = P_{\text{transmitted}} - P_{\text{path Loss}} \quad (dBm) \qquad (3\text{-}2)$$

The SNR at the receiver is computed as:

$$\text{SNR} = P_{\text{received}} - P_{\text{Noise.}} \quad (dB) \qquad (3\text{-}3)$$

The simulator assigns values for the AMC region depending on the user's SNR;
Table 3.2 provides the relation between SNR and AMC scheme.

3.4 Channel Estimation and Modeling

Wireless channel is modeled as a multipath fading channel which is a source of originating the inter-symbol interference (ISI) in the received signal. To throw out ISI from the signal, detection Algorithms like Least-squares (LS) [49], [50] need to be implemented in the receiver.

Detectors need to know channel impulse response (CIR), which can be obtained by a separate channel estimator. Usually this happens by taking the advantage of an already known sequence of bits, called training sequence bits which are unique for each transmitter and its corresponding receiver. These training bits are transmitted each burst, the channel estimator estimates the CIR by utilize the known transmitted bits and the corresponding received signals.

3.4.1 Mean Least-squares Error (MLSE) channel estimation technique

Figure 3-1 shows a general layout for an OFDMA mobile communication system, which exploits Channel estimation and signal detection. The transmitter source is protected by channel coding against fading phenomenon; subsequently the binary signal is modulated over its own carrier and transmitted over the radio channel. After that additive noise is added and the sum signal is received. Once the signal is detected, the channel is decoded to extract the original message.

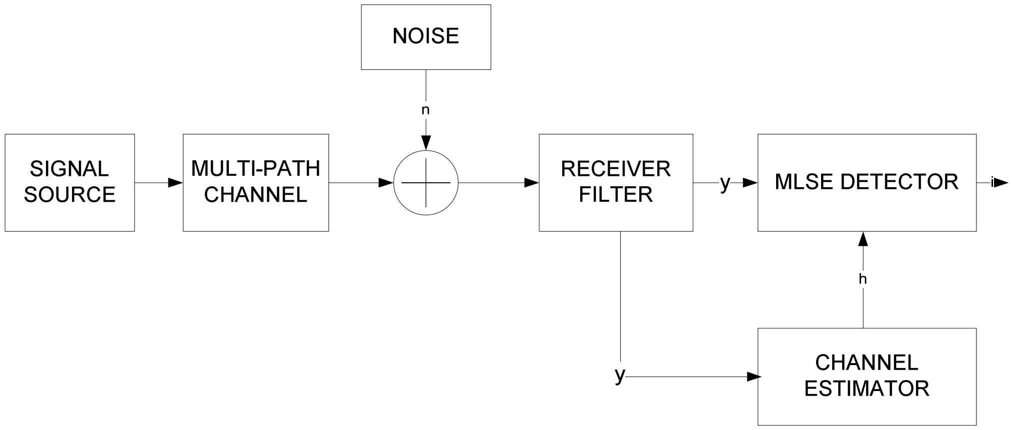

Figure 3-2: System channel estimator and detection

Signal is transmitted over a fading multipath channel and the thermal noise “which is modeled as an additive white Gaussian noise” is added. The demodulation starts by detecting the transmitted bits from the received signal y. to do so; the detector requires also the channel impulse response *(h)*. The sequence of operation can be expressed as follows:

The received signal at the receiver is *y* and it’s the product of the channel transition matrix and the channel impulse response added to the additive noise as in (3-4)

$$\mathbf{y} = \mathbf{M}\mathbf{h} + \mathbf{n} \qquad (3\text{-}4)$$

The complex channel impulse response *h* is expressed as (3-5)

$$\boldsymbol{h} = [\mathbf{h_0} \quad \mathbf{h_1} \quad \cdots \quad \mathbf{h_L} \ ]^{\mathbf{T}} \qquad (3\text{-}5)$$

For each transmission burst, the transmitter sends its unique bit training sequence which is divided into a reference length *P* and guard period of *L* bits. This will be expressed as (3-6)

$$\mathrm{m} = [\mathrm{m_0} \quad \mathrm{m_1} \quad \cdots \quad \mathrm{m_{P+L-1}} \ ]^{\mathrm{T}} \qquad (3\text{-}6)$$

Where $\mathrm{m_j} \in \{-1, +1\}$, and the matrix M is expressed as (3-7)

$$\mathbf{M} = \begin{bmatrix} \mathbf{m_L} & \cdots & \mathbf{m_1} & \mathbf{m_0} \\ \mathbf{m_{L+1}} & & \mathbf{m_2} & \mathbf{m_1} \\ \vdots & \ddots & \vdots & \\ \mathbf{m_{L+P-1}} & \cdots & \mathbf{m_P} & \mathbf{m_{P-1}} \end{bmatrix} \qquad (3\text{-}7)$$

The LS judge the channel based on minimizing the following error amount

$$\mathbf{h} = \mathbf{arg\,min}\,\|\mathbf{y} - \mathbf{Mh}\|^{\mathbf{2}} \qquad (3\text{-}8)$$

Finally, the channel impulse response solution for the previous equation is defined as in (3-11)

$$\mathbf{h_{LS}} = (\mathbf{M^H M})^{-\mathbf{1}}\mathbf{M^H y} \qquad (3\text{-}9)$$

Where $(\,)^{\mathrm{H}}$ and $(\,)^{-1}$ denote the Hermitical and inverse matrices.

3.4.2 User SNR and Channel Impulse Response

SNR for sub-channel *(f)* who belong to user *(i)* was evaluated according to (3-10)

$$\boldsymbol{SNR}_{\boldsymbol{user(i)}}^{\boldsymbol{channel(f)}} = \frac{|\boldsymbol{H(f)}|^{\mathbf{2}}}{\boldsymbol{N(f)}} \qquad (3\text{-}10)$$

3.4.3 Frequency Selective Fading Channel

Radio channel can be modeled for a particular system according to the bandwidth required for communication. In the flat fading approach bandwidth of the channel is greater than in the required bandwidth of the communication system.

$$B_S < B_C \qquad (3\text{-}11)$$

Where BS is the bandwidth of the signal and BC is the bandwidth of the channel. Amplitudes of frequencies within the required bandwidth can be approximated by Gaussian distribution function. In another words, all frequency components of the communication system will pass through the channel unaltered with addition of noise. We have to outline here that signal undergoes constant gain and linear phase response propagating through flat fading channel.

On another hand for frequency selective channel required bandwidth for communication is greater than channel bandwidth.

$$B_S > B_C \qquad (3\text{-}12)$$

Thus at the transmission numerous frequencies are eliminated, as if signal underwent filtration process. Frequency selective fading makes information recovery significantly more difficult. In equation (3-15), we show equation that is used to determine channel bandwidth for modeling technique.

$$\boldsymbol{B_c} \approx \frac{\mathbf{1}}{\mathbf{50}\boldsymbol{\sigma_t}} \qquad (3\text{-}13)$$

Where σ_t is the RMS delay spread - estimated from practical test measurements

It is clear that for OFDMA communication we must use frequency selective channel in order to model accurately wireless system. In figures 3-2 the channel impulse response of the frequency selective system is depicted in time domain for carrier frequency 2 GHz, bandwidth 10 kHz; RMS spread delay of 10μs and finally velocity of the mobile receive 20km/h. With the help of this graph we get a sense of the wireless channel for OFDMA communication system. In order to simulate frequency selective channel we use Rayleigh fading distribution equation (3-14) [51].

$$\boldsymbol{p}(\boldsymbol{r}) = \begin{cases} \frac{r}{\sigma_t^2}\,\mathbf{exp}\left(-\frac{r^2}{2\sigma_t^2}\right) & (\mathbf{0} \le \boldsymbol{r} \le \infty) \\ \mathbf{0} & (\boldsymbol{r} < 0) \end{cases} \qquad (3\text{-}14)$$

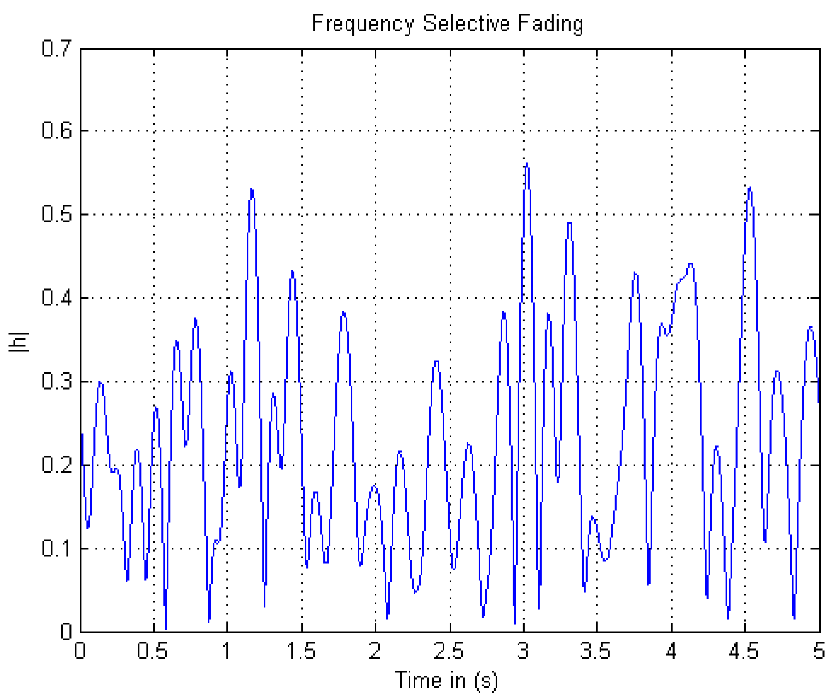

Figure 3-3: Frequency selective fading channel impulse response

3.5 Limited Mobile Model Scenarios

In the limited mobile scenario, the cell will have 24 MSs uniformly distributed among the four MCS zones. Each zone is assumed to have 6 randomly distributed MSs, which means that each MS has two random *x* and *y* coordinates within its region one as the starting point and the other as the destination point, the location for each MS varies each cycle one step toward the destination but never pass the border of the SS modulation zone, i.e. the MS follow his way to the destination point which is located within the same original QAM zone that belong to this SS. Each MS generates three types of connections based on the three classes of service supported here (rtPS, nrtPS, and BE), leading to a total of 72 connections. The QoS parameters for each class of service are listed in Table 3.1

3.6 Full Mobility Model (Free end to end mobility)

A random mobility model was developed in which we represent the cell by a finite set of points "A". Let "A" denote the two dimensional bordered region in which all mobile SS operate. Each mobile SS is assigned a random starting point as $z_s = (x_{start}, y_{start}) \in A$, and a random destination point $z_d = (x_{destination}, y_{destination}) \in A$ and a random velocity v where 0 < v < v_max. For our simulation, we used *v_max* as 20 m/s. The Base station (BS) is located at the center of the cell. In the mobility model, every user moves one step closer to his destination in segment of $v \times cycle\_time$ meters. The model calculates the slope between the user's current location and the user's destination and then moves the user along this path according to his interval. Once the user reaches his destination, a new random destination is chosen such that: $z_d = (x_{destination}, y_{destination}) \in A$.

3.7 Mobility Considerations

Our mobility model considers three possible cases of motion: from left to right, right to left, and vertical. In every case we aim to increment the users x and y coordinates by x_{step} and y_{step} respectively. We calculate the variables as follows:

3.7.1 Case 1: Moving from Left to Right

$$\mathbf{x_{step} = x_1 + d\,Cos\left(tan^{-1}\left(\frac{y_d - y_1}{x_d - x_1}\right)\right)} \tag{3-15}$$

$$\mathbf{y_{step} = y_1 + d\,Sin\left(tan^{-1}\left(\frac{y_d - y_1}{x_d - x_1}\right)\right)} \tag{3-16}$$

3.7.2 Case 2: Moving from Right to Left

$$\mathbf{x_{step} = x_1 - d\,Cos\left(tan^{-1}\left(\frac{y_d - y_1}{x_d - x_1}\right)\right)} \tag{3-17}$$

$$\mathbf{y_{step} = y_1 - d\,Sin\left(tan^{-1}\left(\frac{y_d - y_1}{x_d - x_1}\right)\right)} \tag{3-18}$$

3.7.3 Case 3: Vertical Movement

Where $x_1 = x_d$ and $y_1 \neq y_d$,in this case x_1 has to be replaced by $x_1 - 1$,to avoid getting the slope infinity. Note: slope = $\frac{\Delta y}{\Delta x}$. Note: Horizontal movement is considered as a subset from case 1or case 2. Where x_1 , y_1 represent the user's current x and y coordinates respectively and d is going to be the segment in which the user moves towards his destination given by $= v \times$ cycle_time . x_d and y_d Are the user's destination x and y coordinates respectively.

3.8 Traffic and Simulation Parameters

Table 3.1 : Connections and Traffic Generated characteristics by SSs

Traffic type	Max. sustained traffic rate	Minimum Reserved rate (MRR)	Number of connections
UGS	8 kbps	8 kbps	20
rtPS	100 kbps	60 kbps	20
nrtPS	100 kbps	60 kbps	20
BE	100 kbps	0kbps	20

Table 3.2 : Modulation scheme, SNR (dB) and Slot- Rate (bps)

Modulation Scheme	Coding Rate	Receiver SNR (dB)	Coverage %	Slot Rate (Best-performance-No fading effect) Kbits/s
QAM-64	¾	24.4	4	43.2
	½	22.7	6	38.4
QAM-16	¾	18.2	7	28.8
	½	16.4	6	19.2
QPSK	¾	11.2	27	14.4
	½	9.4	20	9.6
BFSK	½	6.4	30	4.8

Table 3.3 : Simulation Parameters

Parameter	Value
System Band width	20 (MHZ)
FFT size	2048
Number of Sub-Channels	32
Sub-Carrier Spacing , Δf	9.765 (KHZ) (20MHZ/9.765=2048)
Useful Symbol Time ,T_b	91.4 microseconds
Guard Time, T_g	11.4 microseconds
Frame duration	5 milliseconds
OFDMA Symbol Duration	102.9 microseconds
Data frequency Carriers	1536
Number of DL Slots	36 X 32 = 1536
Number of UL Slots	36 X 12 = 384
OFDM Symbols /Frame	48
Cell Size	5 X 5 (Km)
BS Coordinates , Center	(x =2500, y =2500) m
Number of users in the Cell	20 ~ 24
Number of users in each region.	5 or 6 Uniformly distributed in Fixed Scenario
User Speed	15 m/sec , 11 m/sec , 7 m/sec
Transmit power	30 dBm
Noise power	-100.97 dBm
Propagation model	Cost 231 Hata
Channel Model	Frequency selective fading channel(Raleigh fading)
Carrier frequency	2.4 GHZ
Number of flows for each user	4(UGS, rtPS , nrtPS, BE)
Average data rate per flow	100 kb/sec for rtPS , nrtPS BE = 8 kb/sec for UGS
Noise power	-100.97(dBm)
BS antenna Height	30 (m)
Mobile user Height	1.5 (m)
Thermal Noise	-174 dBm
Slot Data rate	Vary according to AMC
BS Transmit Power, P_t	30 (dBm)
BS Height	30 (m)

Chapter 4

A Novel Cross Layer Three Level Hierarchal Algorithm (3LHA).

4.1 State of the art

The motivation beyond the proposed UL cross layer scheduling algorithm is that different classes of service have different and typically conflicting requirements. To address this problem, the proposed algorithm builds on a three-level hierarchical structure in order to meet these diverse QoS requirements. The first-level priority metric of the hierarchy is based on the MCS zone where the SS is located. As indicated in Table 3.2, the MCS zone is selected based on the channel conditions, i.e., the SNR values reported by the MSs to the BS receiver via the UL CQI channel. As can be seen from Figure 4-1, for a given number of "*n*" MCSs, the range of the entire received SNR values is divided into *n* non-overlapping circular zones, in which each zone is mapped to a certain MCS, the coverage % of each zone is calculated by the propagation model and seen in Table 3.2

For simplicity, we assume that the scheduling algorithm supports three classes of service: rtPS, nrtPS, and BE, but it can be easily extended to support any number of services. Each MS has three queues, one for each class of service. At each cycle (frame), MSs report their current queue sizes for each rtPS, nrtPS, and BE connection (admitted flow) in either bytes or the

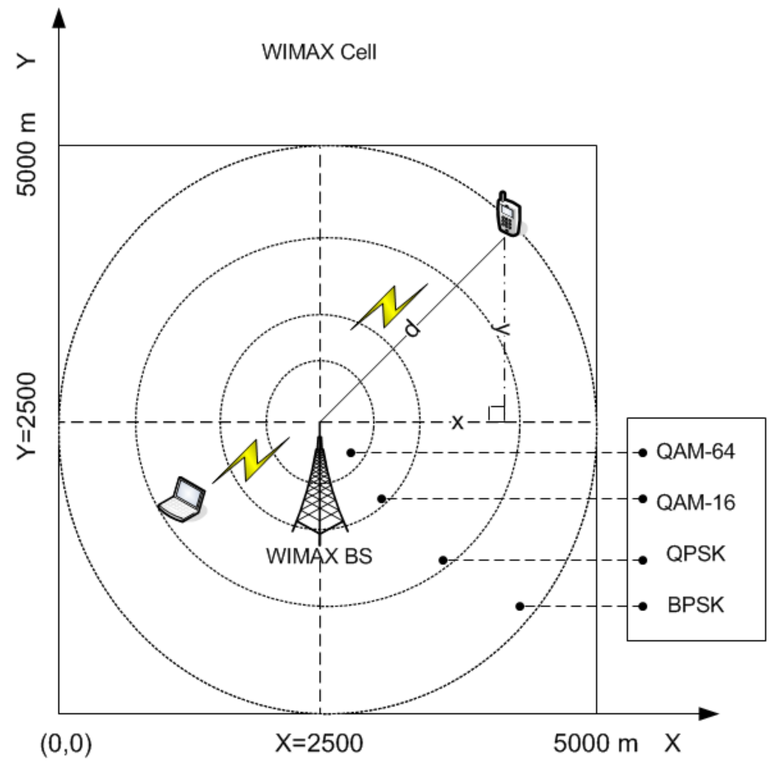

Figure 4-1 : WIMAX Cell Structure

instantaneous data rate to the BS for bandwidth allocation for the next cycle. For rtPS services, the bandwidth request message also includes the arrival time of the head of the line (HOL) packet.

4.2 The scheduler mechanism

The BS scheduler converts the instantaneous data rate of each connection to the appropriate number of Slots based on the connection zone as (4-1).

$$Req\_S_{requested}^{frame_x} = \sum_{i=1}^{a} \frac{\sum_{j=1}^{k} Connection\_BW\_request_{i,j}}{S\_rate_{QAM64}} + \sum_{i=1}^{b} \frac{\sum_{j=1}^{k} Connection\_BW\_request_{i,j}}{S\_rate_{QAM16}} + \sum_{i=1}^{c} \frac{\sum_{j=1}^{k} Connection\_BW\_request_{i,j}}{S\_rate_{QPSK}} + \sum_{i=1}^{d} \frac{\sum_{j=1}^{k} Connection\_BW\_request_{i,j}}{S\_rate_{BPSK}} \quad (4\text{-}1)$$

Where $\mathrm{Req\_S}_{\mathrm{requested}}^{\mathrm{frame_x}}$ represents the total number of Slots requested for each frame x. Variables a, b, c, and d are the total number of users in the different regions (QAM-64, QAM-16, QPSK, and BPSK respectively). K represents the number of QoS connections (typically 3) for each user (rtPS, nrtPS and BE).

The scheduler depend on Exponential averaging to determine the instantaneous rate for different flows(4-2) .this will inform the network of the correct bandwidth request and therefore to let the scheduling model determines whether to increase or decrease the number of Slots required to accommodate a certain flow.

$$R_i^{new} = \left(1 - e^{\frac{-T_i^k}{N}}\right) \frac{l_i^k}{T_i^k}\, e^{\frac{-T_i^k}{k}}\; R_i^{old} \qquad (4\text{-}2)$$

Where$\mathrm{R_i^{new}}$ is the instantaneous rate of each flow ,$\mathrm{T_i^k}$and $\mathrm{l_i^k}$ are the arrival time and the length of the kth packet of flow i. The estimated rate of flow i, $\mathrm{R_i^{new}}$, which is updated every time a new packet is received, is given by [13], Where $T_i = t_i^k - t_i^{k-1}$ and N is a constant between 100 and 500 msec (we use $N = 300$ ms).

Since the number of Slots that are assigned for each user cannot be a fraction, the scheduler will always round the required number of Slots down and increments the MU credit by the remainder. Once the credit accumulates at least 1 Slot, the scheduler will assign an additional Slot for the MU for one frame. The credit function is defined as:

$$\boldsymbol{Credit}_{\boldsymbol{MU(i)\ in\ QAM(E)}}^{\boldsymbol{frame_x}} = \boldsymbol{Credit}_{\boldsymbol{MU(i)\ in\ QAM(E)}}^{\boldsymbol{frame_{x-1}}} + \boldsymbol{remainder}_{\boldsymbol{frame\ x}}\left(\frac{\sum_{j=1}^{k} \boldsymbol{Connection}_{\boldsymbol{BW_{request_j}}}}{\boldsymbol{PS}_{\boldsymbol{rate_{QAM(E)}}}}\right) \quad (4\text{-}3)$$

The number of bytes that can be transmitted with each Slot which depends exclusively on the channel conditions (SNR) at the receiver, which, in turn, depend on the location of the MS is defined as (4-4).Table.1 shows relation between SNR, coding rate and Slot rate for different QAM regions.

$$\text{Slot-Capacity} = \frac{\boldsymbol{No.\_subcarries} \times \frac{\boldsymbol{bits}}{\boldsymbol{symbol}} \times \boldsymbol{coding\_rate}}{\boldsymbol{8}} \qquad (4\text{-}4)$$

The scheduler will check if the total available Slots can accommodate the total requested Slots as in (4-5)

$$Req\_\boldsymbol{S}_{\boldsymbol{Total}}^{\boldsymbol{frame_x}} \leq Tot\_\boldsymbol{S}_{\boldsymbol{available}}^{\boldsymbol{frame_x}} \qquad (4\text{-}5)$$

If the network cannot support the total requested Slots for each frame as in (4-6)

$$Req\_\boldsymbol{S}_{\boldsymbol{Total}}^{\boldsymbol{frame_x}} > Tot\_\boldsymbol{S}_{\boldsymbol{available}}^{\boldsymbol{frame_x}} \qquad (4\text{-}6)$$

The scheduler in this case will grant the Minimum Reserved Rate (MRR) for rtPS and nrtPS connections to maintain their QoS constrains. The remaining available Slots are calculated as follows:

$$\mathbf{Remaining}_f^S = \mathbf{Total}_f^S - \sum_{i=1}^{m}\sum_{j=1}^{k}\mathbf{Ceiling}\left(\frac{MRR_{i,j}}{S_{rate_{QAM(E)}}}\right) \quad \text{if credit} < 1$$

(4-7)

$$\mathbf{Remaining}_f^S = \mathbf{Total}_f^S - \sum_{i=1}^{m}\sum_{j=1}^{k}\mathbf{Ceiling}\left(\frac{MRR_{i,j}}{S_{rate_{QAM(E)}}}\right) - \sum_{i=1}^{m} Floor(Credit_{MU(i)\ in\ QAM(E)}^{frame_x}) \text{, when credit} \geq \mathbf{1}$$

(4-8)

Where m is the total number of Mobile Users (MU's) in the cell, k is the total number of connections in the cell, and $S_{rate_{QAM(E)}}$ is the throughput of each Slot for each "i" user in modulation zone "E".

$\mathrm{Remaining}_f^S$ is the number of remaining Slots after granting the minimum reserved rate for all users in each frame " f ". The remaining Slots are assigned based on the sequence given in Figure 4-2 as follows: (a) the scheduler implements the first-level priority metric by serving the queues of the MSs in the following order: QAM-64 zone, QAM-16 zone, QPSK zone, and finally BFSK zone.

(b) The scheduler applies the second-level typical strict priority algorithm for delay sensitive classes within each MCS zone as follows:

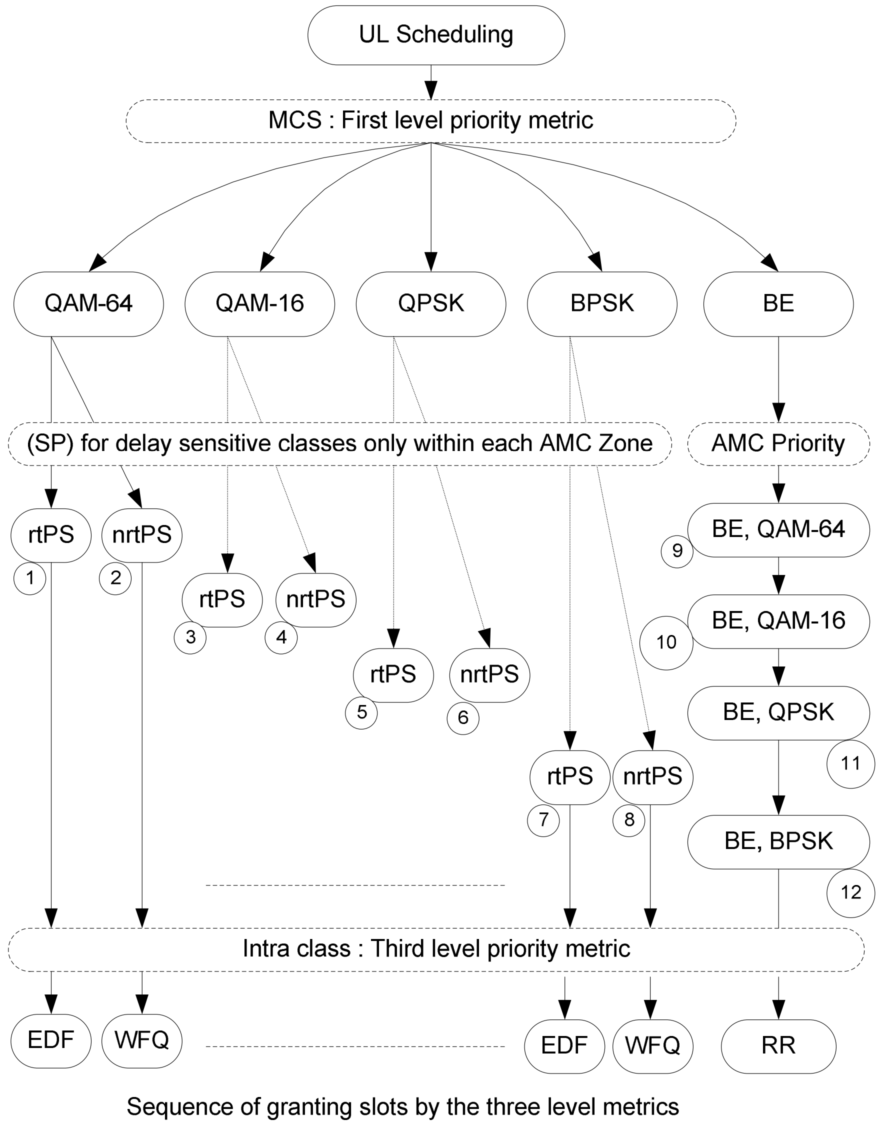

Figure 4-2 : Schematic diagram of the cross layer scheduling

rtPS connections in QAM-64 zone are served first, followed by all nrtPS queues, BE is skipped in this phase to the next AMC zone. The reason that the scheduler will give the highest priority for all delay sensitive connections in the QAM-64 zone regardless the type of traffic is the low cost when it comes to Slot rate in OFDM system, where connections with high SNR lead to increase proportionally Slot rate accordingly decrease the number of Slots required to accommodate any connection in this region (one Slot in QAM-64 zone can support approximately 10-12 times the rate than the capacity of the same Slot in BPSK zone). Next in sequence are connections in QAM-16 zone, in which all rtPS queues are served first, followed by all nrtPS queues, BE in this region is post boned to the end as the Figure 4-2 shows, the reason for that is the relative high cost of Slots for connections with weaker SNR, the same thing will happen to QPSK and BPSK, where the connection served first by the rtPS followed by nrtPS.

Note: The previous order of scheduling will not affect the QoS parameters for various connections, because the scheduler already accommodate the QoS parameters for all rtPS and nrtPS by granting the minimum reserved rate before it goes through this phase. After accommodating the rtPS and nrtPS connection in the aforementioned sequence, the scheduler will consider all the BE queues with the priority giving based on the AMC in the following order: QAM64 queues goes first followed by QAM16 the next are QPSK queues and lastly BPSK queues.

(c)At the third-level in the hierarchy, the scheduler implements three different intra-class algorithms to schedule within each class of service for each of MCS zone; EDF is used for rtPS (HOL packets with earliest deadline are scheduled first), WFQ for nrtPS where the weight of each connection is set as the ratio between the connection’s nrtPS average data rate and total nrtPS average data rates within the specified zone, and RR for BE. all BE connections of QAM-16, QFSK and BPSK share the left over Slots.

4.2 Performance Evaluation

The performance of the proposed UL 3LHA is compared with that of a conventional strict priority Algorithm (SP) which utilizes strict priority between different classes of service by granting all Slots requested by the rtPS queues followed by the nrtPS and finally all BE queues. The performance metrics used here are the overall network throughput as well as the average throughput and packet Delay of all connections for a given class of service in the network.

The queue size of each type of traffic is set to 10 kB. We assume that all users in the cell have line of sight with the BS. The path loss is calculated for each MS based on the channel model described above in Section 3.3.

In Figure (4-3); SP outperforms our proposed schedule in term of rtPS class of service, this is due to the nature of the SP algorithms in which it sacrifice all other CoS and connections to maintain the band width for the highest priority traffic, however our proposed algorithm outperforms SP in term of nrtPS and BE traffic. On the top of that as shown in figure (4-4); 3LHA outperformed the SP in terms of the overall UL system rate.

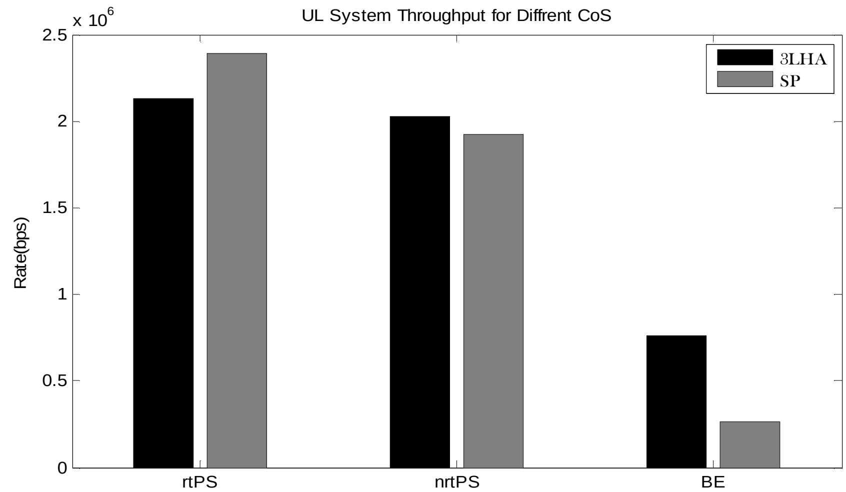

Figure 4-3 : Average UL system throughput for each class of service for all connections for a given class of service

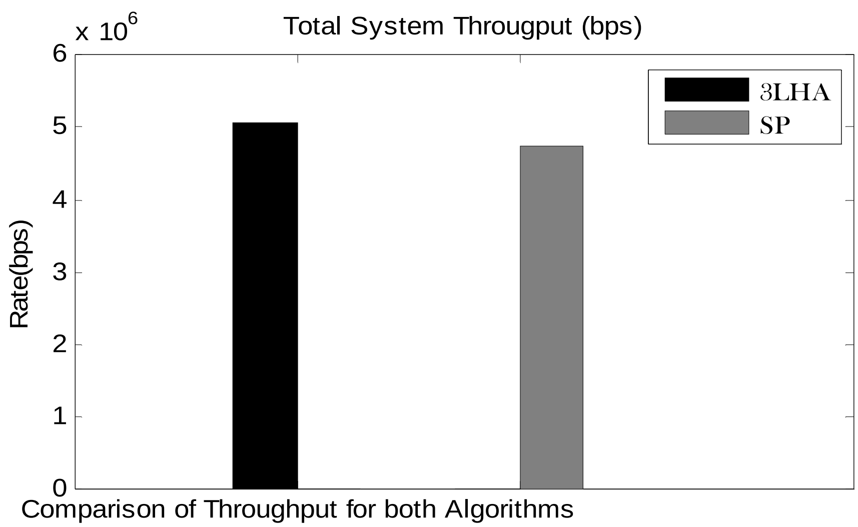

Figure 4-4 : Overall UL system throughput

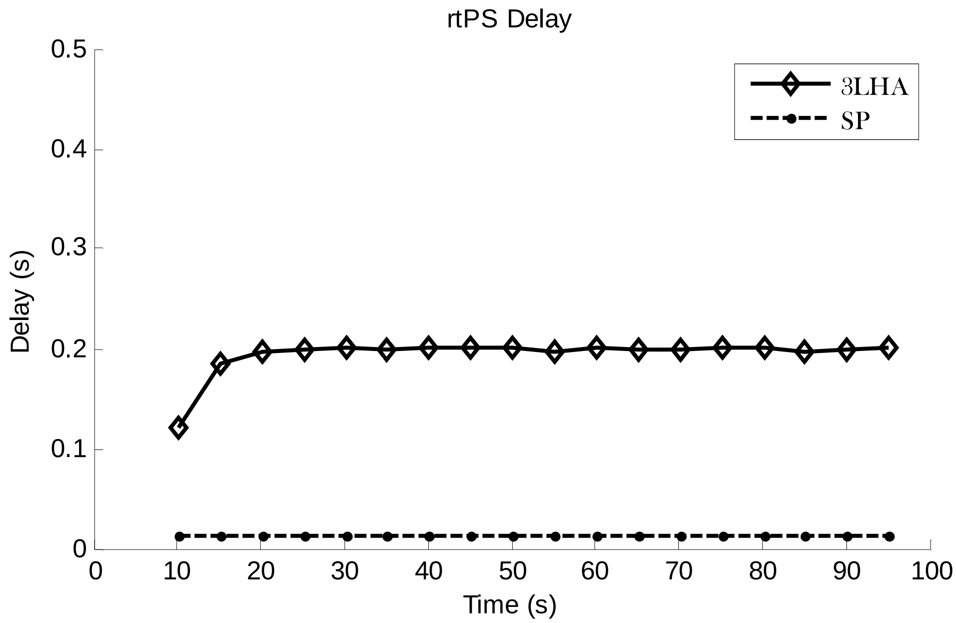

Figure 4-5 : Average Delay for rtPS class

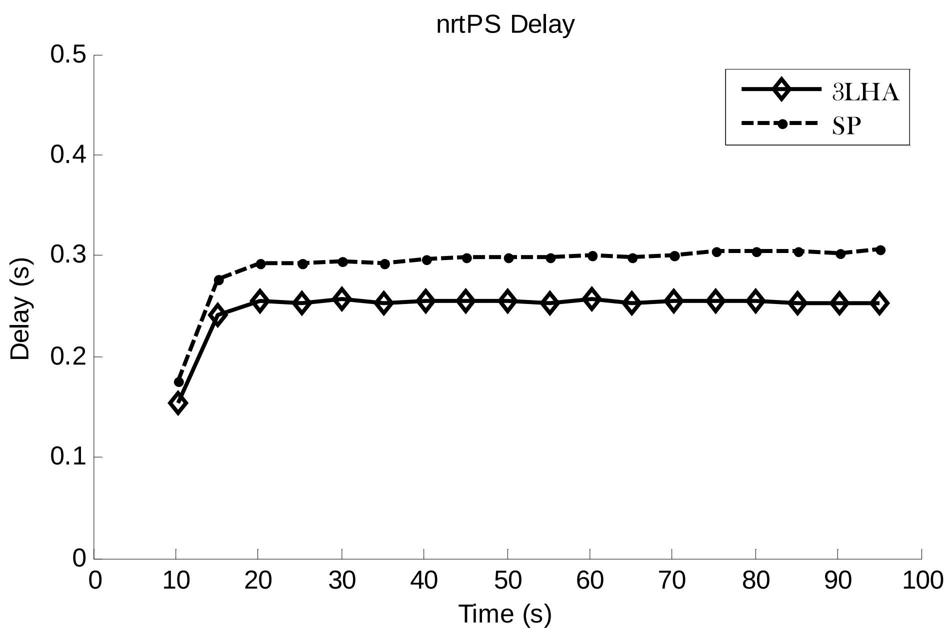

Figure 4-6 : Average Delay for nrtPS

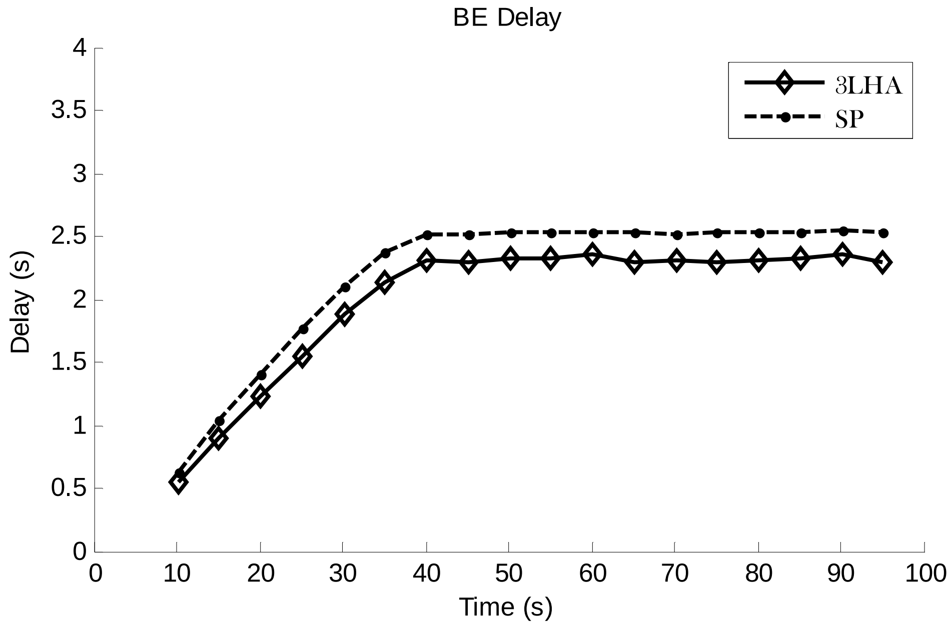

Figure 4-7 : Average Delay for BE Class

Figures (4-5) shows that SP achieved better delay than our proposed algorithm, this is expected because SP is built on starving the low traffic priority which is clear in increasing the delay of the lower classes priority rapidly as in the nrtPS and BE as the figures show, however the rtPS delay for 3LHA still within the limits of the standard values. Figures (4-6) and (4-7) show the average delay for all nrtPS and BE connections, respectively, for both algorithms. It can be seen that the average delay for those two classes for the proposed algorithm much less than those in the SP algorithm.

4.3 Conclusion

This work has presented a three-level hierarchical algorithm for UL scheduling in Mobile WIMAX that targets specific requirements of each scheduling service while enforcing an opportunistic cross-layer approach in order to meet QoS requirements and maximize the overall system throughput. The simulation results indicate that the performance of the proposed algorithm outperforms that of the convention strict Priority scheduling algorithm in terms of network throughput and delay.

Chapter 5

A Novel Cross layer Modified by Zone Balance Ratio QoS Provisioning Algorithm

5.1 Introduction

IEEE 802.16e is a connection oriented network; connection has to be established between a connection that belong to a certain subscribers (SS) and the BS. According to availability of the requested resources which will maintain the minimum reserved traffic rate (M_{RR}), the BS decides whether or not to accept the connection.

Due to unpredictable nature of the mobility pattern of mobile users and the unstable mobile wireless channel, the link quality adaption (LA) is subject to change rapidly, MAC layer has to respond and reacts for any change in the channel condition by changing the adaptive modulation and coding technique (AMC). This will affect the Slot-rate which is granted for MS. Therefore maintaining the QoS during the connection life time has became a challenging issue. On the other hand balance between fairness and bandwidth utilization and their effect on QoS has become the second issue that any scheduler has to consider.

This chapter presents A Novel Cross layer Modified by Zone Balance Ratio QoS Provisioning Algorithm which operate as the second phase of improving the functionalities of the proposed RRM at the MAC layer. Solutions for the fairness problem and different AMC and different slots rates are considered.

In this phase, the QoS is grantee for all connections by securing the minimum reserved rate (M_{RR}), then the zone balance ration is generated (ZBR) “was published in our paper” [52] to set the upper limit threshold for all connections to maintain fairness among them and to prevent anyone from ceasing the whole band width, in addition to use the 3LHA was introduced in chapter four to distribute any leftover slots.

Phase

5.2 Overview of the Algorithm

The scheduler has three stages; in the firs the scheduler will determine whether it can grant the total requested BW to all users. If it cannot, it will begin by granting the Minimum Reserved Rate (MRR) for delay sensitive connections in order to maintain their QoS. Afterward, in the second stage it will generate the Zone Balance Ratio (ZBR) table depending on the base line case; based on the previously mentioned the scheduler will determine the maximum connection threshold bandwidth granted for each connection by calculating the average modified by the zone balance ratio (AMZBR), This threshold will adapt and change the base line case based on the total amount of the requested bit rate in each zone. It will try to make its best to grantee there are enough network resources available for all connections in the cell based on the ZBR table, this will stabilize connection while mobile user is moving. Finally, in the third stage the scheduler determines the type of each connection whether it is high rate request (HRRC) or low rate request (LRRC) in order to grant any leftover Slots. The connection is defined as HRRC if the requested Slots exceed the maximum granted threshold per connection and defined as LRRC if the requested Slots are less than the maximum granted Slots per connection.

For simplicity, we assume that the scheduling algorithm supports three classes of service: rtPS, nrtPS, and BE, but it can be easily extended to support any number of services. Each MS has three queues, one for each class of service. At each cycle (frame), MSs report their current queue sizes for each rtPS, nrtPS, and BE connection (admitted flow) in either bytes or the instantaneous data rate to the BS for bandwidth allocation for the next cycle. For rtPS services, the bandwidth request message also includes the arrival time of the head of the line (HOL) packet.

5.2.1 Stage: 1 Maintain the QoS by Granting the M_{RR}

The scheduler accommodates the QoS parameters for all rtPS and nrtPS by granting the minimum reserved rate. Each cycle The BS scheduler converts the instantaneous data rate for each connection to the appropriate number of physical Slots; by dividing the exponential average requested bit rate by the Slot rate which is a function in the user AMC, user location and wireless channel status represented by the SNR as in (5-1). Equation (5-1) shows that the total requested Slots is a function in the connections and also the users distribution of the network.

$$\boldsymbol{Req\ \_S_{requested}^{frame_x}} = \sum_{i=1}^{a} \frac{\sum_{j=1}^{k} Connection\_BW\_request_{i,j}}{S\_rate_{QAM64}} + \sum_{i=1}^{b} \frac{\sum_{j=1}^{k} Connection\_BW\_request_{i,j}}{S\_rate_{QAM16}} + \sum_{i=1}^{c} \frac{\sum_{j=1}^{k} Connection\_BW\_request_{i,j}}{S\_rate_{QPSK}} + \sum_{i=1}^{d} \frac{\sum_{j=1}^{k} Connection\_BW\_request_{i,j}}{S\_rate_{BPSK}} \quad (5\text{-}1)$$

Where $Req\ \_S_{requested}^{frame_x}$ represents the total number of requested Slots for each frame *x*. Variables *a, b, c,* and *d* are the total number of users in the different modulation regions (QAM-64, QAM-16, QPSK, and BPSK respectively). K represents the number of QoS connections (typically 3) for each user (rtPS, nrtPS and BE).

The scheduler depend on Exponential averaging to determine the instantaneous rate for different flows(2) .this will inform the network of the correct bandwidth request and therefore to let the scheduling model determines whether to increase or decrease the number of Slots required to accommodate a certain flow.

$$R_i^{new} = \left(1 - e^{\frac{-T_i^k}{N}}\right) \frac{l_i^k}{T_i^k}\; e^{\frac{-T_i^k}{k}} \;\; R_i^{old} \qquad (5\text{-}2)$$

WhereR_i^{new} is the instantaneous rate of each flow ,$T_i^k$and l_i^k are the arrival time and the length of the k^{th} packet of flow i. The estimated rate of flow i, R_i^{new} , which is updated every time a new packet is received, is given by [53], Where $T_i = t_i^k - t_i^{k-1}$ and N is a constant between 100 and 500 msec (we use $N = 300$ ms).

Since the number of Slots that are assigned for each user cannot be a fraction, the scheduler will always round the required number of Slots down and increments the Mobile user (MU) credit by the remainder. Once the credit accumulates at least 1 Slot, the scheduler will assign an additional Slot for the MU for one frame. The credit function is defined as:

$$Credit_{MU(i)\ in\ QAM(E)}^{frame_x} = Credit_{MU(i)\ in\ QAM(E)}^{frame_{x-1}} + remainder_{frame\ x} \qquad (5\text{-}3)$$

The Slot rate can be obtained from (5-4), it depends exclusively on the channel conditions (SNR) at the receiver, which, in turn, depend on the location of MS. Table 3-2 shows the relation between SNR, coding rate and Slot rate for different QAM regions.

$$S^{Rtae} = No._{subcarries} \times \frac{bits}{symbol} \times coding_{rate} \qquad (5\text{-}4)$$

The scheduler will check if the total available Slots can accommodate the total requested Slots as in (5-5)

$$Req\_\boldsymbol{S}_{\boldsymbol{Total}}^{\boldsymbol{frame_x}} \leq Tot\_\boldsymbol{S}_{\boldsymbol{available}}^{\boldsymbol{frame_x}} \quad (5\text{-}5)$$

If the network cannot support the total requested Slots for each frame as in (5-6)

$$Req\_\boldsymbol{S}_{\boldsymbol{Total}}^{\boldsymbol{frame_x}} > Tot\_\boldsymbol{S}_{\boldsymbol{available}}^{\boldsymbol{frame_x}} \quad (5\text{-}6)$$

The scheduler in this case will grant the Minimum Reserved Rate (*MRR*) for *rtPS* and *nrtPS* connections to maintain their QoS constrains. The remaining available Slots are calculated as follows:

$$\text{Remaining}_x^S = Tot\_S_{available}^{frame_x} - \sum_{i=1}^{m}\sum_{j=1}^{k}\left\lceil \frac{MRR_{i,j}}{PS_{rate_{QAM(E)}}} \right\rceil \text{ if credit } < 1 \quad (5\text{-}7)$$

$$\textbf{Remaining}_{\boldsymbol{x}}^{\boldsymbol{S}} = \boldsymbol{Tot\_S_{available}^{frame_x}} - \sum_{i=1}^{m}\sum_{j=1}^{k}\left\lceil \frac{\boldsymbol{MRR_{i,j}}}{\boldsymbol{S_{rate_{QAM(E)}}}} \right\rceil - \sum_{i=1}^{m}\left\lfloor \boldsymbol{Credit_{MU(i)\ in\ QAM(E)}^{frame_x}} \right\rfloor,$$

$$\text{when credit} \geq \mathbf{1} \quad (5\text{-}8)$$

Where m is the total number of Mobile Users (MU's) in the cell, k is the total number of connections in the cell, and $\mathrm{S}_{\mathrm{rate}_{\mathrm{QAM(E)}}}$ is the throughput of each Slot for each "i" user in modulation zone "E".

5.2.2 Stage: 2 Set ZBR Table and limit the Max Threshold.

The second phase allocates the residual Slots ($\mathrm{Remaining}_{\mathrm{f}}^{\mathrm{S}}$) based on the Zone Balance Ratio (ZBR). The ZBR is a function that ensures fairness between connections across all zones (from QAM64 to BPSK). The balance ratio is obtained by comparing the modulation ratio of the Slot rate (S_Rate) of a user in QAM64 to the (S_Rate) of the current user's zone. For example, a user in QPSK would require 3 times more Slots than he would if he were located in the QAM64 region. The scheduler would reference the ZBR in order to ensure that the user's connection rate is maintained. The Zone balance ratio is initially calculated using the QAM64 region as the base line case. With a QAM-64 base line, the ZBR table is defined as follows:

$$\boldsymbol{ZBR_i^{QAM64}} = 1 \quad \text{, for QAM-64 region} \qquad \text{(Base line)} \tag{5-9}$$

$$\boldsymbol{ZBR_i^{QAM16}} = \frac{\boldsymbol{Coding{-}rate} \times \frac{\boldsymbol{bits}}{\boldsymbol{symbol}}_{\boldsymbol{QAM64}}}{\boldsymbol{Coding{-}rate} \times \frac{\boldsymbol{bits}}{\boldsymbol{symbol}}_{\boldsymbol{QAM16}}} \qquad \text{for QAM-16 region.} \tag{5-10}$$

$$\boldsymbol{ZBR_i^{QPSK}} = \frac{\boldsymbol{Coding{-}rate} \times \frac{\boldsymbol{bits}}{\boldsymbol{symbol}}_{\boldsymbol{QAM64}}}{\boldsymbol{Coding{-}rate} \times \frac{\boldsymbol{bits}}{\boldsymbol{symbol}}_{\boldsymbol{QPSK}}} \tag{5-11}$$

$$\boldsymbol{ZBR_i^{BPSK}} = \frac{\boldsymbol{Coding{-}rate} \times \frac{\boldsymbol{bits}}{\boldsymbol{symbol}}_{\boldsymbol{QAM64}}}{\boldsymbol{Coding{-}rate} \times \frac{\boldsymbol{bits}}{\boldsymbol{symbol}}_{\boldsymbol{BPSK}}} \tag{5-12}$$

Where $ZBR_i^{(p)}$ is the zone balance ratio for each user "i" located in (*p*) modulation zone, Where *p* is the total number of modulation zones (typically 4). Based on the total requested bit rate for each zone, the scheduler may dynamically change the base line according to the highest requested bit rate region each frame. The new base line could be QAM-16, QPSK or BPSK zone. Each cycle schedule need to choose the most appropriate base line for the current traffic condition each cycle. This will optimize the best performance between the throughput and the bit rate fairness (connection stabilization). Each time the schedule change the base line a new series of ZBR values are created, the following table will guide the scheduler to select the most proper base line for ZBR function each cycle based on the highest connection requested bit rate zone as shown in Table.1 below.

Table 5.1 : Base line changes according to traffic distribution in the cell

AMC & Base Line	QAM-64	QAM-16	QPSK	BPSK	Base Line
$\sum \mathit{Total\ requested\ Traffic\ per\ zone}$	H	L	L	L	QAM-16
	L	H	L	L	QPSK
	L	L	H	L	BPSK
	L	L	L	H	QAM-16

For example if the total requested bit rate is in QAM-64 zone is the highest(H) among all other zones, the base line reference has to be switched to QAM-16., obtain a new series of ZBRs as (5-13), (5-14), (5-15) and (5-16).

$$\mathbf{ZBR_i^{QAM64}} = \left[\frac{\mathbf{Coding\text{-}rate} \times \frac{\mathbf{bits}}{\mathbf{symbol}}_{\mathbf{QAM16}}}{\mathbf{Coding\text{-}rate} \times \frac{\mathbf{bits}}{\mathbf{symbol}}_{\mathbf{QAM64}}}\right] \quad (5\text{-}13)$$

$$\mathbf{ZBR_i^{QAM16}} = \mathbf{1} \text{ for QAM-16 region} \quad \text{(The new base line case)} \quad (5\text{-}14)$$

$$\mathbf{ZBR_i^{QPSK}} = \frac{\mathbf{Coding\text{-}rate} \times \frac{\mathbf{bits}}{\mathbf{symbol}}_{\mathbf{QAM16}}}{\mathbf{Coding\text{-}rate} \times \frac{\mathbf{bits}}{\mathbf{symbol}}_{\mathbf{QPSK}}} \quad (5\text{-}15)$$

$$\mathbf{ZBR_i^{BPSK}} = \frac{\mathbf{Coding\text{-}rate} \times \frac{\mathbf{bits}}{\mathbf{symbol}}_{\mathbf{QAM16}}}{\mathbf{Coding\text{-}rate} \times \frac{\mathbf{bits}}{\mathbf{symbol}}_{\mathbf{BPSK}}} \quad (5\text{-}16)$$

If QAM-16 zone has the highest requested bit rate among all other zones, QPSK would be the base line case as (5-17), (5-18), (5-19), and (5-20).

$$\mathbf{ZBR_i^{QAM64}} = \left\lceil \frac{\text{Coding-rate} \times \frac{\text{bits}}{\text{symbol}}_{QPSK}}{\text{Coding-rate} \times \frac{\text{bits}}{\text{symbol}}_{\text{QAM64}}} \right\rceil \quad (5\text{-}17)$$

$$\boldsymbol{ZBR_i^{QAM16}} = \left\lceil \frac{\boldsymbol{Coding\text{-}rate} \times \frac{\boldsymbol{bits}}{\boldsymbol{symbol}}_{QPSK}}{\boldsymbol{Coding\text{-}rate} \times \frac{\boldsymbol{bits}}{\boldsymbol{symbol}}_{QAM16}} \right\rceil \quad (5\text{-}18)$$

$$\mathbf{ZBR_i^{QPSK}} = \frac{\text{Coding-rate} \times \frac{\text{bits}}{\text{symbol}}_{QPSK}}{\text{Coding-rate} \times \frac{\text{bits}}{\text{symbol}}_{\text{QPSK}}} = 1 \quad \text{(base line case)} \quad (5\text{-}19)$$

$$\mathbf{ZBR_i^{BPSK}} = \frac{\text{Coding-rate} \times \frac{\text{bits}}{\text{symbol}}_{QPSK}}{\text{Coding-rate} \times \frac{\text{bits}}{\text{symbol}}_{\text{BPSK}}} \quad (5\text{-}20)$$

If total requested bit rate in QPSK region is the highest among all other regions, BPSK would be the new base line case as (5-21), (5-22), (5-23), and (5-24).

$$\mathbf{ZBR_i^{QAM64}} = \left\lceil \frac{\text{Coding-rate} \times \frac{\text{bits}}{\text{symbol}}_{BPSK}}{\text{Coding-rate} \times \frac{\text{bits}}{\text{symbol}}_{\text{QAM64}}} \right\rceil \quad (5\text{-}21)$$

$$\boldsymbol{ZBR_i^{QAM16}} = \left\lceil \frac{\boldsymbol{Coding\text{-}rate} \times \frac{\boldsymbol{bits}}{\boldsymbol{symbol}}_{BPSK}}{\boldsymbol{Coding\text{-}rate} \times \frac{\boldsymbol{bits}}{\boldsymbol{symbol}}_{QAM16}} \right\rceil \quad (5\text{-}22)$$

$$\mathbf{ZBR_i^{QPSK}} = \left\lceil \frac{\text{Coding-rate} \times \frac{\text{bits}}{\text{symbol}}_{BPSK}}{\text{Coding-rate} \times \frac{\text{bits}}{\text{symbol}}_{\text{QPSK}}} \right\rceil \quad (5\text{-}23)$$

$$\mathbf{ZBR_i^{BPSK}} = \frac{\text{Coding-rate} \times \frac{\text{bits}}{\text{symbol}}_{BPSK}}{\text{Coding-rate} \times \frac{\text{bits}}{\text{symbol}}_{\text{BPSK}}} = 1 \text{ (base line case)} \quad (5\text{-}24)$$

If total requested bit rate in BPSK region is the highest among all other regions, QAM-16 would be the new base line. ZBR determines the maximum number of Slots that is going to be granted for each connection;($S\_Max\,^{MCS(g)}_{conection(i)}$) this threshold prevents any connection from seizing all available Slots and limits the number of Slots granted to each connection, according to (5-25), (5-26) and (5-27). These equations guarantee that no connection gets more than the average number of remaining Slots modified by the *ZBR*; this causes fairness bit rate distribution among different connections, accordingly, the stability during the connection life time is guaranteed.

$$rtPS\_S\_Max\,^{MCS(g)}_{conection(i)} \leq \left\lfloor \frac{ZBR_{QAM(g)} \times \text{Remaining}_f^{PS}}{\left[\sum_{i=1}^{p} No.\ rtPS\ con_{QAM(p)} \times ZBR_{QAM(p)} + \sum_{i=1}^{p} No.\ nrtPS\ con_{QAM(p)} \times ZBR_{QAM(p)} + \sum_{i=1}^{p} No.\ BE\ con_{QAM(p)} \times ZBR_{QAM(p)}\right]} \right\rfloor$$

(5-25)

$$nrtPS\_S\_Max\,^{MCS(g)}_{conection(i)} \leq \left\lfloor \frac{ZBR_{QAM(g)} \times \text{Remaining}_f^{PS}}{\left[\sum_{i=1}^{p} No.\ rtPS\ con_{QAM(p)} \times ZBR_{QAM(p)} + \sum_{i=1}^{p} No.\ nrtPS\ con_{QAM(p)} \times ZBR_{QAM(p)} + \sum_{i=1}^{p} No.\ BE\ con_{QAM(p)} \times ZBR_{QAM(p)}\right]} \right\rfloor$$

(5-25)

$$BE\_S\_Max\,^{MCS(g)}_{conection(i)} \leq \left\lfloor \frac{ZBR_{QAM(g)} \times \text{Remaining}_f^{PS}}{\left[\sum_{i=1}^{p} No.\ rtPS\ con_{QAM(p)} \times ZBR_{QAM(p)} + \sum_{i=1}^{p} No.\ nrtPS\ con_{QAM(p)} \times ZBR_{QAM(p)} + \sum_{i=1}^{p} No.\ BE\ con_{QAM(p)} \times ZBR_{QAM(p)}\right]} \right\rfloor$$

(5-26)

In general the maximum number of Slots assigned to a particular connection *(i),* in a given modulation and coding zone *(g)* is:

$$S\_Max\,_{conection(i)}^{MCS(g)} = \left\lfloor \frac{ZBR_{MCS(g)} \times S_{frame}^{remaining}}{[\sum_{i=1}^{p} n.con\,_{MCS_{(p)}}^{rtPS} \times ZBR_{MCS(p)} + \sum_{i=1}^{p} n.con\,_{MCS_{(p)}}^{nrtPS} \times ZBR_{MCS(p)} + \sum_{i=1}^{p} n.con\,_{MCS_{(p)}}^{BE} \times ZBR_{MCS(p)}]} \right\rfloor \quad (5\text{-}27)$$

Where,

$S\_Max\,_{conection(i)}^{MCS(g)}$ *is the maximum number of slots has to be granted to a certain connection originated from* $MCS(g)$ *Zone,* $n.con\,_{MCS_{(p)}}^{rtPS}$, $n.con\,_{MCS_{(p)}}^{nrtPS}$, *and* $n.con\,_{MCS_{(p)}}^{BE}$ *are the number of rtPS connections nrtPS and BE connections respectively in different* $MCS's(p)$ *,where* $g \in p$ *,* p *= {1, 2, 3, 4} defines the number of Modulation and coding zones.* $S_{frame}^{remaining}$ *is the number of the current remaining slots after granting the minimum reserved rate for all connections.*

Due to burst of IP traffic, in a given frame some connections might require less than the maximum threshold while others may require more than the threshold. This combined with the remainder of the floor function produces another set of residual Slots each frame. To efficiently distribute these Slots, connections are classified into two sets; namely low request rate connections (L_{RRC}) and high request rate connections (H_{RRC}). The connection is defined as H_{RRC} if the number of requested Slots in the current frame exceeds the threshold, and defined as L_{RRC} if the number of requested Slots for this connection is below the threshold.

During each cycle the L_{RRC} with connection request $(S_i) < S\_Max\,_{conection(i)}^{MCS(p)}$ will produce a second round of remainder Slots as in (5-28)

$$\boldsymbol{S_{frame}^{remaining(Round\ 2)}} = \sum_{i=1}^{p} \sum_{i=1}^{L} (\boldsymbol{S\_Max\,_{conection(i)}^{MCS(p)}} - \boldsymbol{S_i^{MCS(p)}}) \quad (5\text{-}28)$$

Where, L: is the total number of L_{RRC} *connections.*

The high request rate connections require extra Slots than the threshold as (5-29).

$$\boldsymbol{S}_{\boldsymbol{frame}}^{\boldsymbol{over-limit}} = \sum_{\boldsymbol{i=1}}^{\boldsymbol{p}} \sum_{\boldsymbol{i=1}}^{\boldsymbol{H}} \left(\boldsymbol{S}_{\boldsymbol{i}}^{\boldsymbol{MCS(p)}} - \boldsymbol{S\_Max}_{\boldsymbol{conection(i)}}^{\boldsymbol{MCS(p)}}\right) \qquad (5\text{-}29)$$

Where, H: is the total number of H_{RRC} *connections.*

5.2.3 Stage 3: Assign the Residual Slots according to cross Level Hierarchal Principle.

In this stage the scheduler redistributes the Left over Residual Slots ($S_{frame}^{remaining(Round\ 2)}$) Based on Cross Layer Hierarchal Algorithm Figure (5-1) under restriction that each connection can get one additional extra slot only.

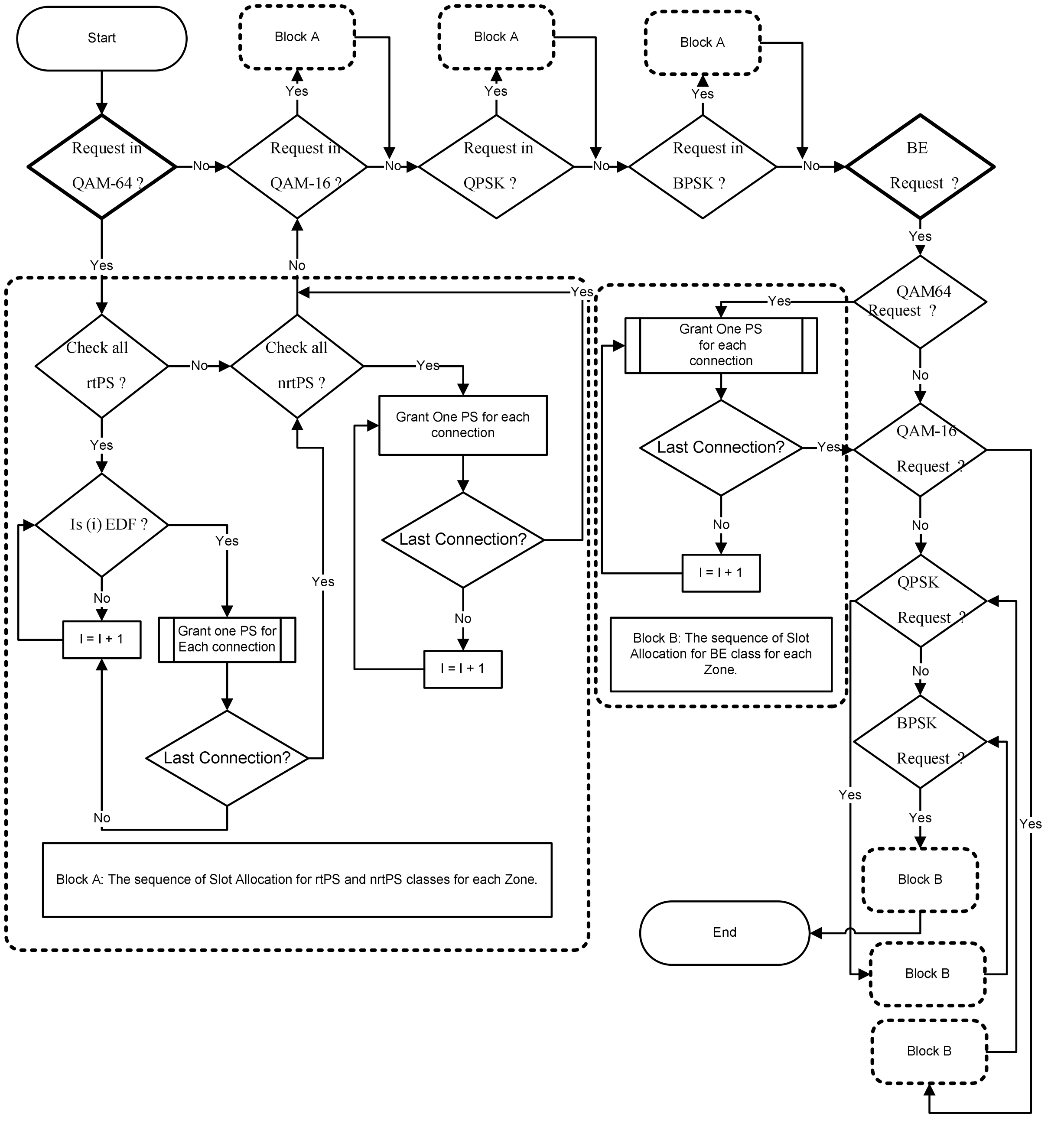

Figure 5-1 : Sequence of allocating the residual Slots

Now the total number of slots granted for each connection (i) located in modulation coding scheme (p), ($S_i^{MCS(p)}$) can be obtained as (5-30)

$$S_{i,MCS(p)}^{Granted} = \begin{cases} MRR + S_i^{MCS(p)} & If\ S_i^{MCS(p)} < S_{Max_{conection(i)}}^{MCS(p)} \\ MRR + S_i^{MCS(p)} & If\ S_i^{MCS(p)} > S_{Max_{conection(i)}}^{MCS(p)},\ and\ S_{frame}^{remaining(Round\ 2)} \geq S_{frame}^{over-limit} \\ MRR + S_i^{MCS(p)} + \Delta & If\ S_i^{MCS(p)} > S_{Max_{conection(i)}}^{MCS(p)},\ and\ S_{frame}^{remaining(Round\ 2)} < S_{frame}^{over-limit} \end{cases}$$

(5-30)

Where Δ=1 or 0, and is distributed according to cross level hierarchy algorithm principle as in Figure (5-1) , i.e. each connection can get one additional slot on round robin base until the reaming slots is equal to zero.

5.3 Results and Discussions

5.3.1 Limited Mobility Scenario: (Mobile User Moves within a Specific AMC Zone)

The performance of the proposed algorithm is evaluated in both cases; static and dynamic against the PF algorithm [54]. In the static case ZBR is calculated using the QAM64 region as a fixed base line regardless of the change in traffic conditions while in the dynamic case, the ZBR dynamically adjust the proper base line case based on the traffic condition as in table (5-1). The algorithm dynamically selects the best base line case based on the zone with the highest requested demand, this limits the absolute fairness option between different connections at the same time improves the total network throughput by optimizing both connection's stability and fairness among different connections. In PF, Slots are granted based on the ratio between the current request and the previous cycle's granted Slots. To make fair comparison among the three protocols, M_{RR} is granted for all connections and the algorithm behavior is evaluated based on the residual Slots, this may be explained as follows:

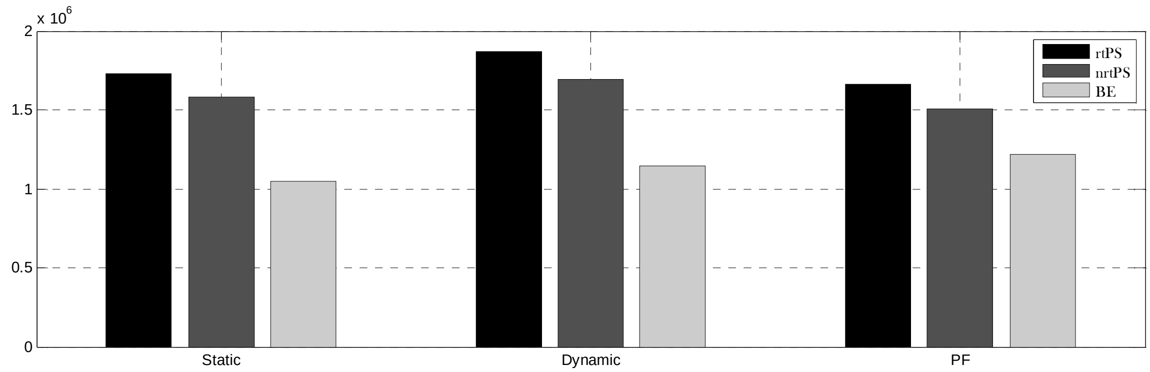

Figure 5-2 : Bar chart show throughput for 3 protocols for different CoS

Figure (5-2) shows the Dynamic ZBR-3LHA rtPS and nrtPS connections have the highest throughput compare to PF algorithm, the reason is that after granting all delay sensitive connections the M_{RR} followed by the maximum threshold the scheduler operates in a cross layer mode and grant the highest priority to rtPS and nrtPS connections in the high modulation zones in terms of the residual Slots, this occurs according to the strategy in phase 3. This will increase the throughput gradually for those connections which increase the throughput of their entire CoS. PF has less throughput because the main concept of it is absolute fairness which grants every connection approximately the same bit rate after MRR is granted; regardless of the user's location. This could cost too many Slots out of the bandwidth grid compared to our Dynamic ZBR-SP-AMC, especially if more users located in the low order modulation zones. The next highest rtPS, and nrtPS throughput is the Static Algorithm which outperforms also the PF due to the principle of usage of the cross layer approach as in phase 3 in terms of the reaming Slots to be granted for connections belong to users with good SNR.

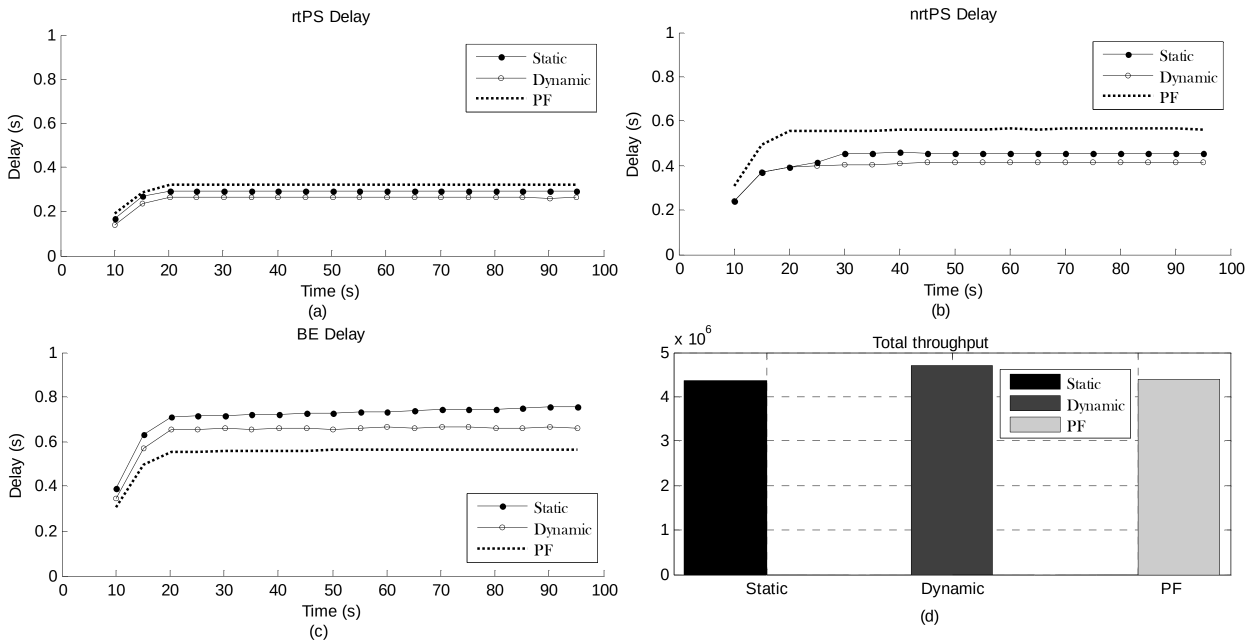

Figure 5-3 : (a),(b) and (c): show the average delay for various CoS for different algorithms. (d) is a Bar chart show the total throughput for the three protocols.

The performance metrics used in Figure (5-3) shows the overall network throughput as well as the average delay and of all connections for a given CoS within all MCS zones. Figure (5-3) (d) shows that Dynamic behavior of ZBR-3LHA has the highest overall system throughput because it can allocate more Slots to users in the higher modulation order zones; the sequence of granting the residual Slots follows 3LHA. This will maximize the throughput of rtPS and nrtPS connection, at the same time will reduce the throughput of the BE compare to PF algorithm. The reason is that the total number of residual Slots is not enough to cover all connections under high requested traffic. BE traffic in both dynamic and static ZBR-3LHA will never take advantage of any residual slot since it's the last in sequence of granting the left over Slots. In addition BE connection has zero M_{RR} Slots, most likely the only Slots available for BE connections are the Maximum threshold after all delay sensitive connections receive their M_{RR}.

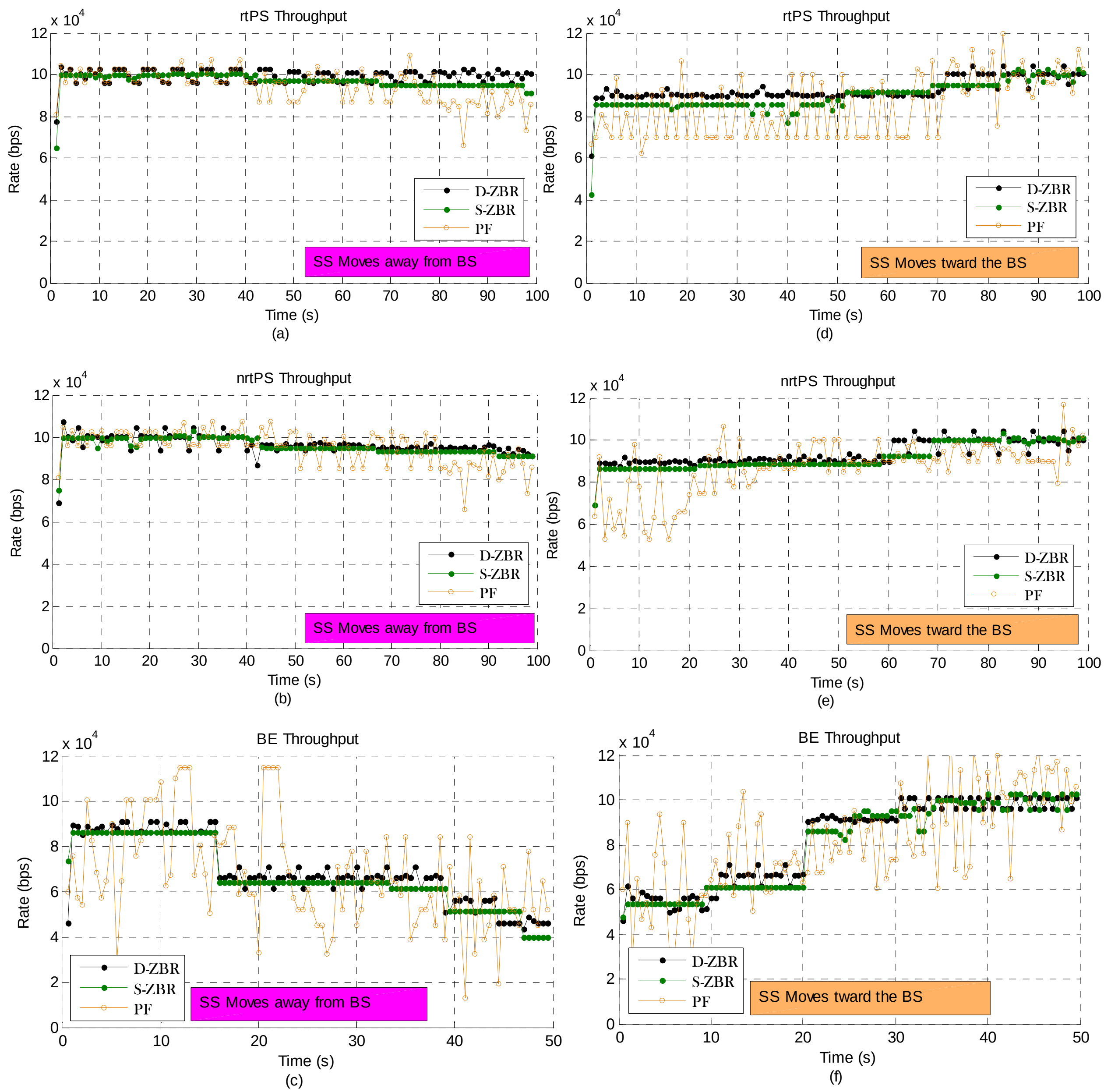

Figure 5-4 : (a), (b) and (c) rtPS throughput, nrtPS throughput, and BE throughput respectively for mobile user who is moving toward the BS. (d), (e) and (f) rtPS throughput; nrtPS throughput, and BE throughput respectively for Mobile user who moves away from the BS.

5.3.2 Unlimited Mobile Scenario (User Free to Move across all Zones):

We evaluate the Static and Dynamic ZBR-3LHA performance by measuring the average class throughput and delay for two sample users in different mobility scenario environment; both of these two mobile users move horizontally a cross the cell by a certain speed in opposite directions, one starts its trip at the cell border (at simulation time =0) and moves toward the BS to the cell center, by the end of the simulation time (t=100 sec), the user reaches the cell center and the other starts it’s trip at the cell center(at time=0) and moves away from the BS, by the end of the simulation time the user reaches the cell border (at t=100). For each scheduling algorithm, the performance of three sample connections for two mobile SSs is evaluated. Figures (5-4) (a), (b) and (c) visualize the average throughput of these connections. The curves in the figure present the up-to-date average throughput of the sample connection that belongs to both mobile users. Therefore, with the evolvement of time, each curve reflects overall traffic rate of the connection. It is explicitly shown in the figure that the proposed Dynamic ZBR-3LHA outperforms all other algorithms, especially for rtPS and nrtPS.

The difference in throughput of both mobile users who are moving toward and away from the BS is due to randomness of the location of the sample users in the user’s service table. i.e. (all users who are currently served in the cell belong to finite set of users service table and granted the BW according to their order in the table). This could make a difference between different SSs in case of assigning any extra Slots according to the third phase of the cross layer algorithms, in which a few available remaining Slots need to be distributed for many users. So it’s obvious that the user who is first in the table and satisfy the algorithm requirement’s conditions (i.e. to be located in the QAM-64 area and has the rtPS or nrtPS connection) has the highest priority.

On the other hand, Figure (5-4) (d), (e) and (f) show the stability of connections for mobile user in case of dynamic and static behavior compared to the PF in terms of the throughput and delay. The throughput of all delay sensitive connections is above the MRR (50% of the requested BW), except for that of BE in which the MRR is zero. Also the sample connection hasn’t suffered from

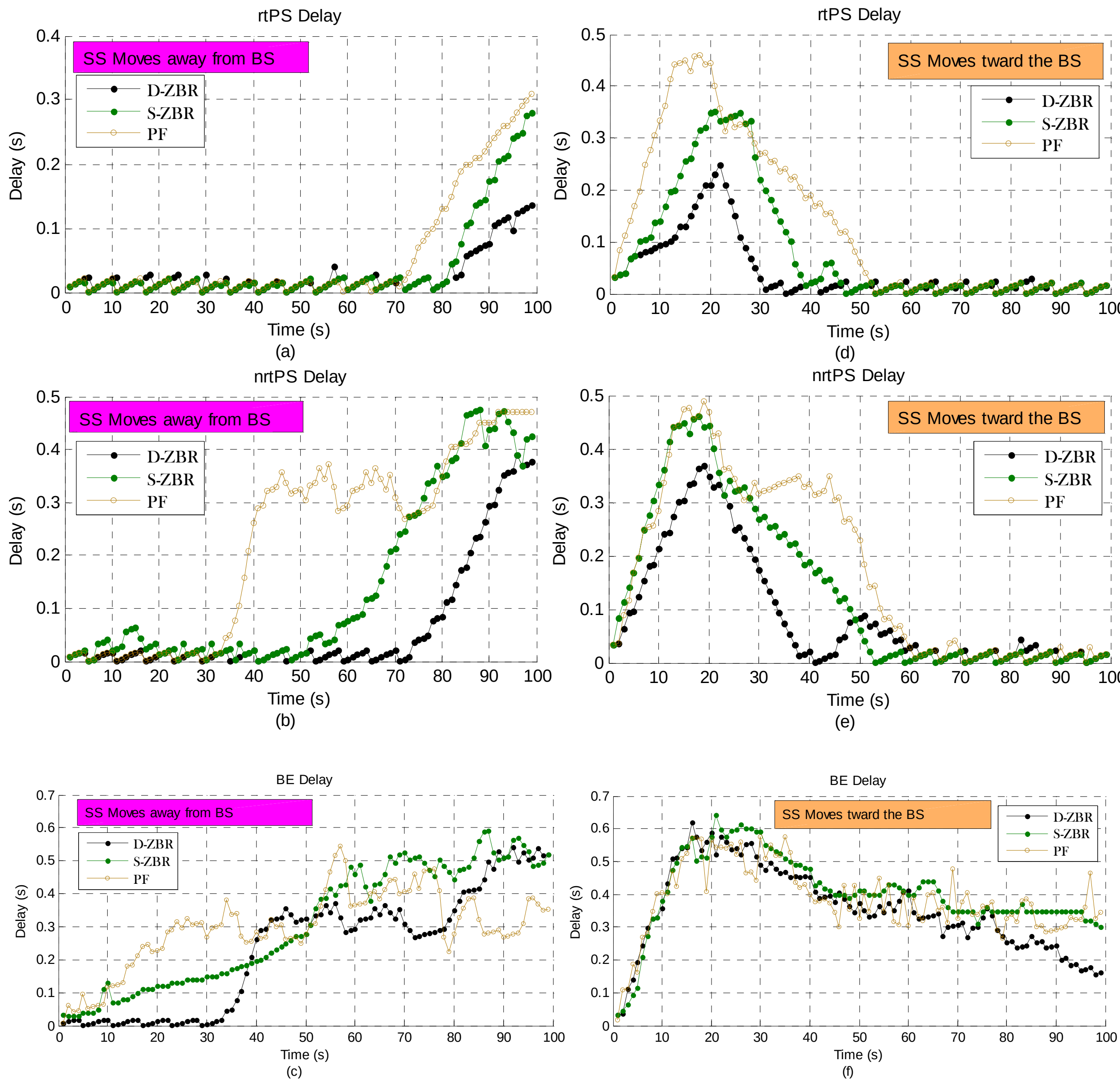

Figure 5-5 : (a), (b) and (c) rtPS delay, nrtPS delay, and BE delay respectively for mobile user who moves toward the BS. (d), (e) and (f) rtPS delay; nrtPS delay, and BE delay respectively for mobile user who moves away from the BS.

link degradation at the cell border in case of dynamic and static modes of the ZBR-3LHA compare to PF, this is due to the behavior of the ZBR function which works in accompany with other phases and guarantee fairness and stabilization across the trip. Another point is the Dynamic behavior of the proposed algorithm showed the least variance in terms of throughput and delay than static and PF. PF shows the highest variance compare to the other algorithms. Figure (5-5) Visualize the average delay over time for each CoS for the dynamic and static behavior compared to PF algorithm. Figure (5-5) (a), (b) and (c) show Dynamic ZBR-3LHA outperforms the static and the PF for rtPS and nrtPS. On the contrary, PF outperforms our proposed algorithm in term delay and throughput for BE CoS.

Figure (5-5), (d), (e) and (f) shows delay increases at the beginning of the simulation time until it reaches a certain limit and stabilizes, this is due to the fact that at t=0 all queues belong to that user are empty this explains why the delay is zero then queues start to build up as time goes by considering that the mobile user is getting farther from the BS through the simulation time.

5.4 Conclusion

This work has presented two versions of the ZBR-3LHA; static and dynamic for UL scheduling in Mobile WiMAX that targets specific requirements of each scheduling service while enforcing an opportunistic cross-layer approach in order to meet QoS requirements and maximize the overall system throughput. MRR granted to all connections to maintain QoS requirements. The ZBR function deals with different Slots rates for different QAM modulation zones to achieve fairness among different connections in the cell up to a certain level. Cross layer approach used to distribute the left over Slots to increase the overall network throughput and to minimize the delay variations.

It's evident that the Dynamic ZBR-SP-AMC shows stability over the life time of a connection as compare to the static and PF algorithms. The simulation results indicate that the performance of the Dynamic ZBR-3LHA outperforms that of the static and PF scheduling algorithms in terms of overall network throughput and delay since the Dynamic ZBR-3LHA switches the base line according to the zone that has the highest traffic demand, which optimizes the throughput accordingly and reduces the average end to end delay of the overall system.

Chapter 6

A Novel Combined Cross Layer with Dynamic Slot Allocation (Fully integrated PHY-MAC).

6.1 Introduction

OFDMA is one of the most robust modulation schemes against inter symbol interference (ISI) in which the total channel Bandwidth (BW) is divided into many orthogonal narrowband sub-channels. The serial high-rate data stream is converted into several parallel low-rate streams, each modulated on a different subcarrier.

Data is mapped into Slots; each Slot is formed by one OFDMA symbol and a certain number of frequency carriers named sub-channel. WiMAX frame consists of certain number of Slots with variable rates; the rate of each Slot is a function of user's SNR which depends on channel condition and his location.

Prior to the assigning of Slot for different Subscribers, user's SNR should be considered and channel frequency response should be estimated. In packet based communications, block of data is preceded by a number of training symbols of known data which usually used for synchronization and equalization purposes.

This chapter presents a novel dynamic Slot allocation as the third phase of the fully integrated radio resource mangament algorithm explained through this thesis. In this phase, the scheduler Adaptively allocate every slot on the optimum sub-channel that belongs to each user. It uses the information of the transmission guide matrix at the PHY layer , the former consists of average scanned pilot SNR carriers for each Slot on frame bases; more specifically for the sub-channel that form the Slot and every connected user in the cell.

The third phase decides the optimum sub-channel for a user to have his slots assigned on at the PHY level while the ZBR-3LHA located at the MAC Layer decides the number of Slots needs to be granted for each connection on frame by frame bases.

6.2 State of the Art

Exploit multi-user diversity on frequency and time domains in OFDMA system maximizes the network throughput; in such system each sub-channel is assigned to the user with the highest SNR for that sub-channel. Traditional way of estimating the Slot rate is to estimate the SNR as well as the rate for each frequency carrier who belongs to this Slot individually.

The former is very accurate way to calculate the exact Slot rate however its extremely computationally complex to obtain and impossible to implement; the reason is that the elapsed computation time required to calculate the impulse response for each carrier that form the Slot will exceed the WiMAX frame time, this will make the traditional way to calculate the capacity of each Slot impossible to implement and unpractical.

The algorithm should be able to estimate the impulse response for different sub-channels for each user in just a fraction of the WiMAX frame time, since it needs to MAP all users to the available sub-channels through the UP-MAP message for the next frame. If the scheduler exceeds the frame time to get these estimation then estimating the impulse response and the SNR for certain sub-channels will be useless. In this chapter a low-complex adaptive Slot allocation technique is proposed. The adaptive technique estimate the SNR by averaging the observable pilot SNR for each Slot, these pilot carriers are shifted each frame until they scan the whole Slot. the sub-channels are assigned to users according to their channel conditions, which is usually measured through channel state information (CSI) . The proposed work is analyzed and compared against the static algorithm in which; the sub-channels are assigned for SSs in a traditional way; first come first served. The static is unlike the dynamic way CSI never involve in the sub-channel allocation process.

Performance evaluation of the low complex algorithm shows the effect of dynamic subcarrier allocation in reducing fading effect by increasing the total network throughput over a frequency selective fading channel as compared to the static algorithm. In the Static Subcarrier and Slot Allocation (SSSA) algorithm the Slot is assigned for MSS under FIFO algorithm without returning to the transmission matrix, the DSSA outperformed the SSSA because the first takes into account that the capacity of each sub-channel is a function in the wireless channel impulse response; SNR for each frequency carrier is calculated by dividing the squared amplitude of the channel impulse response by the noise power.

6.3 Dynamic Subcarrier and Slot allocation (DSSA)

This section describes algorithm to determine average signal to noise ratio of a single frequency channel. The assumption of slow fading is a considered in which, no change occurs to the channel state conditions during the full time frame channel; impulse response is constant within a single frame. In another words there is no distinct differences between SNR of the same frequency within one frame at different OFDM time symbols.

Slot rate varies according to user's who gets it. Shannon's approach can be used to predict the capacity of each subcarrier based on the wireless channel variation gain which varies according to the location of every user (*i*) in the cell and the channel status. The rate of each sub-carrier (*f*) that is assigned to user (*i*) under the assumption of continuous use of it in the time domain is shown in (6-1), To simplify calculations a flat power allocation assumption is considered in which the total transmit power over each sub carrier is $\frac{P}{N}$, where *P* is total transmit power and *N* is the total number of subcarriers in the system, the total bandwidth of the system is *B*, and the band width of each subcarrier is $\frac{B}{N}$

$$\boldsymbol{R_{f,i} = \frac{B}{N} \log_2\left(1 + SNR_{f,i}\right)} \qquad (6\text{-}1)$$

The rate of each Slot can be defined by adding the rates of all subcarriers who form the Slot and dividing the sum by the total number of OFDM symbols in the frame, the reason is that the subcarrier rate is limited by the duration of one OFDM symbol time; this is expressed as

$$\boldsymbol{R_{PS,i} = \frac{B}{\gamma \times N} \sum_{n=1}^{m} \log_2\left(1 + SNR_{ps,i}\right)} \qquad (6\text{-}2)$$

Where m is the total number of subcarriers in each Slot and (γ) is the total number of OFDM symbols in each frame.

Equation (6-2) shows that each Slot in the system has a different rate that varies according to the user's SNR. This will raise a problem in two dimensions for the resource allocation process at the PHY layer level; the first is to determine which subcarrier is assigned to which Slot and the second which Slot is given to which user. The proposed bandwidth allocation algorithm deals with the aforementioned problems and provides a solution based on exploiting the multi-user diversity by dynamically allocating these Slots among different users; it is hardly ever that a sub-channel is in deep fade for all the users in the cell concurrently.

In this work two new techniques are introduced to compute the average Slot's rate that's different from the traditional method (TM) which calculates the Slot rate by adding the rate of all subcarriers that form the Slot. The first technique is Pilot- Selective Average (PSA) in which system determines average SNR within each frequency sub-channel (k) for each user (i) as in (6-3) and (6-4).

In order to minimize computations only pilot subcarriers are considered. For simplicity, we assume limitation of x measurements of SNR; corresponding to x pilot sub-carriers within one frequency sub-channel. For a sub-channel k, SNR is determined as follows:

$$\boldsymbol{n_k = xk} \qquad (6\text{-}3)$$

$$\boldsymbol{SNR_{user(i)}^{channel(k)} = \frac{k}{n_k} \sum_{m=1+n_{k-1}}^{n_k} \left| H\left(f_{\left\lfloor \frac{y}{x} \right\rfloor m}\right) \right|^2 / N\left(f_{\left\lfloor \frac{y}{x} \right\rfloor m}\right)} \qquad (6\text{-}4)$$

Where H (f) is the channel impulse response, N (f) is the noise, $n_0 = 0$*, x: is number of pilot subcarriers in each frequency sub-channel, y: is the total number of frequency carriers grouped in one channel, and k is the frequency block number.*

PSA computes the average Slot SNR for every $\left\lfloor \frac{y}{x} \right\rfloor$ frequency within given frequency sub-channel which is not sufficient enough and better approximations must be implemented; the reason for that is one of the considered frequencies might be sufficiently degraded within a channel for a particular user which will increase the probability of error in estimating the average Slot SNR and in sequence the average Slot rate; accordingly making the selected average SNR not viable and forcing system to allocate resources for a certain user on the wrong channel.

This issue can be avoided if selected frequency for calculating average SNR is dynamically allocated as the second technique Pilot Scan Selective Average (PSSA). This means that at every frame different frequencies are chosen for average SNR calculation as in (6-6) and (6-7) this helps to eliminate the natural random fading in wireless channel and improves the fairness factor among different Slots. In the next expressions we consider dynamic frequency scanning approach; in which the PSSA-SNR within a sub-channel of the different frequencies set is computed on frame by frame bases.

$$\boldsymbol{let} \quad \boldsymbol{i} = \left\lfloor \frac{y}{x} \right\rfloor \tag{6-5}$$

$$\frac{\boldsymbol{S}}{\boldsymbol{N_{kj}}} = \frac{\boldsymbol{k}}{\boldsymbol{n_k}} \sum_{\boldsymbol{m}=\boldsymbol{1}+\boldsymbol{n_{k-1}}}^{\boldsymbol{n_k}} \left| \boldsymbol{H}\left(\boldsymbol{f_{im-(j-1)}}\right) \right|^{\boldsymbol{2}} / \, \boldsymbol{N}\left(\boldsymbol{f_{im-(j-1)}}\right) \quad \boldsymbol{for\, j} \le \boldsymbol{i} \tag{6-6}$$

$$\frac{\boldsymbol{S}}{\boldsymbol{N_{kj}}} = \frac{\boldsymbol{k}}{\boldsymbol{n_k}} \sum_{\boldsymbol{m}=\boldsymbol{1}+\boldsymbol{n_{k-1}}}^{\boldsymbol{n_k}} \left| \boldsymbol{H}\left(\boldsymbol{f_{im-\left(j-i\left\lfloor \frac{j}{i} \right\rfloor-1\right)}}\right) \right|^{\boldsymbol{2}} / \, \boldsymbol{N}\left(\boldsymbol{f_{im-\left(j-i\left\lfloor \frac{j}{i} \right\rfloor-1\right)}}\right) \quad \boldsymbol{for\ j} > \boldsymbol{i} \tag{6-7}$$

Where; j is the frame of performed calculations.

Equation (6-6) will be considered as initialization phase, where with the increase number of sequenced frames equation (6-7) will take over.

6.4 Dynamic Sub-Channel Allocation Mechanism

Frequency channel allocation phase provides the foundation for the implementation of the dynamic frequency scanning technique. We need to distinguish between two expressions the first is the resource allocation phase which is managed by the algorithm in the MAC Layer; the former determines the number of Slots have to be granted for each connection belong to a specific connection or user. Second is the channel allocation phase which will cooperate with the upper MAC Layer phase to form the fully integrated Slot allocation RRM.

The transmission guide matrix is set in the sub-channel allocation phase, the former contains the pilot selected Average PSA-SNR for all sub-channels for each individual user. This matrix will be updated on frame basis and will work as a guide for the resource allocation phase in the upper layer (Mac) Figure (6-1) and (6-2).

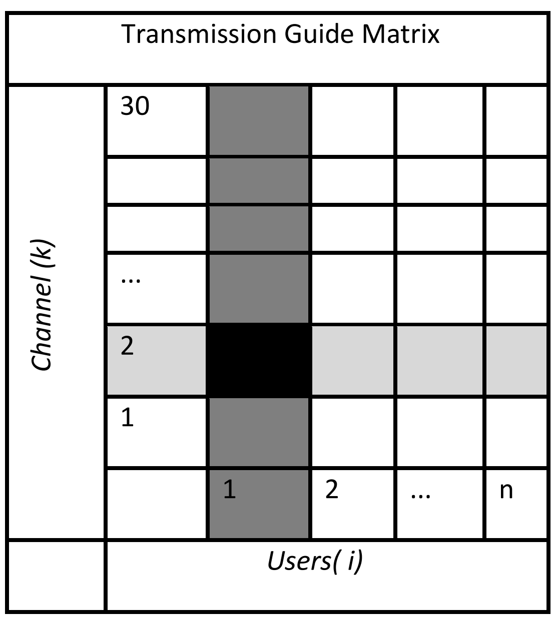

Figure 6-1 : Transmission Guide Matrix

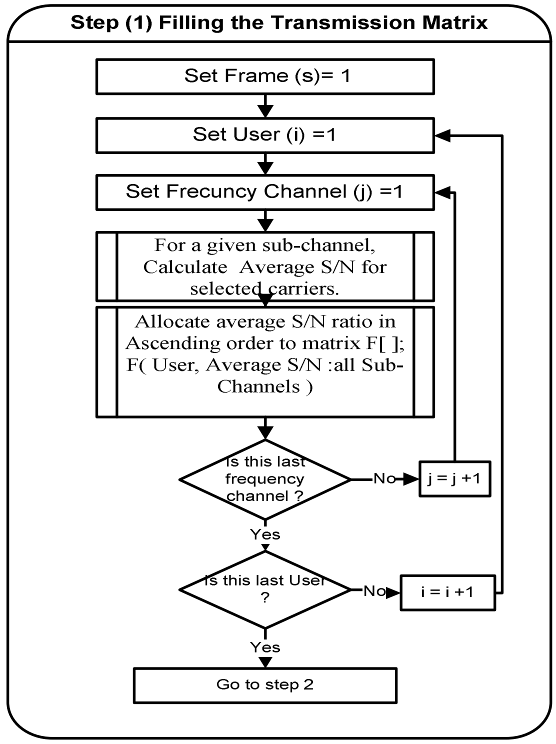

Figure 6-2: Filling the Transmission Guide Matrix

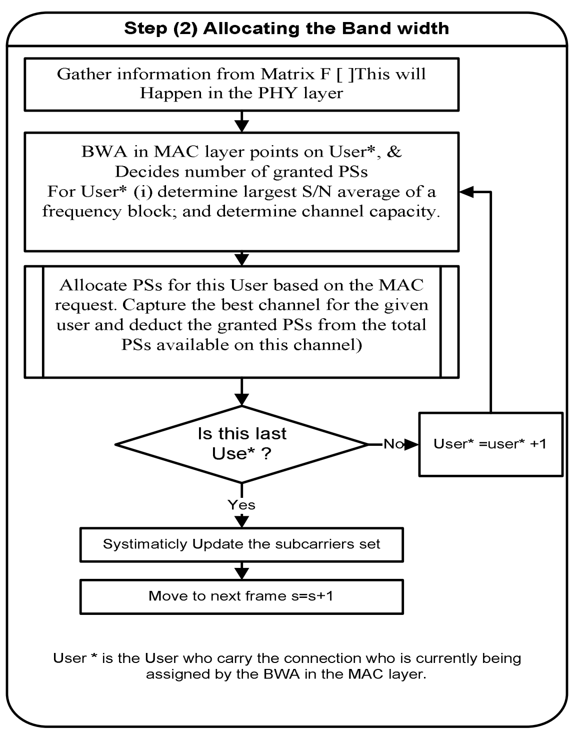

Figure 6-3 : Interaction between TGM and the BWA

6.5 Implementation and Results Discussion

The setup is relatively not complex. The MAC layer "ZBR-3LHA" decides the total number of Slots that should be granted to each connection based on the bit rate request and CoS. The system must determine the highest SNR channel for each user on frame by frame basis and to determine if the allocated sub-channel meets the QoS required by the user which is implemented by two dimensional matrixes: users (i) and frequency sub-channel (k). Once a frequency sub-channel within an OFDM symbol time is allocated to a user within the frame, other users cannot use this Slot within the frame.

Figure (6-2) and (6-3) explain how DSSA works. Figure (6-2) shows the sequence of filling the transmission matrix while Figure (6-3) shows the interaction between PHY layer in which the SNR for each Slot is estimated and the MAC layer which act as the master mind to guarantee the QoS and to maintain service differentiation among different connections in different modulation zones.

The ZBR-3LHA at the MAC layer determines the number of Slots that should be granted for each connection while the DSSA algorithm chooses the best sub-channel that these Slots need to be granted on for every user.

6.6 Performance and Evaluation

The performance of DSSA algorithm is compared against the static approach algorithms; both of them use the same MAC layer scheduler.

The static does not consider the gain of the sub-channel that forms the Slot in the bandwidth assigning process. The algorithm performance was evaluated by measuring the average class throughput and delay for two mobile users as a random sample. Both users move horizontally a cross the cell by a certain speed in opposite directions; first user (SS1) starts it's trip at the cell border (t=0 sec) and moves toward the BS to the cell center as in Figure (6-4) & (6-5), (a, b, c), by the end of the simulation time (t=100 sec), it reaches the cell center.

The second user (SS2) starts it's trip at the cell center (t=0 sec) and moves away from the BS as in Figure (6-4) & (6-5) (d, e, f), by the end of the simulation time the user reaches the cell border at (t =100 sec).

Figure (6-4) (a), (b) and (c) shows the average throughput of these connections belongs to SS1. As can be seen the proposed DSSA throughput outperforms the static one, especially for rtPS and BE. However, for the nrtPS both algorithms show the same performance. Moreover the amount of the average throughput degradation for SS2 in Figure (6-4) (e) and (f) in DSSA is much less than the Static one.

Figure (6-5) shows that the DSSA outperforms the static in terms of delay. (b) and (c) shows the average delay for both classes nrtPS and BE for SS1, this can be interpreted as follows; at the beginning SS1 is at the cell's edge in the lowest modulation order, but also traffic queues are empty the delay of different CoS is increasing until it reaches the max between t=10 and t=20 sec, with considering that the mobile user is also getting closer to the BS delay starts decreasing until it reaches zero as SS1 reaches the BS.

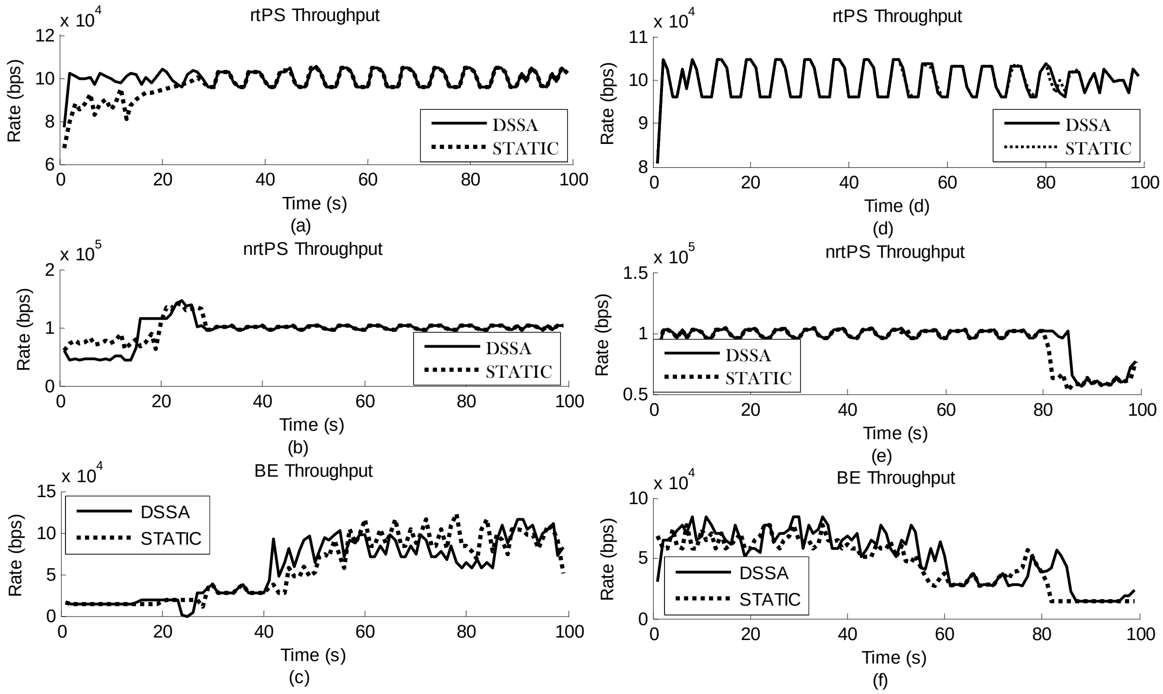

Figure 6-4 : (a), (b) and (c) rtPS throughput, nrtPS throughput, and BE throughput respectively for mobile user who moves toward the BS. (d), (e) and (f) rtPS throughput; nrtPS throughput, and BE throughput respectively for mobile user who moves away from the BS.

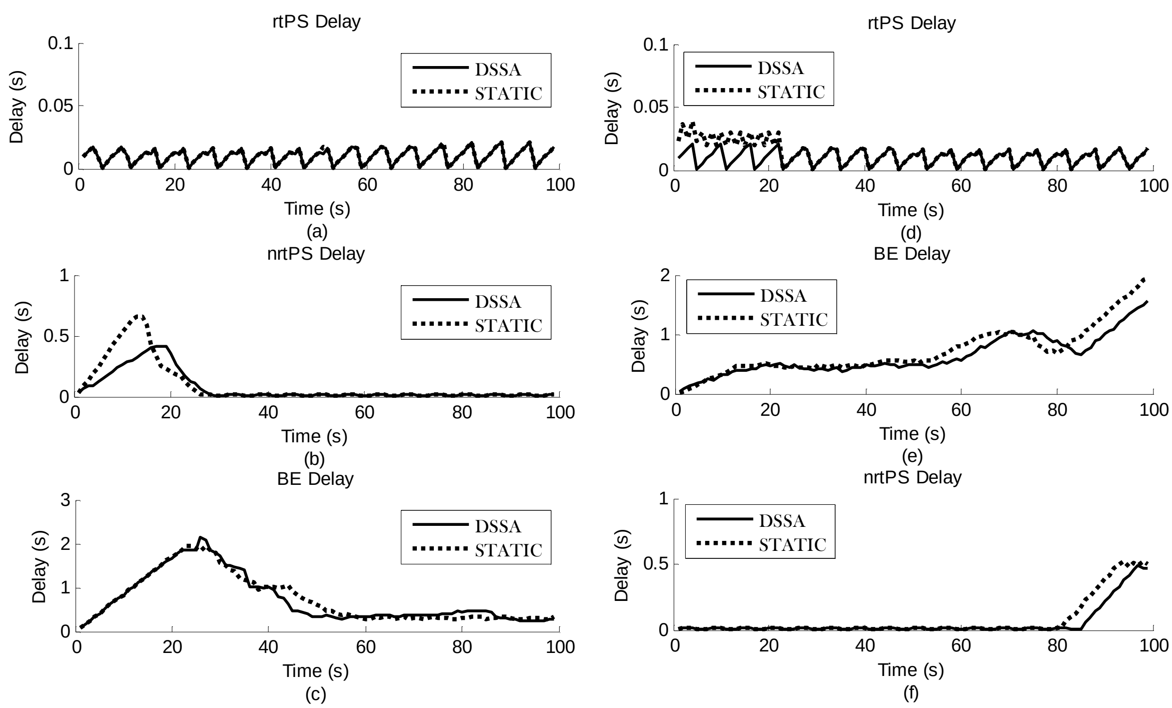

Figure 6-5: (a), (b) and (c) rtPS delay, nrtPS delay, and BE delay respectively for mobile user who moves toward the BS. (d), (e) and (f) rtPS delay; nrtPS delay, and BE delay respectively for mobile user who moves away from the BS.

6.7 Error Analysis

In this section, an error analysis method is introduced to address the correctness of the proposed work of getting Slot rate based on estimating the average Slot-SNR for the number of carriers that form the Slot. The former will be compared and referenced to the traditional way of getting the Slot-rate which estimates each subcarrier SNR individually subsequently calculate the average SNR for all subcarriers that shape the Slot, after that Shannon formula is applied to get the Slot rate.

Numerical errors are computed using the MATLAB. Two approaches (selective and scanning) are measured and compared to the traditional way under typical channel conditions through the assumption that the error in estimating the Slot rate in traditional way is zero, Frequency selective fading channel is modeled considering the parameters were introduced in chapter 3. Channel is modeled as slow fading in which the channel stay stable and never change status during the frame time period.

programming elapsed time (computation time) is ignored in formulating the traditional way technique to calculate the Slot rate since our main concern is not to prove this method practically but is to measure the difference in Slot rate between the traditional and proposed techniques.

Long computation time is usually required for the traditional technique of getting the Slot rate. The reason is the need to predict the channel impulse response for each carrier in the frequency domain and then calculate the SNR for all carriers for all users in the cell on frame by frame base then the capacity of each carrier is measured individually then added together to get the capacity of the individual Slot.

The next section will present the comparison between the two proposed techniques (selective and scanning) against the reference (traditional way) in a Simple scenario in which SNR of five pilot carriers are considered and one hundred frames are simulated.

6.7.1 Selective approach

In this approach predetermined signal to noise ratio carriers (pilots) are considered, the locations of those carriers are fixed in each Slot and never change from frame to frame. The error percentage in this approach is calculated as (6-8)

$$\boldsymbol{Error}\% = \frac{\sum_{j=1}^{\alpha}\frac{k}{n_k}\sum_{m=n_{k-1}}^{n_k}\frac{\left|H\left(f_{\left\lfloor\frac{y}{x}\right\rfloor m}\right)\right|^2}{N\left(f_{\left\lfloor\frac{y}{x}\right\rfloor m}\right)} - \sum_{j=1}^{\alpha}\frac{1}{y}\sum_{m=1}^{y}\frac{|H(f_m)|^2}{N(f_m)}}{\sum_{j=1}^{\alpha}\frac{1}{y}\sum_{m=1}^{y}\frac{|H(f_m)|^2}{N(f_m)}}\; \boldsymbol{x}\, \mathbf{100}\% \qquad (6\text{-}8)$$

Where α - total number of frames considered in simulation. First portion of numerator is the average of selected pilots SNRs throughout all frames in the experiment and the second portion as well as the denominator is the average of all Slot sub-carriers SNRs throughout the simulation time. Entire function is then multiplied by 100% to reflect the error percentage.

6.7.2 Selective Scanning Approach

In this approach the pilot locations changes within the Slot on frame bases. The error is calculated via two steps according to the frame number as it shown in (6-9) and (6-10).

If $\left\lfloor\frac{y}{x}\right\rfloor < \alpha$ the error percentage is:

$$Error\% = \left(\frac{\left[\sum_{j=1}^{\left\lfloor \frac{y}{x} \right\rfloor} \frac{k}{n_k} \sum_{m=1+n_{k-1}}^{n_k} \frac{\left| H\left(f_{im-(j-1)}\right)\right|^2}{N\left(f_{im-(j-1)}\right)} + \sum_{\left\lfloor \frac{y}{x} \right\rfloor}^{\alpha} \frac{k}{n_k} \sum_{m=1+n_{k-1}}^{n_k} \frac{\left| H\left(f_{im-\left(j-i\left\lfloor \frac{j}{i} \right\rfloor-1\right)}\right)\right|^2}{N\left(f_{im-\left(j-i\left\lfloor \frac{j}{i} \right\rfloor-1\right)}\right)} \right]}{\sum_{j=1}^{\alpha} \frac{1}{y} \sum_{m=1}^{y} \frac{|H(f_m)|^2}{N(f_m)}} - 1 \right) x\ 100\% \quad (6\text{-}9)$$

where numerator in square brackets is the summated average for the scanning approach.

Like, (6-6) and (6-7). First portion summates signal to noise average starting from first frame to $\left\lfloor \frac{y}{x} \right\rfloor$ frame, so that to meet required conditions shown in (6-6). Second portion in square brackets, summates frames from $\left\lfloor \frac{y}{x} \right\rfloor$ to α - total considered frames in experiment, thus fulfilling requirements of (6-7). It must be noted here that in a condition where $\left\lfloor \frac{y}{x} \right\rfloor > \alpha$ this equation will be meaningless. For example in the appendix B, set up shows $\left\lfloor \frac{y}{x} \right\rfloor = 10$, now if we wish to simulate only nine frames $\alpha = 9$ different equation must be used. Equation 9 will not be able to address requested requirements in simulation, thus for this unique cases we must calculate percent of error a bit differently. To calculate percent of error for the scanning approach where $\left\lfloor \frac{y}{x} \right\rfloor > \alpha$ we must use following equation:

$$\boldsymbol{Error\%} = \left(\frac{\left[\sum_{j=1}^{\alpha} \frac{k}{n_k} \sum_{m=1+n_{k-1}}^{n_k} \frac{\left| H\left(f_{im-(j-1)}\right)\right|^2}{N\left(f_{im-(j-1)}\right)} \right]}{\sum_{j=1}^{\alpha} \frac{1}{y} \sum_{m=1}^{y} \frac{|H(f_m)|^2}{N(f_m)}} - \mathbf{1} \right) \boldsymbol{x\ 100\%} \quad (6\text{-}10)$$

First portion summates signal to noise average starting from first frame to α frame, so that to meet required conditions shown in (6-6).

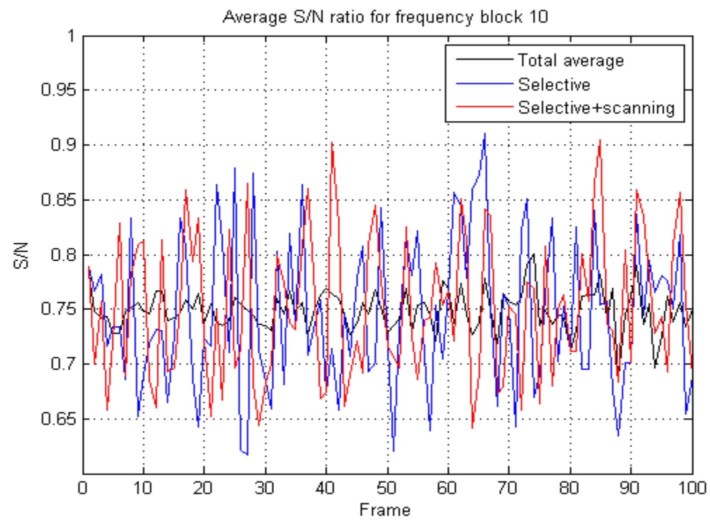

Figure 6-6 : Average SNR for Sub-channel number 10

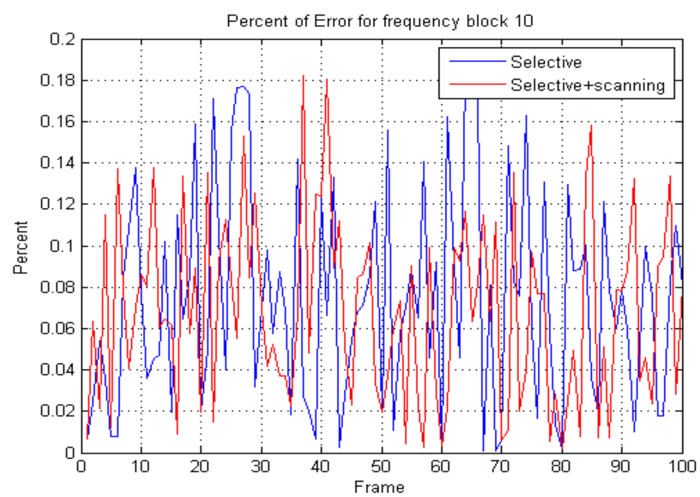

Figure 6-7 : Error % for Sub-channel number 10

6.7.3 Error Analysis Discussion and Conclusion

Over All Average parameters

Traditional way = 0.6743
Selective average = 0.7480
Scanning average = 0.6676

Percent of error for selective = 10.94%
Percent of error for scanning = 0. 99%

From the final error percentage above, the scanned selective approach performed less error than the selective one. The error analysis was performed under the same conditions and parameters for both techniques. The analytical results reflects the simulated plots results Figure (6-6) and

(6-7). In the selective approach the results can be analyzed considering a simple scenario that first five cut-off frequencies of Slot are overlooked. In this situation system assumes signal to noise ratio to be of the first pilot signal (carrier number ten), which in fact is incorrect, thus system overestimates the available bandwidth of the channel. In the system with predetermined pilot channels, where system is blind to cut-off frequencies outside the pilot channels. This will lead to over estimating the Slot capacity and will let the scheduler in the upper layer send more information than the Slot can handle result in loss of information.

On the other hand in the scanned selective, cut-off frequencies will be determined and closer estimates of signal to noise average will take place. In the simulation results provided above we consider average signal to noise ratio of one frequency block averaged over hundred frames.

As expected scanning approach is much closer to total average Slot SNR, thus approximating the channel closer. Finally, Percent of error for selective, predetermined method, is almost 11% where for the scanning only 1%.

6.8 Chapter Summary

Simulation results indicate that the performance of the DSSA outperforms that of the conventional algorithms in terms of network throughput and delay. The advantage of the proposed technique is very practical to implement, the elapse time to fill the transmission guide matrix is much less than the frame time, on the other hand the traditional method (TM) the time consumed to calculate the impulse response for all sub-channels is much more than the frame time, which makes this method very complex and resource consuming and impossible to implement.

Nonetheless, the proposed scheme DSSA suffers from lake of accuracy in estimating the capacity of the Slot which is obtained through the error analysis that we studied but for the space limitation it can’t be included in this paper, but will be presented in future work.

The error analysis shows that PSSA-SNR has less error in estimating the Slot capacity than PSA-SNR. The probability of error in estimating Slot capacity is about$\mp 10\%$, while the PSSA-SNR has about $\mp 1 - 2\%$ margin of error.

The error margin has a direct effect on over and under estimating the Slot capacity; under estimating the Slot capacity will lead to waste available band width resources, on the other hand, over estimating the Slot capacity will cause data loss. This shortage in the algorithm can be overcome by salvation mechanism, which will be discussed separately with Error analysis in future work

Chapter 7

Overall Conclusion and Future Work

7.1 State of the Art

Several studies have been done on uplink band width allocation in WiMAX, they followed different strategies; some center the attention on having a fair data rate among connections,

Some focus on maximizing the overall network throughput, some use priority queuing to priorities delay sensitive applications. However none of the aforementioned work focuses on stabilizing delay and throughput connections due to wireless mobility. Our proposed works spot the light on this issue of connection delay and throughput stabilization.

Our work is based on fully cooperative integration between PHY and MAC layers to exploit the fast link adaption with different QAM modulation in addition to the state of the wireless channel and QoS requirements to reach the best performance of scheduling the dynamic shared uplink channel.

Our fully integrated end to end algorithm was designed on a number of phases implemented in the MAC layer and cooperated with the PHY layer, each has its own function, the first was to grantee the connection QoS, the second is provide some kind of fairness which will cause connection stability, and will set an adaptive capacity threshold to prevent any connection from occupying the available band width

The proposed scheme estimates the Slot capacity according to the user channel conditions at the PHY layer level, as a result an accurate number of Slots required to maintain a certain connection is obtained. Zone balance ratio (ZBR) function is introduced to cope with the variation of the characteristic of the Slot capacity due to change in the fast link adaptive or wireless channel conditions.

The ZBR function stabilizes the ratio of any connection when the user moves across different QAM regions. Transmitter has to change modulation technique due to power drop or SNR, the connection with ZBR function is resistance to channel variation.

7.2 Work Summary and Conclusion.

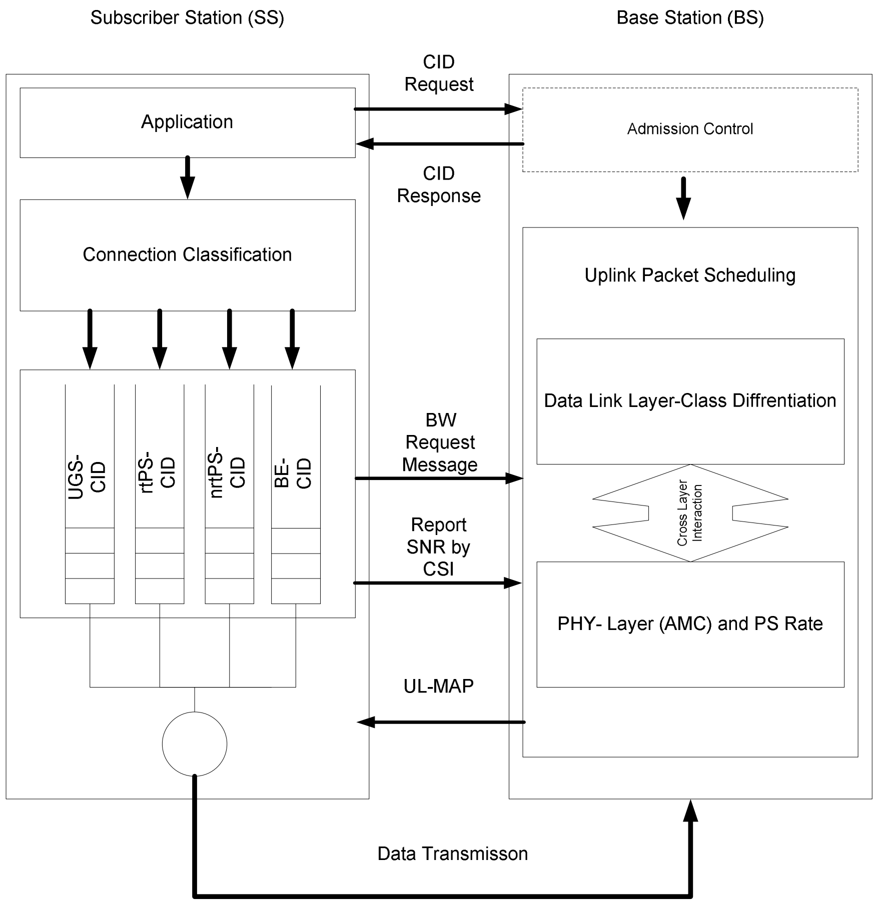

Figure 7-1 : Interaction between SS and RRM module for resource allocation & Functionality Integration between PHY and MAC layer in the making the decision

This thesis has addressed the important problem of Radio Resource Management (RRM) and Quality of Service (QoS) support for the emerging 4G broadband wireless access technologies. Specifically, this thesis has examined the technical feasibility and assessed the performance analysis for implementing an integrated framework for uplink (UL) RRM techniques in mobile WiMAX systems that holistically address many of the important WiMAX outstanding issues that have received little or no attention in the literature. A set of novel UL RRM functional modules, which form the foundation for the integrated framework, including Admission Control (AC), Packet

Scheduler (PS), Dynamic Bandwidth Allocation (DBA), and subcarrier allocation and mapping (SAM) are developed for mobile WiMAX.

The integrated RRM framework is enabled via introducing two mechanisms. First, the functionalities of all RRM modules (AC, PS, and DBA) at the MAC layer are first integrated. The QoS-aware scheduling decision (obtained from the unified RRM scheduling modules) at the MAC layer is then integrated with the channel state information and corresponding opportunistic AMC schemes as well as the SAM at the PHY layer. Thus, the proposed integrated RRM framework supports a fully integrated PHY-MAC layer strategy, which is the key for obtaining optimum (or near optimum) scheduling and resource allocation decisions each cycle.

The implementation of the proposed RRM framework proceeded in three sequential phases. Each succeeding phase builds upon previous phase(s) in order to address its shortcomings and enhance its performance. All RRM PS and DBA algorithms and schemes presented in the three phases utilize the OFDMA-based two-dimension time-frequency Slot assignment. Thus, the set of RRM PS and DBA algorithms presented in the third phase establish the complete envisioned integrated RRM framework for Mobile WiMAX.

The main salient features of the proposed RRM framework are:

1. QoS support mechanisms are inherently built into the main RRM's functional building blocks (AC, PS, and DBA) to guarantee that QoS commitments and control are met under normal network and traffic scenario as well as under congested traffic and network degradation scenarios. Thus, AC, PS, and DBA algorithms are all QoS-aware.

2. Because the developed RRM's AC and scheduling algorithms are all housed in the BS with fast and easy access to WiMAX air interface measurements and monitoring, a complete and efficient cross-layer optimization is implemented.

3. In addition to the typical network traffic load as well as differentiated QoS requirements, the dynamic resource allocation strategy takes also into account Link Adaptation (LA) including AMC and time and frequency selective fading of the wireless channel such that the multiuser diversity is exploited by assigning each user to the resources which exhibit optimum conditions for that user.

It is shown proposed integrated RRM framework provide optimum or close to optimum UL scheduling and resource allocation strategies that meet the typical mobile WiMAX mandatory set of QoS parameters and requirements associated with each CoS including CBR streaming applications; while concurrently striking a balance between maintaining nearly a constant throughput per every CBR streaming connection in the network, fairness among all mobile users irrespective of their channel conditions and locations (throughput fairness), and the overall system capacity. These results were obtained under the most stringent QoS requirements scenario in which the concept of CBR applications was generalized by assuming that all CoSs supported by the IEEE 802.16e standard strive to maintain a fixed data rate throughout the entire life of the connection.

The results presented in this thesis are obtained through extensive computer simulations and modeling using substantial system simulator development, which is carried out during the course of this work. This includes both mathematical modeling considerations as well as software design, implementation, and testing. To ensure the viability of the detailed performance analysis presented in this work, the system parameters used here to assess the overall performance of the proposed RRM framework are identical to the performance evaluation parameters specified in Mobile WiMAX system evaluation documents and WiMAX profiles.

7.3 Topics for Future Research

This study is done for delay tolerable data traffic, for one cell only, the ZBR-3LHA should be studied for more than one cell, it would be quite interesting to study the performance of the proposed algorithm for new admitted users and hand off users when hand off between different cells take place.

Moreover, load control issues; when different connections belongs to different users have a gigantic verities of requested bit rate and their effect on the algorithm behavior especially on the delay sensitive applications. Taking into consideration that due to mobility the amount of resource required for certain user (connection) could increase or decrease rapidly according to his current location and channel condition. It is possible that due to the user mobility in specific direction will cause some other user to be not admitted because the available resources are not going to be valid to accommodate their minimum QoS requirement.

Assigning the Slots for users based on their current location in the user's table is not fair, usually the first user in the table have the best available choice from shared resources and will get Slots with the height SNR compared to others; consequently will have the highest Slot rate. Another adaptive algorithm could be implemented to deal with this matter to provide fairness between users and could also include different metrics, like the mount of requested Slots and the traffic requested type.

Additionally, our proposed dynamic channel allocation will save plenty of time in filling the guide matrix but it will not going to be accurate 100% due to over and under estimation of Slots rate more error analyses can be done for the case of over and under estimating the Slot rate.

Integration with optical networks and providing end to end Resource Management Protocol (RMP) is and interested subject to extend this work, attention has to be paid to the major difference in cycle time for both technologies.

Appendix A

Practical Example for Calculating the Average SNR in DSSA

Consider frequency block #3 ($k = 3$) where $y = 50$ carriers within one frequency block, spread from frequency number 100 to 150 with $x = 5$ pilots.

$$n_k = xk = 15 \qquad (eq.3a)$$

Using *eq. 3a* $\quad n_k = 15$

$$\frac{S}{N_k} = \frac{k}{n_k}\sum_{m=n_{k-1}}^{n_k}\left|H\left(f_{\lfloor\frac{y}{x}\rfloor m}\right)\right|^2 / N\left(f_{\lfloor\frac{y}{x}\rfloor m}\right) \qquad (eq.3b)$$

Using *eq. 3b*

$$\frac{k}{n_k} = \frac{3}{15} = \frac{1}{5}$$

Division to acquire average.

$$\left\lfloor\frac{y}{x}\right\rfloor = floor\ \left(\frac{y}{x}\right) = floor\left(\frac{50}{5}\right) = 10$$

We will consider every tenth frequency (e.i. f_{110}, f_{120}, f_{130}, f_{140}, f_{150})

The lower limit in our summation will be.

$$1 + n_{k-1} = 1 + x(k-1) = 1 + 5 * 2 = 11$$

Now we are fully equipped to calculate average signal to noise ratio.

$$\frac{S}{N_k} = \frac{k}{n_k}\sum_{m=1+n_{k-1}}^{n_k}\left|H\left(f_{\lfloor\frac{y}{x}\rfloor m}\right)\right|^2 / N\left(f_{\lfloor\frac{y}{x}\rfloor m}\right) = \frac{1}{5}\sum_{m=11}^{15}|H(f_{10m})|^2 / N(f_{10m})$$

$$\frac{S}{N_3} = \frac{1}{5}\left(\frac{|H(f_{110})|^2}{N(f_{110})} + \frac{|H(f_{120})|^2}{N(f_{120})} + \frac{|H(f_{130})|^2}{N(f_{130})} + \frac{|H(f_{140})|^2}{N(f_{140})} + \frac{|H(f_{150})|^2}{N(f_{150})}\right)$$

Appendix B

Practical Example shows DSSA and Frame sequence Number.

Consider frequency block #3 ($k = 3$) where $y = 50$ carriers within one frequency block, spread from frequency number 100 to 150 with $x = 5$ pilots and frame number $j = 7$.

$$n_k = xk = 15 \qquad (eq.3a)$$

Using *eq. 3a* $n_k = 15$

$$\left\lfloor \frac{y}{x} \right\rfloor = floor\left(\frac{y}{x}\right) = floor\left(\frac{50}{5}\right) = 10$$

Yet again we will consider every tenth frequency however now we must take in to account frame number.

Frame number j is less than i (e.i 7 < 10), therefore we must use equation 5a.

$$\frac{S}{N_{kj}} = \frac{k}{n_k} \sum_{m=1+n_{k-1}}^{n_k} \left|H\left(f_{im-(j-1)}\right)\right|^2 / N\left(f_{im-(j-1)}\right) \quad for\ j \le i \qquad (eq.5a)$$

Thus,

$$im - (j-1) = 10m - (7-1) = 10m - 6$$

We will consider every tenth frequency (e.i. f_{104}, f_{114}, f_{124}, f_{134}, f_{144})

The lower limit in our summation will be.

$$1 + n_{k-1} = 1 + x(k-1) = 1 + 5 * 2 = 11$$

Now we are fully equipped to calculate average signal to noise ratio.

$$\frac{S}{N_{kj}} = \frac{k}{n_k} \sum_{m=1+n_{k-1}}^{n_k} \left|H\left(f_{im-(j-1)}\right)\right|^2 / N\left(f_{im-(j-1)}\right) = \frac{1}{5} \sum_{m=11}^{15} |H(f_{10m-6})|^2 / N(f_{10m-6})$$

$$\frac{S}{N_3} = \frac{1}{5}\left(\frac{|H(f_{104})|^2}{N(f_{104})} + \frac{|H(f_{114})|^2}{N(f_{114})} + \frac{|H(f_{124})|^2}{N(f_{124})} + \frac{|H(f_{134})|^2}{N(f_{134})} + \frac{|H(f_{144})|^2}{N(f_{144})}\right)$$

Appendix C

Pseudo Code for Cross Layer Cross Hierarchal Algorithm

$Tot\_PS^{frame_x}_{requested} =$

$\sum_{i=1}^{a} \frac{\sum_{j=1}^{k} Connection\_BW\_request_{i,j}}{PS\_rate_{QAM64}} + \sum_{i=1}^{b} \frac{\sum_{j=1}^{k} Connection\_BW\_request_{i,j}}{PS\_rate_{QAM16}} + \sum_{i=1}^{c} \frac{\sum_{j=1}^{k} Connection\_BW\_request_{i,j}}{PS\_rate_{QPSK}} +$

$\sum_{i=1}^{d} \frac{\sum_{j=1}^{k} Connection\_BW\_request_{i,j}}{PS\_rate_{BPSK}}$

If($Tot\_PS^{frame_x}_{requested}$ < $Tot\_PS^{frame_x}_{available}$)

(1)

Grant Requested BW to everyone in the cell (2)

Else

{

Grant MRR to everyone (3)

$Remaining^{PS}_{f} = Total^{PS}_{f} - \sum_{i=1}^{m}\sum_{j=1}^{k} \text{Ceiling}\left(\frac{MRR_{i,j}}{PS_{rate_{QAM(E)}}}\right)$

for(i=0; i<Num_MU;i++)

{

Grant requested ugs BW for all MU in the cell

Decrement $Remaining^{PS}_{f}$ by the number of Slots granted

}

Zone = 4; //(4 is QAM 64, 3 is QAM 16, etc)

while(zone>0)

{

for(i=0; i<Num_MU;i++)

{

if((MU(i)_zone == zone) && (MU(i)_rtPS has traffic) &&($Remaining^{PS}_{f}$ >= 0))

{

Grant BW for this user's rtPS connection up to $Remaining^{PS}_{f}$;

Decrement $Remaining^{PS}_{f}$ by the number of Slots granted

}

if((MU(i)_zone == zone) && (MU(i)_nrtPS has traffic) &&($Remaining^{PS}_{f}$ >= 0))

{

Grant BW for this user's nrtPS connection up to $Remaining^{PS}_{f}$;

Decrement $Remaining^{PS}_{f}$ by the number of Slots granted

}

}

zone --;

}

Zone = 4;

While(zone>0)

{

for(i=0; i<Num_MU;i++)

{

if((MU(i)_zone == zone) && (MU(i)_BE has traffic) &&($Remaining^{PS}_{f}$ >= 0))

{

Grant BW for this user's BEconnection up to $Remaining^{PS}_{f}$;

Decrement $Remaining^{PS}_{f}$ by the number of Slots granted

}

```
            }
            zone --;
        }
    }
```

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