Resource management strategies for quality of service in IEEE 802.16 networks

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**RESOURCE MANAGEMENT STRATEGIES
FOR
QUALITY OF SERVICE IN IEEE 802.16 NETWORKS**

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Summary

WiMAX is an industry standard for wireless broadband to provide last mile internet access as an alternative to wired broadband such as cable and DSL. The Physical, Medium Access Control and Security layers for this are specified by the IEEE 802.16 series of standards. The standards aim at providing Quality of Service for different multimedia traffic like voice, video and data that have different service requirements. Though the standards emphasize on handling different traffic classes according to their QoS requirements, the details of resource management is left unspecified. The most recent multi-hop relay standard (802.16j) aimed at increasing the WiMAX base station coverage also requires an efficient packet scheduling architecture at each hop to provide QoS to different traffic classes.

In our thesis, we have proposed fixed and adaptive queue-aware uplink scheduling principles which will efficiently share bandwidth among different traffic classes to provide QoS in a Point to Multi-Point topology. The uniqueness of our strategy is that it schedules real-time and non real-time traffic flows together such that real-time traffic can achieve the good delay performance while keeping non real-time traffic within their buffer limits. We have also proposed a scheduling architecture to support QoS over multiple hops by using relay stations as specified in IEEE 802.16j standard. The proposed scheduling strategy effectively combines centralized and distributed resource allocation principles to provide the required QoS for different traffic classes. The architecture and scheduling algorithm are compatible with the legacy IEEE 802.16 standard and hence does not need any functional change at the subscriber stations. We have also proposed a modified Earliest-Due Date scheduling scheme that

out performs the Strict Priority queuing and also provides fair resource allocation to lower layers of the Relay Tree.

Chapter 1

Introduction

1.1. WiMAX Networks

The IEEE 802.16 standard defines the air interface specification for wireless broadband access, popularly known as World-wide Interoperability for Microwave Access (WiMAX). The standard was first introduced in 2001 for supporting Point to Multi Point (PMP) communication and Mesh mode communications. The standard was revised as IEEE 802.16d in 2004 and later as 802.16e in 2005 to include mobility support. WiMAX aims at providing last mile wireless broadband access as an alternative to wired broadband like cable and DSL. WiMAX provides fixed, portable and mobile wireless broadband connectivity even without the need for direct line-of-sight with a base station. A typical WiMAX cell radius is three to ten kilometers, with a capacity of up to 75 Mbps per channel for fixed and portable broadband access. This is enough bandwidth to support hundreds of offices with T-1 speed connectivity and thousands of homes with DSL speed connectivity. Mobile WiMAX network can provide up to 15 Mbps of capacity within a typical cell radius deployment of up to three kilometers and the cell coverage can be increased by using relay stations as proposed in IEEE 802.16j draft. It is expected that WiMAX technology will be incorporated in notebook computers and PDAs by 2008, allowing for urban areas and cities to become “metro zones” for portable outdoor broadband wireless access. WiMAX can operate in both licensed and unlicensed bands. The unlicensed band can be utilized for personal communications and the licensed band can be used for

commercial installations. WiMAX has a robust OFDMA PHY layer with adaptive modulation and coding features and H-ARQ. The IEEE 802.16 series standard lays great emphasis on providing Quality of Service (QoS) at the MAC layer. This implies that wireless broadband with QoS could be available at vehicular speeds and can potentially replace the cellular system, which is not capable of providing broad band internet connection with QoS guarantees. The following table provides the comparison of different versions of the standard.

TABLE 1: IEEE 802.16 Standards

	IEEE 802.16	IEEE 802.16d	IEEE 802.16e
Completed	December 2001	May 2004	December 2005
Spectrum	10-66Ghz	2-11Ghz	2-6Ghz
Application	Backhaul	Wireless DSL & Backhaul	Mobile Internet
Channel Condition	Line of Sight only	Non-Line of Sight	Non-Line of Sight
Bit Rate	32-134Mbps at 28MHz Channelization	Up to 75Mbps at 20Mhz Channelization	Up to 15Mbps at 5Mhz Channelization
Modulation	QPSK, 16QAM & 64 QAM	OFDM 256, OFDMA 2048 QPSK, 64QAM	Same as 802.16d, Scalable OFDMA
Channel Bandwidths	20, 25 & 28MHz	Selectable Channel Bandwidths between 1.5 & 20 MHz	Same as 802.16d

1.2. IEEE 802.16 PMP Architecture

The IEEE 802.16d standard [1] specifies a Point to Multi Point (PMP) architecture for WMAN. The PMP wireless access system has at least one Base Station (BS) and two or more Subscriber Stations (SS) as shown in Fig 1. Each SS can be connected to multiple computing devices that generate multimedia traffic.

Figure 1: Wireless Broadband Access PMP Network IEEE 802.16

In the PMP architecture, all SS nodes directly communicate with the BS which controls uplink and downlink transmissions in the network. The individual SS supports multiple real time multimedia traffic like Voice over Internet Protocol (VoIP), Video on Demand (VoD), Motion Pictures Expert Group (MPEG) streaming and non real time traffic such as Hyper Text Terminal Protocol (HTTP) and File Transfer Protocol (FTP).

1.2.1 IEEE 802.16 Protocol Layers

The IEEE 802.16 standard defines a set of protocol layers for providing QoS to multimedia traffic as shown in Fig 2.

Figure 2: Protocol Architecture

The functions of each of these protocol layers shown in Fig. 2 and are briefly described below.

- (i) *Service specific convergence sub-layer:* The service specific convergence sub-layer (CS) provides transformation or mapping of external network data, received through the CS service access point (SAP). This includes classifying external network service data units (SDU) and associating them with the proper service flow identified by the connection identifier (CID). A service flow is an unidirectional flow of packets that is provided with a particular QoS. The CS provides compatibility to upper layer technologies like ATM and IP.
- (ii) *MAC common part sub-layer:* The function of the MAC common part sub-layer (CPS) is to provide wireless system access, bandwidth allocation, connection establishment, and connection maintenance. It receives data from various CSs classified to particular CIDs. QoS is applied to

transmission and scheduling of data over the PHY layer. IEEE 802.16 is mainly optimized for point to multipoint (PMP) configurations where several subscriber stations (SSs) are associated with a central base station (BS). As an optional feature it allows for a flexible Mesh deployment where a direct communication between SSs is possible.

- (iii) *Security sub-layer:* The security sub-layer provides subscribers with privacy across the Fixed Broadband Wireless Access (FBWA) network by encrypting connections between SS and BS. Whether the payload of a Protocol Data Unit (PDU) is encrypted or not is indicated in the MAC header. The exact type of encryption and its usage is negotiated during connection setup between the BS and SS.
- (iv) *OFDM PHY layer:* The IEEE 802.16 physical layer uses Orthogonal Frequency Division Multiplexing (OFDM) with a 256 point transform, designed for both line of sight (LOS) and non-line-of-sight (NLOS) operation in frequency bands below 11 GHz, both licensed and unlicensed bands. Time division duplex (TDD) and frequency division duplex (FDD) variants are defined. Channel bandwidths vary from 1.25 to 28 MHz. Additional air interfaces based on orthogonal frequency division multiple access (OFDMA) with a 2048-point transform and based on single-carrier (SC) modulation are specified.

The MAC Layer is of prime importance for QoS guarantees. The IEEE 802.16 MAC is optimized for point to multipoint (PMP) configurations as shown earlier in Fig.1, where several subscriber stations are associated with a central base station. All flows between a SS and BS are treated as connection-oriented with an assigned connection identifier (CID) and are governed by a set of QoS parameters. An SS would be

provided three management connections by the BS, namely (a) a *basic management connection* that is established during SS initial ranging and used to transport delay-intolerant MAC management messages, (b) a *primary management connection* that is established during initial SS ranging and used to transport delay-tolerant MAC management messages and (c) a *secondary management connection* that may be established during SS registration and is used to transport standards-based (e.g. SNMP, DHCP) messages.

1.2.2 IEEE 802.16 Frame Structure

The system supports a frame-based transmission shown in Fig. 3. TDMA/TDD is the most popularly used multiple access technology, though the standard [1] also specifies FDD techniques.

Figure 3: IEEE 802.16 MAC Frame in TDD Mode

In the TDD system, each frame is divided into up-link (UL) and down-link (DL) sub frames with the DL sub-frame always preceding the UL sub-frame as shown in Fig 3.

1.2.3 IEEE 802.16 Service Types

Each sub-frame has a number of time slots and the BS assigns the time slots in the uplink sub-frame dynamically to SSs upon requests from those SSs. At an SS, if any

new MAC Protocol Data Unit (MPDU) generated at the convergence sub-layer does not belong to current existing flows, it will be classified into one of the four service types and a new traffic flow will be created. This flow creation would be communicated to the BS via a dynamic service addition (DSA) message which carries the QoS parameters of the flow over the primary management connection. The BS would approve the new flow based on its admission control strategies, if any, and would set a connection identifier (CID) for that flow. Each flow is also identified by a 32-bit service flow identifier (SFID). Individual flow's QoS parameters can be changed using dynamic service change (DSC) or could be deleted by dynamic service deletion (DSD) message which are carried over the primary management connection. The BS periodically polls every SS to invite each to send its bandwidth request in the polling time slot allotted. The bandwidth request for the new traffic flow will be sent to the BS along with the QoS parameters of the flow. The BS will then assign a Connection Identifier (CID) for the flow and will send a bandwidth grant back to the SS. The BS allocates bandwidth to each SS based on the aggregate bandwidth required by all the traffic flows within the SS in a Grant per Subscriber Station (GPSS) system. In a Grant per Connection (GPC) system, the BS handles the bandwidth requirement of each connection individually without any support from the SS. An SS can send additional bandwidth requests in an incremental or aggregate fashion against specific CIDs when it is polled again. At the start of each downlink frame, the BS informs all SS when their respective data bursts have been scheduled on the downlink via the DL-MAP. It also specifies the starting time slot and the number of time slots for each SS to transmit on the uplink through the UL-MAP. The BS also provides contention time slots on the uplink for the transmission bandwidth requests by the Best Effort service from SSs.

In order to guarantee QoS to different types of multimedia traffic, the standard [1] defines four different types of services classes. This is necessary to deal with different types of multimedia traffic which need to be handled differently from each other by the MAC layer in order to satisfy their individual traffic requirements. Each of the traffic classes are associated with their respective QoS parameters like maximum delay, delay jitter, minimum sustained rate and maximum sustained traffic rate. The different traffic classes are -

(i) *Unsolicited Grant Service (UGS):*

UGS is meant for real-time traffic flows which require strict delay guidelines. Generally these kinds of service flows have fixed data length MPDUs. UGS is best suited for T1, E1 and VoIP flows without silence suppression having constant bit rate. UGS is provided with fixed size grants on a periodic basis to enable the flows to meet real-time delay deadlines. Since the UGS flows do not need to wait for polling and the subsequent granting of bandwidth by the BS, they provide the minimum latency.

(ii) *Real Time Polling Service (rtPS):*

The rtPS service is meant for real-time service flows (such as Motion Pictures Experts Group (MPEG) video) that generate variable size data packets and hence differ in their out-put bit rate. These traffic flows may also have soft delay deadlines. In order to serve these types of flows the BS will have to poll the SSs for their bandwidth request. Based on the bandwidth request received, the BS can then assign adequate uplink bandwidth for the SS.

(iii) *Non-Real Time Polling Service (nrtPS):*

The nrtPS service is meant for variable bit rate flows which do not have a delay dead line but have to obtain better than best effort service and guaranteed data transfer. Examples of this will be bandwidth intensive file transfers as in FTP applications. For nrtPS, the standard [1] recommend a polling procedure for bandwidth request and grant operations. However, maximum latency parameters are not defined for these services.

(iv) *Best Effort Service (BE):*

Best Effort traffic like HTTP does not have any QoS demands. For this service, the SSs are allowed to contend for bandwidth during the contention part of the uplink sub-frame. If the SS is successful then it can send its bandwidth request for the BE service flows.

1.3. IEEE 802.16j Multi-hop Relay Architecture

The cellular WiMAX architecture can be successfully deployed to provide mobile broadband access. However, poor signal quality and low SNR at the cell boundaries may limit the broadband coverage. Hence there is a need to extend the coverage of the traditional PMP network. The Mesh network with inter-SS communication which was part of the initial documentation had the ability to solve this problem. However since it was not made compatible with the PMP architecture, we need a new architecture to extend the coverage of the IEEE 802.16 BS. The IEEE 802.16j task group [2] is working on the construction of such a multi-hop tree based relay network that is compatible with the legacy standard [1]. It aims at achieving coverage extension without any functional change to the SS. The multi-hop scheme will also reduce the transmission power and increase battery life of the mobile stations (MS). A typical

relay based 802.16j system is shown in Fig 4. Here the BS operates in the same PMP mode while the relay stations serve the subscribers which are not able to be directly served by the BS through a TDMA framework. The relay stations need to have the intelligence to work on the MAC and PHY layers. The relay stations should also be able to provide the same level of QoS as provided by the BS to various multimedia traffic and also needs to be compatible with the PMP system. The relay stations also have to handle different traffic types defined for the PMP mode according to their QoS and provide delay guarantees to the real-time flows, especially UGS voice traffic which can be difficult on the multi-hop network. The relay system should also use frame structure that is compatible to the PMP TDMA frame structure shown in Fig. 3 so that they do not require changes in the protocols at the subscriber and mobile stations.

Figure 4: IEEE 802.16j Multi-Hop Relay System

1.4. Motivation

The IEEE 802.16 standard [1, 3] gives a detailed description of MAC, PHY and Security sub-layers for PMP operation. It also specifies four types of multimedia traffic classes and their associated QoS parameters. However it has left unspecified the resource management strategies like scheduling and admission control for the different traffic classes. In a wireless broadband system, the available bandwidth should be used efficiently in order to support a large user base with the QoS guarantees that are required. For achieving this, proper resource management strategies and scheduling algorithms for efficient bandwidth sharing must be proposed, designed and implemented. These should be such that the QoS support required by different types of multimedia traffic can be met while sharing the available bandwidth efficiently between large numbers of users. There is also a bandwidth request-grant delay in the TDMA based system which can adversely affect real-time traffic with stringent delay requirements. Hence there is a need for a scheduler at the SS to provide delay guarantee to the real-time flows. This motivates the need for developing effective scheduling mechanisms which can provide QoS guarantee (especially for the real-time traffic) within the QoS service framework specified in the standard. Furthermore, co-ordination is needed between the BS and the SS to allocate available resources between different multimedia flows and guarantee QoS. Resource allocation on the uplink is generally difficult since the BS has to deal with many flows from different SS. Hence there is a requirement for a scheduler at each SS that efficiently shares the available bandwidth among different traffic classes to ensure the required QoS.

The IEEE 802.16j Multi-hop relay specification is introduced to extend the coverage of the PMP mode. Resource allocation and scheduling in a multi-hop relay network will be complex as it has to be compatible with the legacy standard. The relay system also needs to consider the QoS aspects which are provided by the traditional PMP mode and offer QoS support to multimedia traffic. In a multi-hop network, inefficient bandwidth allocation at different hops may be a performance bottleneck for the whole system. Thus resource allocation in a multi-hop relay based WiMAX system is a challenging issue that need to be addressed. Further the bandwidth request-grant process introduces complexity and increase the packet delay. Hence an efficient scheduling algorithm is needed to provide nominal delay bound for real-time traffic over multiple hops while being fair to other traffic classes. There is also a need for MAC architecture for the Relay Station which has not been discussed by the task group [2].

1.5. Objective of Research

The above discussion provides an impetus to analyze the existing IEEE 802.16 PMP protocol and propose algorithms for effective scheduling on the up-link to provide QoS. The Multi-hop relay scheme also needs an efficient scheduling algorithm to differentiate QoS among different multimedia traffic classes. The following were the objectives of our research.

1. To propose an efficient scheduling algorithm for the PMP mode, which takes into consideration both the queue occupancy and latency of the real-time flows and the buffer limit of the non-real time flows in order to provide differentiated

QoS. The real-time flows requires delay guarantee while the non-real time flows have a buffer bound.

2. To improve the performance of the proposed fixed scheduling algorithm so that it adapts to changes in the relative queue sizes of the real-time and non-real time flows.

3. To analyze the scheduling algorithms using a discrete Markov Chain model.

4. To propose a dynamic scheduling algorithm which varies the bandwidth allocation to real-time and non-real time flows based on the latency requirement of the real-time flows and the buffer limit of the non-real time flows.

5. To analyze the 802.16j protocol and propose a scheduling architecture and frame structure to provide QoS over multi-hop network without any functional change to the subscriber stations.

6. To propose a novel scheduling algorithm that combines centralized and decentralized scheduling to overcome bandwidth request-grant delay and minimize delay for real-time traffic. The scheme needs to handle UGS, rtPS, and nrtPS traffic.

7. Improve on the scheduling algorithm and provide a modified EDD based scheduling for real-time flows to be fairer to the nodes which are farther away from the base station and avoid bandwidth hogging by higher layers.

1.6. Organization of the Thesis

This thesis provides three variants of QoS scheduling strategy proposed for the IEEE 802.16 PMP network with mathematical analysis. The thesis also proposes a QoS scheduling architecture for IEEE 802.16j relay based multi-hop

communication and provides a frame structure for the same. The organization of the thesis is as follows:

Chapter 2 provides a literature review of the current resource allocation schemes available for the WiMAX PMP system, it also analyses the applicability of traditional scheduling schemes available. The chapter also reviews the IEEE 802.16j standard and the resource allocation schemes proposed in the existing literature.

Chapter 3 provides the detailed description of the proposed resource allocation strategies for up-link scheduling in the IEEE 802.16 PMP mode. It describes the fixed scheduling scheme and the queue based adaptive scheduling schemes and provides a Markov chain based approximate analysis. It also discusses the delay feedback based dynamic scheduling scheme. Finally it provides detailed analytical and simulation results to validate the proposed algorithms.

Chapter 4 provides the details of the proposed hybrid scheduling architecture for tree based multi-hop networks. It also discusses the QoS scheduling for multimedia flows in multi-hop network. It also provides an improvised EDD scheme to improve the performance of the earlier algorithm and includes simulation results to validate the performance of both algorithms.

Chapter 5 concludes the thesis and provides recommendation for future work.

Chapter 2

Literature Review

Worldwide Interoperability for Microwave Access (WiMAX) based on the IEEE 802.16 standards have been proposed since early-2001 for cheap, ubiquitous wireless access with high bandwidth. This technology has tremendous potential for providing cheap broadband wireless access, and is being currently refined through extensive research contributions. Performance evaluations of the IEEE 802.16 system have reported 40-50% improvement in spectral efficiency compared to current 3G High Speed Downlink Packet Access system (HSDPA)[4] and its MAC and PHY layers have also been proven to be quite robust [5]. A substantial amount of literature is also available on resource allocation in wireless media. In this chapter, we discuss the current literature on WiMAX scheduling which are relevant to our work and have also motivated our own research in this area.

2.1. Traditional Scheduling Schemes in Wireless Networks

Even though the IEEE 802.16 specified diverse QoS schemes need a new scheduling approach, it is also necessary to study the effect of traditional scheduling algorithms used in wired networks in order to set the appropriate bench marks that new

algorithms must meet. There have been several studies on scheduling algorithms in wireless networks in general [6] - [10]. Most of them concentrate on adapting wire-line scheduling techniques to wireless networks. However, it has been observed that direct application of conventional wire-line scheduling techniques to wireless networks tends to give poor utilization of the limited bandwidth available [6]. A major reason for this is that wireless physical links are prone to losses due to fading. They also suffer from limited bandwidth availability. Hence Packet Fair Queuing (PFQ) techniques were adapted to wireless networks with some modifications. The Weighted Fair Queuing (WFQ) was adapted to wireless network by [7] where in order to make efficient use of bandwidth, the scheduling postpones the packet transmissions of users experiencing poor channel quality and compensates them later with additional bandwidth when their channel quality improves. The authors in [7] determine the bandwidth that is required to be compensated in this fashion by simulating an idealized WFQ policy in real-time and comparing it with the actual number that has been awarded. The users who had been enjoying more than their fair share of bandwidth were forced to forgo bandwidth in favor of users who had been previously deprived in order to ensure long-term fairness in the system. Moreover, the scheme imposes a maximum bound on the bandwidth made available to users whose wireless channel is always in a good state, this scheme suffers from a few drawbacks as indicated in [6]. First, there is an overhead associated in making scheduling decisions based on simulation results obtained from an idealized WFQ policy. Secondly, wireless channels have a constrained bandwidth and hence WFQ may not work perfectly since it can provide delay guarantee only if bandwidth is guaranteed. Thirdly, a WFQ policy does not handle link-sharing objectives. Therefore, the scheduling policy in [7] may not be able to comply with the link-sharing

objectives. Hence, in WiMAX, where the up-link is shared between multiple flows arising out of the same SS, the flows cannot be adequately scheduled by using PFQ techniques. Furthermore, since WiMAX has a robust PHY, what it really requires is a scheduling principle which would provide QoS for multiple flows which share the same robust link.

The work in [6] tries to fix the shortcomings in [7] by using a Server Based Fairness Approach (SBFA). The aim of this scheme is that all users whose channel is in good state should receive their promised service. The other users who have not received adequate service due to poor quality of the channel are provided bandwidth using a pre-allocated portion. SBFA creates special sessions called Long-Term Fairness Servers (LTFS) to keep track of the amount of bandwidth a user needs to be compensated with. The bandwidth is shared among several wireless sessions. To ensure overall fairness over a long time, bandwidth allocated to the LTFS is used to compensate the bandwidth deficiency of other sessions.

The authors in [8] proposed a link level channel state dependent packet (CSDP) scheduling methods to overcome the problems associated with multiple users sharing the wireless link. They look at the problem from the perspective of a base station serving several mobile stations on the downlink. For each mobile host, the base station maintains a separate queue of packets. Within each queue, packets are served in First-In-First-Out (FIFO)/ Strict Priority (SP) order. The service difference between the queues depends on specific policies set up for each queue. These policies are based on the current channel states of each mobile node. The authors argue that using traditional protocols designed for wire-line network will cause Head of Line (HOL)

blocking due to repeated transmissions from a single user with bad channel state. This motivated their CSDP approach and they used simulations to show the interaction of FIFO, Round Robin (RR), Earliest Timestamp First (ETF), and Largest Queue First (LQF) schedulers with the dynamics of multiple TCP sessions.

The authors in [9] proposed a scheme based on combining a modified version of Class Based Queuing (CBQ) with the Channel-State Dependent Packet Scheduling introduced in [8]. They incorporate link sharing objectives so that the scheme works well with wireless links based on radio channels. These channels are shared on-demand among multiple users, and are subjected to dynamic bandwidth variations due to spatial and temporal fading. The authors model the channel as a two state Markov chain (bad and good) and study the system through simulations. They conclude that the basic CBQ and CSDSP schemes need to be modified to work well with wireless links instead of a direct combination of the two. They observe that throughput and fairness can conflict with each other and that throughput may have to be sacrificed for fairness or vice-versa. They also advocate studying the impact on buffer allocation at the transmitter since the burstiness of one connection may be modulated by fading while interference affects another connection.

Fairness issues in wireless networks have also been studied. Authors in [10] *introduced a wireless credit-based fair queuing (WCFQ) scheduler which will work* for wireless packet networks that have provable statistical short-term and long-term fairness guarantees. The scheduler makes use of the fact that users contending for channel access will have different costs of transmission depending on their current channel condition, e.g. in systems using variable coding, a user experiencing good

channel quality can use their low-cost channel and transmit at a higher data rate. Similarly, a user in a code-division multiple access system with a good channel quality can use a lower transmission power. The authors in [10] use this to provide a mechanism to exploit inherent variations in channel conditions and select low-cost users in order to increase the system's overall performance (e.g. total throughput). WCFQ, introduced in [10], uses a credit abstraction and a general "cost function" to opportunistically select the best user while balancing fairness considerations. WCFQ is flexible and provides system operators a range of performance behaviours between perfect fairness for temporal access which is independent of channel conditions and purely opportunistic scheduling of the best user without consideration of fairness. The authors in [10] quantify the system's fairness characteristics within this range, by developing an analytical model that provides a statistical fairness bound in terms of the cost function and the statistical properties of the channel.

An opportunistic packet scheduling algorithm based on buffer management (OSBM) over the downlink of a packet cellular network was proposed in [11]. The authors also present an analytical approach for performance analysis of opportunistic scheduling in wireless networks based on the concept of effective downlink capacity (EDC). This approach uses a service curve tool for deterministic quality-of-service analysis to the wireless environment. The EDC concept has been used to bridge the deterministic method and the stochastic nature of the wireless link. The OSBM provides differentiated services to support real-time and non-real-time applications and uses queue management techniques to accommodate both changing channel conditions and delay constraints of real-time data. A proactive packet discard mechanism has also been used by the authors to use the wireless channel more effectively.

Another scheme that addresses multi-user scheduling over the downlink channel in wireless networks is given in [12]. The authors consider a time-slotted system with a single base station serving multiple subscriber stations. The strategy of the authors is to optimize the time-slot utilization by users with QoS specifications and maximize the remaining fraction of time-slots which will be made available to best-effort users. The authors have provided formulas that relate the achievable throughput rate guarantee to the number of QoS users that can be supported by the system. They have also provided ways to calculate the time-slots allocated to the best-effort users in a scenario where the system encounters symmetric Rayleigh fading.

Queue length based scheduling on the down-link has also been studied in the context of cellular networks. The authors in [13] show that a combination of base station controlled queue-length based scheduling and congestion control implemented either at the base station or at the end users may lead to fair resource allocation and queue-length stability. The base station scheduler makes scheduling decisions based on the different downlink queue lengths and the current channel condition of different users. The authors in [13] claim that throughput-optimal scheduling is more suitable for inelastic traffic where the sources do not adapt their transmission rate based on congestion in the network while fair scheduling is more suited for elastic traffic sources which can adjust their traffic rates in response to feedback from the network regarding the network conditions.

Feedback based control algorithms have also been developed for providing soft QoS in WLAN networks. The authors in [14] present an optimal controller for the scheduling of real time traffic in 802.11e Wireless LANs. This paper presents a

centralized scheduling scenario where, during the contention free period, the access point shares the channel utilization among a set of user nodes with real-time traffic. To overcome the problem of under-provisioning or over-provisioning of resources for up-link transmission, the authors in [14] advocate a dynamic scheme. In this scheme, a feedback controller present inside the access point adjusts the allocation of resources to the different flows based on their time-varying needs. The feedback controller makes decisions based on the “sensor” measurements of the state of the system which is associated with the buffer levels. The Transmission Opportunity (TXOP) which represents the maximum time a flow can use the channel (including management and control frames) is varied by the access point at the beginning of each polling cycle. This strategy allows the access point to control the number of queued packets awaiting transmission. This effectively reduces the number of queued packets and will consequently also reduce the need for buffer memory and the end-to-end latency.

To summarize, complex wire-line scheduling techniques have been modified for use in wireless networks. In most cases scheduling algorithms were made to adapt to changes in channel quality and concentrate on down-link scheduling alone. Further traditional algorithms like PFQ can handle only a single class of traffic. However WiMAX has a robust PHY and has multiple flows arising out of a single SS sharing the up-link. Hence what WiMAX really requires is a simple-to-implement MAC scheduling principle which would provide QoS for multiple flows which share the same robust link.

2.2. WiMAX Resource Allocation

2.2.1. Resource allocation and QoS Architecture

Many resource management algorithms have been proposed to handle multimedia data. However, the WiMAX network is unique in the sense that QoS has to be provided over a wireless medium. Moreover, since the WiMAX PMP network is centrally controlled by a BS, the BS needs to have a robust algorithm to handle the UGS, rtPS, nrtPS and BE traffic according to their respective QoS demands. Alavi [15] proposed a general architecture for WiMAX system to ensure QoS for the different classes of traffic. The components of the architecture at the SS and BS are shown in Fig. 5. This architecture provides an outline of data and signal transmission paths from the time they enter the sender's MAC layer to the time they leave on the receiver side. The IEEE 802.16 standard [1] specifies a connection-oriented MAC layer, where each user (SS) first sends a Connection Establishment Request to its BS. The request is then analyzed in Call Admission Control. If the connection request is accepted, then the BS registers the connection by providing two connection identifiers (one each for up-link and down-link) and the corresponding QoS attributes are stored in its Service Flow Data Base. In order to provide QoS support, the Classifiers at the MAC entrance point classifies incoming packets according to their respective identifiers. Admission control schemes are designed for controlling the generation of traffic flows. Since only limited bandwidth is available which is shared by multiple traffic flows, there is a need for restricting new traffic flows depending on their priorities. The authors in [15] define the transmission process of data from SS to BS in the following order. BS polls each SS at specific intervals according to the Polling mechanism defined in the standard [1]. This provides each of the connections with the

opportunity to send their respective bandwidth requests. This function is done by the Polling Manager at the SS. The SS will use this polling opportunity to send its bandwidth request to the BS. The bandwidth request is made after considering QoS attributes and the queue state of different connections within the SS. After obtaining the bandwidth request from all the active connections, the Uplink Scheduler in the BS divides the entire bandwidth on the uplink among the active SSs based on the bandwidth requests it has received and the QoS attributes of each connection. The allocation of up-link bandwidth is communicated to each SS in the downlink and they are allocated specific time slots to transmit on the uplink. The authors define a Grant Allocator in each SS which divides the total bandwidth it has obtained between its different connections according to their respective QoS characteristics. During downlink, the BS downstream generator delivers specific amount of data from each of the downlink queues based on bandwidth allocation by the downlink scheduler. This module is also responsible to send messages generated by the Uplink Scheduler. The authors specify components such as Traffic Shaper and Policer in both the SS and the BS which examines whether the traffic flow on the uplink and downlink conform to the QoS attributes. This work [15] provides a good understanding of the working of both the BS and the SS for the connection setup and bandwidth request grant mechanisms. However, there are no schedulers at the SS and schemes for providing QoS guarantees are also not discussed.

Figure 5: QoS Architecture for IEEE 802.16 Network
(Adapted with modifications from [15])

2.2.2. Admission Control Scheme

Wang [16] proposed an Admission Control scheme for the different services defined in IEEE 802.16 and provided an analytical model to evaluate the system performance of the proposed scheme. The paper describes two control strategies (a) Admission Control where BS receives a request from one of the SS and makes a decision whether to admit the connection or not and how much bandwidth should be set aside for the connection during its entire transmission duration and (b) Granting Control where after the connections (traffic flows) are established, the BS polls all the SS and each SS uses this opportunity to send bandwidth requests for each of its flows to the BS. The BS then makes a decision on whether to grant the bandwidth requested or not. The BS may also wish to ignore bandwidth requests from some flows while granting

bandwidth requests from other flows. Both admission control and granting control are based on the current bandwidth utilization of the system and the QoS requirements. However, neither of these processes has been directly addressed in the IEEE 802.16 standard [1]. In the admission control scheme proposed in [16], the BS dynamically decides whether to accept or reject a SS request for connection. The decision made by the BS is based on the long term bandwidth requirements of all the connections handled by it and also the current state of the network. The proposed admission control performs a tradeoff between accepting a request for a new connection that may result in QoS degradation of the current connections and rejecting a request for a new connection in order to maintain the QoS support of the current connections at a certain level. In [16], the authors observe that blocking new UGS flows (e.g. telephone call traffic) causes more serious problems than blocking new non-UGS flow (e.g. video streaming). Hence the BS accepts every request for UGS connection and the bandwidth required for the UGS connection's transmission is guaranteed. The algorithm proposed for this is as follows. If the total bandwidth available in the system is B then UGS connections are allowed into the system up to a maximum bandwidth of B . If at any instant UGS occupies a bandwidth of U ($U < B$), then the remaining $B-U$ bandwidth is made available for rtPS and nrtPS traffic. Next the rtPS connections can occupy a maximum bandwidth of $B-U$. Non-UGS traffic is only accepted if the total bandwidth utilized is less than the total available bandwidth B . Moreover, when an UGS or rtPS connection enters the network and the network is already operating at its maximum bandwidth, the bandwidth for each of the nrtPS connections are reduced in steps by a degrading factor δ . The paper goes on to provide an analytical model for this admission control and degradation process assuming the arrival process of new connection requests for UGS, rtPS and nrtPS to

be Poisson distributed. The paper further models each SS with a four dimensional continuous Markov Chain with states as the number of packets in UGS, rtPS and nrtPS queues and the degrading factor. The author derives performance measures like new connection blocking probability, bandwidth utilization and transmission capacity for their proposed approach.

2.2.3. Traditional Scheduling Algorithms Applied to WiMAX

Even though the IEEE 802.16 QoS schemes are novel, it is necessary to study the effect of traditional scheduling algorithms used in wired networks in order to set an appropriate bench mark that new algorithms must meet. It will also be interesting to study the effect of combining traditional algorithms for scheduling the different traffic classes defined in [1]. Immediately after the initial draft of IEEE 802.16 standard [1] came into being, Wongthavarawat [17] proposed a scheduling algorithm assigning a fixed bandwidth for UGS, using Earliest Deadline First (EDF) technique for rtPS, Weighted Fair Queuing for nrtPS and equal distribution for BE. Vinay [18] improved on these algorithms and has done a performance evaluation of end-to-end delay by using a hybrid scheduling algorithm. Some of the traditional scheduling algorithms evaluated are WFQ, virtual clock (VC) and EDD. In [18] it is claimed that for real-time communications over a network with stringent time limits, the EDD scheduling algorithm will be generally preferable. A WiMAX node can select one of the scheduling algorithms for all the services in a node or different scheduling algorithms for different services. Extensive research is currently underway to propose an optimum scheduling algorithm to support real-time and non-real time communications in WiMAX. The paper [18] goes on to evaluate EDD delay where, after receiving the packets in the queue, the scheduler will stamp the packets with an

expected deadline value and the ordering of the packets in the queue is done as per the increasing order of deadlines. The deadlines are considered according to the priorities of the packets. The Weighted Fair Queuing strategy is also analyzed, The WFQ scheduling algorithm is similar to generic fair queuing scheduling algorithms that are available in literature. The strategy of fair queuing algorithms is to provide equal network resource allocation to all the services. The basic idea of fair queuing is to transmit data from each service in turn. Instead of giving equal share of resources to all users, WFQ provides a specific amount of resource to each service thus assuring requested resource reservation for each user. However, WFQ does not impose a stringent time limit to each packet. Hence, there is a possibility of service fluctuation in case of bursty traffic. The paper also assumes that the packets arriving into each service queue are buffered. Each packet is treated with its respective priority according to the EDD or WFQ scheduling algorithm. However there is no explanation on how the priority is actually decided in WFQ. The paper implements a hybrid protocol which is a combination of EDD for real time data traffic and WFQ for non real time traffic. The simulations are carried out using a 1 Mbps channel and 20 Subscriber stations. It compares the performance of traditional schemes like EDD, WFQ and the proposed hybrid scheme. It also evaluates the GPSS and GPC approaches. An important inference made is that GPSS has a better performance than GPC and the former can accommodate more nodes in the system than GPC for the same end-end delay. This is due to the lower processing time in GPSS where the BS schedules only the subscriber stations and not the individual traffic flows inside a SS (This assumes that each SS has multiple traffic flows). Note that this is another motivation to have a scheduler at the SS which re-distributes the granted bandwidth

locally instead of having the BS take care of scheduling and QoS treatments of each flow in each SS.

To summarize, though traditional scheduling schemes can be applied to WiMAX, it will not be possible to schedule different classes of traffic together. Each traffic flow would need a specific scheme and would not lend itself to provide over-all QoS to all the flows. For example if a particular class of flow achieves its QoS goal with resources to spare, it would not be possible to use this excess resource to another class of flow. Further the literature survey motivates the use of GPSS scheme which requires a scheduler at the subscriber station.

2.2.4. Scheduling Stand Alone VoIP flow

Some research works have also looked into stand alone voice traffic flows, e.g. scenarios where mobile stations transmit just voice traffic over the air interface. Though simple UGS scheduling schemes can take care of independent constant bit rate voice traffic, it would result in wastage of bandwidth [19]. As indicated earlier, in the UGS scheduling scheme the BS allots a fixed amount of bandwidth for each flow in a periodic fashion such that the bandwidth requirements of these constant bit rate flows are satisfied. However, advanced voice modulators present in VoIP devices generate data packets only when there is voice activity and do not transmit during silent periods. The operation of general UGS scheduling in WiMAX is illustrated in Fig. 6. The authors for their analysis use a enhanced variable bit rate traffic model with packet sizes of 171bits for full rate, 80bits for $\frac{1}{2}$ rate, 40 bits for $\frac{1}{4}$ rate and 16 bits for $\frac{1}{8}$ rate with voice activity factor of 0.403.

Figure 6: Operation of UGS scheduling
(Adapted from [19])

Fig. 6 shows UGS scheduling, the white bar shows the resources assigned by the BS and the amount of uplink resources utilized by the user, is shown by the black bar. The blank regions represent the waste of uplink resources. Since in the UGS algorithm the BS always allocates the same amount of uplink resources to each user regardless of the actual voice status, a substantial amount of uplink resources may be wasted. Howon [19] also examines the suitability of rtPS algorithm to handle stand-alone VoIP traffic. As pointed out earlier, the design goal of the rtPS scheduling algorithm is to support real-time service flows that periodically generate variable size data packets. The uplink resource allocation by the BS is based on the bandwidth requested by the MS/SS in its bandwidth request. Hence for VoIP, it is possible for the SS/MS to determine the bandwidth needed based on the status (active/silent) of its voice codec and send bandwidth requests accordingly. Since this algorithm always uses a bandwidth request process for specific size grants, it transfers data more

efficiently than the UGS algorithm. However, this bandwidth request process will result in extra MAC overheads and additional access delays. The rtPS algorithm will therefore have larger MAC overheads and access delays than UGS.

Figure 7: Operation of rtPS algorithm
(Adapted from [19])

In Fig. 7, for the rtPS algorithm, the white (hidden) and black bars show the amount of uplink resources assigned by the BS and the amount of uplink resources utilized by the VoIP user respectively. Since the user requests the exact amount of uplink resources needed for transmitting his/her voice packets, the black bar and the white bar are nearly the same in case of rtPS algorithm this is the reason for the white bars are being hidden behind the black bars. The grey bar represents the bandwidth request header. The VoIP traffic is not tolerant to the delay caused by the polling process. The delays involved in the process of polling, bandwidth request and bandwidth grant make it unsuitable for VoIP applications. Howon [20] devised an extended rtPS (ertPS) algorithm for VoIP services with variable data rates and silence suppression in IEEE 802.16 systems to overcome the defects of both UGS and rtPS in serving VoIP

applications. The proposed ertPS algorithm allocates uplink resources according to the status of the voice users.

In the ertPS algorithm [20], the BS is informed about the voice codec status of the user using grant management sub header along with the MAC header. This is achieved by using extended PBR (piggyback request) bits of grant management sub-header to send bandwidth requests for voice packets. The grant management sub-header format is defined by the IEEE 802.16 standard [1]. The Grant Management sub-header is two bytes in length and is used by the SS to convey bandwidth management needs to the BS. The Grant management sub-header has a 16 bit piggyback request bit which can be used to request variable bandwidth grant. In the ertPS algorithm, the user sets the MSB of PBR bits to 1 to differentiate these extended PBR bits from the general PBR bits. The BS periodically assigns uplink resources based on previously requested size until the user codec requests another size of the bandwidth. As the size of the voice data packet changes at the voice codec, the codec informs the BS of its voice status using the bandwidth request header. The bandwidth request header defined in [1] consists of an 11-bit MSB and 8-bit LSB bandwidth Request fields along with CID to identify the flow.

To summarize, the codec requests the bandwidth for sending the voice packets using BR (bandwidth request) bits of bandwidth request header only if the size of voice packet changes. Then, the BS also periodically assigns the same uplink resources according to the requested size, until the codec requests another size of the bandwidth. The operation of the ertPS algorithm is shown in Fig 8.

Figure 8: Operation of the ertPS Algorithm
(Adapted from [19])

To summarize, stand-alone VoIP traffic scheduling can only be applicable for traffic whose ON and OFF time can be pre-determined. It cannot be applied to bursty internet traffic that requires specific QoS. It also cannot be used to schedule multiple flows arising out of a single SS and will cause network overhead.

2.2.5. Queue Aware Strategies for Scheduling

The primary focus of current research is to develop an efficient algorithm which will allocate resources to different traffic flows based on their needs to ensure QoS. Most SS would generate multiple types of traffic flows. Proper allocation of scarce bandwidth resources between multiple traffic flows would be important for practical applications of WiMAX systems.

Niyato [21] has proposed a queue aware strategy for resource management in the polling schemes (rtPS and nrtPS). The work in [21] presents an adaptive queue-aware uplink bandwidth allocation and rate control mechanisms in a subscriber station for polling service. The proposed resource allocation scheme dynamically adjusts the amount of bandwidth allocated for polling service according to the following factors – (a) variations in traffic load, (b) channel quality and (c) queue length. This is done so that the MAC layer QoS performances such as protocol data unit (PDU) delay and PDU dropping probability are maintained at the desired level. In addition, rate control is also suggested to limit the transmission rate of the connections under these polling service classes so that the QoS performances can be controlled.

The approach of [21] specifies b_{max} ($b_{max} \in \mathbb{N}$) as the maximum number of MAC PDUs that an SS can transmit per uplink transmission sub-frame. It considers two modes of bandwidth allocation for PS, namely, complete partitioning (CP) and complete sharing (CS). With complete partitioning, a fixed amount of bandwidth b_{ugs} (from the total bandwidth allocated to an SS) is statically allocated for UGS while the remaining bandwidth (i.e. $b_{max} - b_{ugs}$) is allocated for the PS and BE services. In case of complete sharing, when the bandwidth requirement for UGS traffic is less than b_{ugs} , the remaining available bandwidth is provided to PS. After the required amount of bandwidth has been allocated for UGS and PS traffic, the remaining bandwidth is allocated to BE traffic. A conceptual model of this system is shown in Fig. 9.

Figure 9: System Model for Queue Aware Resource Management Strategy
(Adapted from [20])

Niyato [21] also proposes an uplink bandwidth allocation scheme for PS which takes the current number of PDUs in the PS queue into account. The allocation is done on a frame by frame basis in which the amount of bandwidth is individually determined for each transmission frame. In this scheme, the set of thresholds for bandwidth allocation is defined by the authors as follows:

$$\psi = \{\psi_1, \psi_2, \dots, \psi_b, \dots, \psi_{b_{\max} - b_{ugs}}\}$$

where $\psi_b \in \{1, \dots, X\}$, $\psi_b < \psi_{b+1}$ and $b \in \{1, \dots, b_{\max} - b_{ugs}\}$

The amount of bandwidth required in each uplink sub-frame is indicated by the set of thresholds. The author calculates the amount of bandwidth allocated to the polling service as a function of the number of PDU's in the PS queue for complete sharing and complete partitioning as given below.

$$B_{CP}(x) = \begin{cases} 0, & x = 0 \\ b, & \psi_b \leq x < \psi_{b+1} \\ b_{\max} - b_{ugs}, & \psi_{b_{\max} - b_{ugs}} \leq x \end{cases}$$

$$B_{CS}(x) = \begin{cases} 0, & x = 0 \\ b, & \psi_b \leq x < \psi_{b+1} \\ b_{\max}, & \psi_{b_{\max} - b_{\text{sgs}}} \leq x \end{cases}$$

The authors advocate a rate control to be implemented either at the data source or in the polling service queue. In the former case, the SS informs the traffic sources of the queue status. In the latter case, rate control is implemented in a manner similar to the random early detection (RED) mechanism used in an Internet router where some PDUs received at the PS queue are randomly dropped. The rate control policy is expressed as a function of the number of PDUs in the PS queue (x) as follows:

$$\lambda(x, \lambda_0, \lambda_{\min}) = \begin{cases} \lambda_0, & x < \tau_{\min} \\ F(\lambda_0, x), & \tau_{\min} \leq x < \tau_{\max} \\ \lambda_{\min}, & \tau_{\max} \leq x \end{cases}$$

where $\tau_{\min}, \tau_{\max} \in \mathbb{N}$ denote the rate control thresholds for the number of PDUs in the queue $\tau_{\min}$ denote the minimum guaranteed arrival rate of λ_0 and $\lambda(\cdot)$ is the controlled arrival rate and $F(\lambda_0, x)$ is a non increasing function of x with a constraint $\lambda_{\min} \leq F(\lambda_0, x) \leq \lambda_0$. The suggested rate control is done either on an aggregate or on a per flow basis. While per-flow rate control can differentiate the QoS between different connections, aggregate rate control is simpler to implement and is applicable when all connections have similar QoS requirements. The authors in [21] also look into error control in the MAC layer based on Automatic Repeat Request (ARQ). The performance of the queue aware strategies are analytically studied using a Discrete Time Markov Chain (DTMC). The queuing model for PS is formulated by the authors for both cases of complete partitioning and complete sharing schemes. The queuing model is solved to obtain the steady state and the transient state

probabilities of the system. These are then used to calculate performance parameters like the average number of PDU's in PS queue, average delay for PDUs in the PS and BE queues and the average bandwidth allocated.

In [21] the authors also suggest a way of devising a queue aware strategy which is subsequently used for studying the performance of the queue aware bandwidth allocation and queue aware rate control systems. However the queue aware uplink bandwidth allocation scheme of [21] does not differentiate between real time and non real time data traffic. The major factor that differentiates these two types of real-time traffic is their tolerance to delay. Hence, it is necessary to separate real-time and non real-time traffic in different queues and include the maximum latency specification on the real-time traffic in the scheduling process. It is also important to quantify QoS parameters like maximum latency, maximum sustained rate and minimum reserved rate for different traffic and schedule them based on specifications rather than qualitatively differentiating traffic based on their behavior. The work [21] motivates the use of GPSS scheme and use of a scheduler at the subscriber station which we have done in our work.

2.2.6. Adaptive Bandwidth Allocation

Mukul [22] proposed an adaptive bandwidth scheduling algorithm at the SS for real time polling service (rtPS), where the SS predicts the arrival of rtPS packets prior to arrival and requests the BS for bandwidth in advance. In this system, a SS will request a time slot not only for data currently present in the queue but also for data which will arrive in the queue in the time interval between when the request is made and the corresponding time at which rtPS queue will be served . Since data arrival is random,

this requires stochastic prediction for the data that may arrive in the queue in the time interval between the bandwidth request and grant. The method adopted in [22] is called a "Differential Time Grant" method where the author estimates the average rate of incoming data and average duration of the above time interval. This is used in estimating the amount of data that would arrive in that time interval. The method uses an arrival service curve and assumes the average data rate as 1Mbps. Here the departure rate is either the maximum transmission rate at the output link for rtPS (C) or zero.

In a WiMAX system however the SS does not transmit at a constant rate and the duration of a transmission burst would also vary. It is also inefficient to predict the number of packets that may arrive in a certain period and then request for extra bandwidth. This is because real time traffic may be very bursty and may be hard to predict. Moreover, such predictive allocation would put extra burden on the BS if all the nodes make such predictions and request bandwidth. In addition, each SS may contain multiple flows with different QoS factors and it would not be prudent to request for more bandwidth by mere prediction.

An alternative approach would be to distribute the allotted bandwidth efficiently among different traffic based on their QoS in an adaptive fashion and predict the buffer occupancy of different flows to schedule them appropriately.

2.2.7. Cross Layer Approaches

Some researchers have considered cross layer scheduling using the MAC scheduler and the PHY resource allocator. Sang [23] proposes a cross layer design of packet scheduling and resource allocation in OFDMA wireless networks which concentrates on downlink scheduling in the BS. In OFDMA systems, each carrier is subdivided into a number of sub-carriers which can be controlled or allocated separately making the PHY very robust. In the proposed system, the sub-carriers are grouped into allocation units (AU) for allocation so that overheads are minimized. The BS estimates the sub-channel condition of each user and allocates resources to the users on a frame-by-frame basis. The authors consider a system where some SS have real time traffic and other systems have non-real time traffic, i.e. real time and non-real time traffic do not co-exist in the same SS. The BS can estimate the channel gain of each user on a sub-channel and an AU can be independently allocated to a particular user. The aim is to maximize the overall utilization while satisfying the rate requirements of individual users. To do this, authors in [23] formulate a linear programming method and subsequently obtain a sub-optimal solution which reduces the computational time. The scheme also proposes a packet scheduler at the BS MAC layer which provides equal priority to both real-time and non-real time traffic except for the condition when the ratio of the waiting time of a real time packet in the MAC queue and the delay constraint of the packet exceeds a certain threshold. The paper [23] proposes a 3-stage resource allocation algorithm where the first stage schedules the urgent packets. The second stage schedules the non-real time packets and the real-time packets which are not urgent where higher priority is given to users whose channel quality is better regardless of the traffic type. The third stage allocates the unallocated AU's if any. After the packet scheduler decides the rate requirements of

each user, the actual resources are allocated in the PHY layer. This is carried out by using a sub-optimal heuristic algorithm which first allocates the sub-channels (or sub-carriers) to the user with the best channel conditions. This is first done without considering the rate constraints in order to maximize the utilization (or minimize the transmission power). After that, the algorithm relocates (or swaps) sub-channels so as to satisfy the rate requirements of each user.

The cross layer scheme though can be followed in the downlink at the BS, it cannot be adopted for uplink scheduling since individual nodes cannot decide the channel condition and place request for those particular channels. This also advocates the use of GPSS scheme which was found to be complex and inefficient.

There is very little prior work that deals with cross layer scheduling at the higher layers, i.e. the application and transport layers. Chen [24] aims at providing end to end QoS guarantee using Intserv and Diffserv in connection oriented WiMAX PMP and mesh networks. The work maps the RSVP in the IP layer and DSA/DSC/DSD in MAC layer. The authors in [24] propose that message exchange for DSA and DSC can be deployed to carry QoS parameters of IntServ services for end-to-end resource (bandwidth/buffer) reservation. For DiffServ services, on the other hand, a number of per-hop behaviors (PHBs) for different classes of aggregated traffic can be mapped into different connections directly.

In the proposed scheme shown in Fig.10, when a new application flow arrives in IP layer, it is first parsed according to the definition in PATH message (for InteServ) or Differentiated Services Code Point (DSCP for DiffServ). It is then classified and

mapped into one of four types of services (UGS, rtPS, nrtPS or BE). The proposed dynamic service model in SS sends a request message to the BS, where the admission control determines whether this request is approved or not. If it is not approved, the service module informs the upper layer to deny this service; else the admission control notifies the scheduling module to make a provision based on the parameter values in the request message. At the same time, the accepted service is transferred into a traffic grooming module. According to the grooming result, the SS will send Bandwidth Request message to BS. The centralized scheduling module in BS will retrieve the requests and generate UL-MAP message carrying the bandwidth allocation results. Finally, the SS will package SDUs from IP layer into PDUs and upload them in its allocated uplink slots to BS.

Fig. 11 shows the PATH, RESERV messages for Intserv. Here the sender sends a PATH message including traffic specification (TSpec) information. Parameters such as max/min bounds for bandwidth, delay and jitter is mapped into parameters of the DSA message such as Maximum Sustained Traffic Rate, Minimum Reserved Traffic Rate, Tolerated Jitter and Maximum Latency. According to the response received for the DSA message, the provisioned bandwidth can also be mapped into Reserved Specification (RSpec) in the RESV message. These parameters can be used at the MAC scheduler to allocate resources according to the QoS requirements of flows along their path.

Figure 10: Integrated QoS Architecture
(Adapted with modifications from [24])

Figure 11: Traffic Classification and Mapping for IntServ Services
(Adapted with modifications from [24])

Cross layer approaches provide advantages in terms of exchanging information between layers that might help to allocate resources efficiently. However having cross

layer scheduler at the SS using MAC and PHY information can be complex and expensive. We in our work have tried using application layer information such as end-end delay at the MAC layer to allocate resources efficiently according to QoS requirements.

2.3. Resource Allocation in WiMAX Multi-Hop Networks

2.3.1. MULTI-HOP RELAY SYSTEMS

The Wireless Metropolitan Area Network (WMAN) air interface standard IEEE 802.16 [1] specifies a low-cost “last-mile” Point to Multi Point (PMP) wireless broadband access technology. However the broadband coverage of the base station is limited by low SNR and shadowing at the cell boundaries. Hence there is a need to extend the coverage of the traditional PMP network. There has been many wireless coverage extension scenarios and techniques proposed in literature for traditional cellular networks.

The authors in [25] proposed a new architecture for multi-hop cellular network (MCN). The proposed MCN preserves the benefit of the conventional single-hop cellular network (SCN) and also provides the flexibility of ad-hoc networks, where transmission through mobile hosts is allowed. The MCS reduces the fixed number of base stations required and found to improve the throughput performance while

limiting path vulnerability encountered in ad-hoc networks. The MCN and SCN were also analyzed in terms of their hop-count, hop-by-hop throughput, end-end throughput and also the mean number of channels required. It was found that in the multi-hop network the throughput is not only better than that of single hop, but also increases with reduction in transmission range. But such systems where mobile user nodes act as routers have numerous short comings. There is a need for special routing protocol to route packets through the mobile stations to the required destination. There exist security problem because other users have access to data. Battery life of mobile nodes is an issue if they are going to be used as intermediate routers even when they are not sending their own data.

Another such paper [26] aims at efficiently designing a multi-hop wireless backhaul to connect multiple wireless access points to a wired gateway. The work [26] provides a generalized link activation framework for scheduling packets over this wireless backhaul, such that any existing wire-line scheduling policy can be implemented locally at each node of the wireless backhaul. The authors propose an even-odd link activation framework to increase throughput in a wireless mesh where interferences can be a substantial problem. The authors also provide techniques for determining good interference free routes within the scheduling framework, given the link rates and cross-link interference information. They also show that when multi-hop wire-line scheduler with worst case delay bounds (such as WFQ or Coordinated EDF) is implemented over the wireless backhaul, their scheduling and routing framework guarantees approximately twice the delay of the corresponding wire-line topology.

An article [27] provides a summary of numerous contributions to present an overview of important topics in the context of relay networks. It covers different approaches to

obtain benefits from multi-hop communication via relays, such as solutions for radio range extension in mobile wireless broadband cellular networks and solutions to combat shadowing at high radio frequencies. Also, relaying is presented as a means to reduce infrastructure and deployment costs. The article [27] also shows that through the exploitation of spatial diversity, multi-hop relaying can enhance capacity in cellular networks. Different application scenarios have been studied for multi-hop communication and it has been proved that multi-hop communications can provide a substantial increase in link and network capacity when applied in areas suffering from heavy path loss. The authors also observe that novel radio resource management protocols for multi-hop networks are required to improve performance.

There have also been many schemes proposed for resource allocation in WiMAX mesh networks where each SS also functions as a router. Mesh networks can have a centralized or distributed resource allocation schemes. In a centralized scheme each subscriber station is allocated time slots by the BS for data transfer, while in a distributed scheme a mesh subscriber station interacts with its neighbors and decides upon the transmission and reception time slots.

The authors in [28] propose an algorithm for the data sub-channel allocation in WiMAX based mesh networks. Their goal was to increase spatial reuse, achieve high system throughput, and provide fair access for the subscriber stations. They proposed a scheduling problem whose objective is to maximize the system throughput and fairness. In this fairness model, the bandwidth allocation is based on the actual traffic demands in such a way that the capacity is not sacrificed by imposing the fairness constraints. They have simulated the scheduling algorithm and showed that it can

achieve higher system throughput than “hard” fairness schemes, while ensuring fairness among the SSs in the WiMAX mesh networks. A coverage extension scenario using WiMAX mesh is also presented in [29].

Another scheduling strategy to improve throughput of WiMAX mesh networks using spatial reuse is given in [30]. The authors use the odd-even scheduling outlined in [26] to reduce interference and enable concurrent transmission to increase the capacity and scalability of WiMAX mesh network. The odd-even approach was implemented in centralized mesh scheduling to have interference-aware data transmission. The mesh network was considered as a tree with the base station at the root of the tree and the BS controls the scheduling. The key assumption of the interference aware scheduling is that in each branch of the routing tree, when a layer is set to active state, its one-hop upper and either neighboring layers are set inactive to avoid interference. If this is achieved then non interfering links are made to transmit simultaneously hence increasing the throughput of the network. The authors in [30] show the proposed scheduling algorithm achieves better throughput and latency performance over the basic solution for IEEE 802.16 mesh network through simulations.

The advantages of wider BS cell-coverage, higher throughput, lower power consumption motivates the use of relaying concept in WiMAX. Some schemes such as tree topology based scheduling can be adapted to perform WiMAX relay-scheduling.

2.3.2. WiMAX RELAY NETWORK

Though the Mesh network with inter SS communication was part of the initial documentation, it was not made compatible with the PMP architecture, it is also less secure and reliable. The Mesh operation is also restricted to two hops [29]. Hence a new architecture is needed to extend the coverage of the IEEE 802.16 BS. The IEEE 802.16j task group is working on the construction of such a multi-hop tree based relay network that is compatible with the legacy standard [1]. It aims at achieving coverage extension without any functional change to the SS. The definitions of multi-hop terminologies are given by the IEEE 802.16j task group [31]. In June 2007, the task group came up with a base line document [32] which is not yet a standard. The document [32] provides a protocol stack relation between the BS, RS and Mobile Station (MS) for multi-hop relay scheme as shown in Fig. 12

Figure 12: Relay Protocol Stack for Simple Relay Station
(Adapted with modification from [32])

The communication between the MS and RS is similar to communication between SS and BS in [1]; however modifications have been made to adapt to multi-hop transmissions. The document [32] supports bandwidth request and grant procedure for

scheduling polling services. It provides guidelines for network entry procedures for mobile stations and relay-station hand-off.

The base line document [32] includes support for QoS classes like UGS, rtPS, nrtPS and BE but does not have any implementation details. The broadband wireless access (BWA) standard lays strong emphasis on quality of service (QoS) and defines different types of service classes to handle different types of multimedia traffic. Hence the multi-hop relay network must also be able to support these QoS services namely: Unsolicited Grant Service (UGS), Real-time Polling Service (rtPS), Non Real-time Polling Service (nrtPS) and Best Effort (BE).

Various research groups have studied the performance of relay systems with respect to performance gain. RS deployment and radio-resource reuse strategies for IEEE 802.16j multi-hop relay networks are investigated in [33] for a Manhattan layout. The RS deployment can also reduce transmission power to better conserve the battery power of mobile devices. Capacity enhancement through radio reuse is proposed in [33], but details of MAC layer implementation and required bandwidth request mechanisms were not presented.

The authors in [34] have proposed two schemes to integrate multi-hop communication into the IEEE 802.16 system [1]. The first scheme follows a centralized approach where the base station takes full control over the relays and their connections. The second scheme follows a distributed approach where the relay stations themselves control their connections and directly connected subscriber stations. The authors extend the IEEE 802.16 frame format shown in Fig. 3 to provide time division access

to the relay stations. They add a multi-hop sub frame along with the existing uplink and down link sub-frame. The position and duration of the multi-hop sub-frame will be indicated in the broadcast portion of the TDMA frame.

Figure 13: Frame Structure for Decentralized Control Scheme
(Adapted with modifications from [34])

In the de-centralized scheme, the relay appears like an ordinary SS to the BS and it appears as a regular BS to the SSs. The relays need to have full functionality of the BS and entirely controls the operations of the SSs associated with it. The BS allocates a part of the regular frame called multihop sub-frame for the operation of the relay station. The relay itself builds up an entire MAC frame within the multi-hop sub-frame allotted to it. A contention part is also provided by the BS and the RSs for network entry by SSs near them. The relay like a regular BS provides all necessary procedures to setup transport and management connections. Apart from maintaining transport connections, the relay also maintains its management connections with the BS. These connections are used by the relay to exchange signaling information, such as connection management or periodic network reports. The normal SS creates both management connection with the RS as well as management connection with the BS.

The relay opens an associated connection to the BS for every transport connection that a SS sets up with it. Both connections have the equal level of QoS. The RS maintains a mapping table to map the connection identifier (CID) of one hop to the CID of the other and vice versa. Figure 14 shows the connection establishment signaling for decentralized control scheme.

Figure 14: Connection Establishment in a Decentralized Control Scheme
(Adapted from [34])

In the centralized control scheme, the BS has full control over the entire relay network. The BS directly controls all SSs and all RSs that are associated to that BS. This will help to reduce the complexity of the RS since it just forwards packets from hop to hop but will increase the computational load on the BS especially when the network is large. The BS schedules all UL and DL transmissions, both on the first hop and on subsequent hops before the start of the frame and informs the relays on the broadcast portion of their allotted time slots. The relay forwards the relevant subset of the control information to the SSs in its sub cell. The centralized control frame structure is shown in Fig. 15.

Figure 15: Frame Structure for Centralized Control Scheme
(Adapted with modifications from [34])

The connection establishment is shown in Fig. 16. Whenever a SS establishes a management connection to the RS, the RS in-turn will establish a corresponding management connection to the BS. Each transport connection established by the SS with the RS results in a new transport connection of the relay. The connection mapping is taken care of by mapping tables.

Figure 16: Connection Establishment in a Centralized Control Scheme
(Adapted from [34])

In the centralized control scheme, the BS gathers all necessary information to schedule downlink (DL) and uplink (UL) bursts for the relays and for the SSs in all the hops. One critical draw back in this scheme is that for polling service, the SSs have to send their bandwidth request and it has to traverse all the way to the BS for it to schedule time slots appropriately. As the number of hops increases, this scheme

might perform badly since there will be a long bandwidth request grant delay in such cases.

The authors in [35] provided an analytical approach to dimension cellular multi-hop WiMAX network. They say that the introduction of relay stations may significantly enhance the link quality leading to throughput enhancements and coverage extensions. The dimensioning approach presented in this paper [35] uses traditional hexagonal cell structure and cluster, inter-cell interference is assumed to be generated by the six co-channel cells of the first tier. The authors consider two scenarios namely throughput scenario, coverage scenario. In the throughput scenario relays are placed within the BS's coverage area to increase the capacity. The coverage scenario studies the extension of the cell's coverage using relays placed at the border of the BS's transmission range. The results show that there is potential for relays to be used to extend the coverage area of BSs in cellular WiMAX networks.

The authors in [36] address the problem of path selection in a relay enhanced network. Concatenating MPDU's in the Up link (UL) and Down link (DL) bursts to reduce resource wastage has also been proposed in [37]. These concatenation and aggregation schemes significantly reduce overhead compared to the legacy standard.

Resource allocation and scheduling in a multi-hop relay network will be complex as it has to be compatible with the legacy standard. In a multi-hop network, inefficient bandwidth allocation at different hops may be a performance bottleneck for the whole system. Performance evaluation of bandwidth allocation in 802.16j is given in [38] along with a spectrum-efficiency based adaptive resource allocation algorithm. This

considers a linear network rather than a more general tree implementation. The details of the IEEE 802.16 MAC frame structure or the division of frame durations between various hops were also not considered. However, the bandwidth allocation is analyzed in terms of the spectrum utilization. The authors in [39] have done a capacity analysis of such a multi-hop system. They show that the capacity of the multi-hop degrades as the number of hops increases, but can be improved using higher level modulation schemes and turbo coding. Not many researchers have considered QoS over WiMAX multi-hop networks. The authors in [40] provide an overview of QoS considerations in next generation peer to peer networks.

An admission control scheme for providing QoS over multi-hop networks is given in [41]. The authors in [42] have developed a heuristic algorithm for admission control and apply the principle of odd-even scheduling outlined in [30] to perform interference aware scheduling in a multi-hop network. The scheduling algorithm is designed for the downlink and it activates links such that the packets can be delivered within their delay bound. However the work [42] does not discuss uplink scheduling which is more complex over a multiple hop network, and have not taken the IEEE 802.16 TDMA frame structure into consideration.

The literature review on multi-hop WiMAX network leads to a conclusion that there is a need for QoS scheduling. The scheduling strategy should be compatible and provide similar QoS as the PMP mode. Further the scheduler must be able to provide resources over multiple hops without much degradation to QoS as the number of hops increase.

Chapter 3

Queue Based Scheduling Algorithms for PMP Network

3.1. Queue Based Fixed Scheduling Algorithm

Form the literature review it is clear that WiMAX has a robust PHY and has multiple flows arising out of a single SS sharing the up-link. The GPSS scheme is found to perform well, but there has not been enough scheduling algorithms devised for the SS to provide resource sharing between the flows. Hence what is required is a simple-to-implement MAC scheduling principle which would provide QoS for multiple flows which share the same robust up-link. We have proposed resource allocation strategies for up-link scheduling in the IEEE 802.16 PMP mode.

3.1.1. DESCRIPTION OF FIXED SCHEDULING ALGORITHM

The IEEE 802.16 [1] series standard lays strong emphasis on providing QoS and has defined different traffic classes with separate QoS specifications for various types of multimedia traffic. Scheduling and resource allocation are necessary to ensure QoS guarantees along with admission control and traffic policing. However, [1] leaves the details of the design of those components for the QoS to the developers. This motivates the need for an effective scheduling mechanism to deliver QoS guarantees, especially for real-time traffic. The standard [1] has defined the following classes of

QoS services at the MAC layer namely Unsolicited Grant Service, real-time Polling Service, non-real time Polling Service, Best Effort Service. These service types have different QoS requirements which need to be met.

Niyato [21] has proposed an adaptive queue-aware algorithm for uplink bandwidth allocation and a rate control mechanism at a SS for polling services in a WiMAX network. Using this resource allocation scheme, the amount of bandwidth allocated for polling service can be adjusted dynamically according to the queue occupancy thresholds at SSs. Packet-level performances such as MAC protocol data unit (MPDU) delay and MPDU dropping probability can then be maintained at desired levels. Though [21] has provided a good framework for queue based scheduling, there is also a need to treat real-time and non-real-time flows differently under the polling scheme according to their respective QoS requirements. In [21], the real-time and non-real-time traffic use the same queue and therefore the system cannot distinguish between the different QoS requirements of these two different classes of traffic flows. It may also be noted that the burstiness of the traffic should not affect the scheduling policy and that the real-time flows should be immune to the request-grant delay in the polling scheme. Moreover, while real-time flows are delay intolerant, guaranteeing their required QoS should not be unfair to the non-real-time flows, i.e. by causing them to lose MPDUs due to buffer overflows. In this thesis, we propose a scheduling policy that can provide QoS guarantees (on the delay performance) to real-time flows while simultaneously ensuring good performance (i.e. low buffer loss) for non-real-time flows. The advantage of this scheduling scheme is that it distributes the bandwidth granted by the BS among the real-time and non-real-time queues according to their QoS requirements at the instant of scheduling at each SS. This helps to overcome the problem of request-grant delay and burstiness of the traffic flows.

A preliminary version of our work had been reported earlier in [43]. Our scheduling principle differs from the EDD and WFQ approaches used for different traffic classes in [17] and [18] since we schedule real-time and non-real time flows together taking advantage of their individual QoS requirements.

In the 802.16 polling scheme, the bandwidth request scheduler at a SS demands time slots for data transmission from the BS. In response to the request from the SS, the BS will schedule available bandwidth for the SS, which may be less than or equal to the number of time slots requested. The time slots requested at a particular moment depend on the total MPDUs in the rtPS/nrtPS queues at that time. It may be noted that, because of various constraints, the BS may not always be able to allocate bandwidth equal to the bandwidth requested. Hence, we need to have an efficient scheduling algorithm at the SS which can effectively distribute the available bandwidth provided by the BS between the real-time and non-real-time connections. This is done such that, while QoS in terms of maximum delay is nominally met for real-time traffic, the allocation is not unfair to non-real-time traffic in terms of their buffer requirements. We propose a scheduling algorithm for this purpose which can be used at a SS in a GPSS working mode. Here, the BS schedules each SS based on the aggregate bandwidth required by all the connections within the SS and the SS redistributes the granted bandwidth among the connections. This scheduling algorithm works based on the queue length of each flow at the time of scheduling and the maximum delay specification of each rtPS flow.

We consider only the UGS, rtPS and nrtPS service classes for the scheduling algorithm. Each traffic flow in a SS is mapped to one connection of one of the three (UGS, rtPS, nrtPS) service classes defined in [1]. This is done based on the port numbers and the source and destination addresses provided by the higher layer. In each frame, the BS allocates a certain number of time slots to each SS depending on the bandwidth requested by the SS as well as on the constraints imposed by the other SS in the system. In effect, we assume that while the bandwidth granted by the BS may or may not be equal to the bandwidth requested earlier, the BS will allot a part of the uplink frame of burst duration D_i to each SS_i ($i \in (1, 2, \dots, n)$) in proportion to the aggregate bandwidth grant and will indicate this in the UL_MAP of the downlink frame. For SS_i , given B_w bps as the UL bandwidth, the total UL frame-length available (in bits) will be

$$F_{tot} = D_i * B_w \quad (1)$$

Since the UGS flows are constant bit rate flows such as VoIP without silence suppression with stringent latency requirements, they are scheduled first based on their maximum sustained rate. If there are k UGS flows arriving into the SS for uplink, the total bandwidth allocated to all UGS flows is given by (2).

$$F_{ugs} = \sum_{j=1}^k B_j^{ugs} * (C_j^t - C_j^{t-1}) \quad (2)$$

Here, for flow j , $j \in (1, 2, \dots, k)$ at the SS, B_j^{ugs} is the maximum sustained rate of UGS flow in bits/second, C_j^t is the current time and C_j^{t-1} is the last bandwidth allocation time for flow j . This generally takes a fixed portion of bandwidth as the frame size is fixed. The remaining UL frame length F_{poll} that may be allotted for the polling services (rtPS and nrtPS) will be as given in (3). It may be noted that any part of F_{poll}

which is not used for rtPS and nrtPS services will be subsequently available for use by Best Effort (BE) traffic.

$$F_{poll} = F_{tot} - F_{ugs} \quad (3)$$

The polling service consists of both real-time and non-real-time connections. The crucial factor differentiating the two is that while a real-time flow has a strict maximum latency specification, a non-real-time flow does not have such a specification. Instead, the latter requires service only at a data rate fast enough so that none of its data is lost due to the finite queue size of the corresponding queue. Though the non-real-time flows do not have a latency requirement, realistically they cannot wait in the queue forever to be served. Here, we assume for convenience that MPDU's belonging to an nrtPS flow can only wait in the queue for at most max_mpdu_delay seconds, even though, in practice, they will not have any maximum latency defined. (The max_mpdu_delay parameter depends on the buffer length of the queues at the SSs.) Therefore, we propose to provide real-time traffic with high priority so that they are better able to satisfy their latency requirements. In order to achieve this, we define a parameter α_j for each real-time flow $j \in (1, 2, \dots, l)$ at the SS as

$$\alpha_j = max_mpdu_delay / max_latency_rtPS_flow_j \quad (4)$$

Here α_j may be considered as a design parameter to control the QoS of the j^{th} rtPS flow so as to obtain a desired value for its average delay.

In order to take the burstiness of the traffic flows into consideration, we take into account the queue state of each real-time and non-real-time flow for scheduling. Let N_j^{rt} be the number of MPDU's in the real-time queue j , $j \in (1, 2, \dots, l)$ and let N_j^{nr} be

the number of MPDU's in the non-real-time queue $j, j \in (1, 2, \dots, m)$ at the SS at the start of the UL frame. The total number of MPDU's for all non-real-time traffic flows in the SS at that instant would be

$$N^{nr} = \sum_{j=1}^m N_j^{nr} \quad (5)$$

We define a real-time allocation weight K as

$$K = \sum_{j=1}^l N_j^{rt} \alpha_j \quad (6)$$

and use this to allocate the total frame length for the all the rtPS flows at that SS as

$$F_{tot}^{rt} = F_{poll} \frac{K}{(K + N^{nr})} \quad (7)$$

The remaining frame length which is then allocated for the nrtPS flows at that SS will then be

$$F_{tot}^{nr} = F_{poll} \frac{N^{nr}}{(K + N^{nr})} = F_{poll} - F_{tot}^{rt} \quad (8)$$

The individual real-time flow $v, v \in (1, 2, 3, \dots, l)$, at that SS is then allotted frame length based on its queue length and α_v value. This will give more bandwidth to the flows which have strict delay requirements and more back logged MPDUs.

$$F_v^{rt} = F_{tot}^{rt} \frac{N_v^{rt} \alpha_v}{K} \quad v \in (1, 2, 3, \dots, l) \quad (9)$$

The individual non-real-time flows $w, w \in (1, 2, 3, \dots, m)$, is allotted frame length based on

$$F_w^{nr} = F_{tot}^{nr} \frac{N_w^{nr}}{\sum_{j=1}^m N_j^{nr}} \quad w \in (1, 2, 3, \dots, m) \quad (10)$$

This process is repeated at the beginning of every UL frame by every SS. The advantage of this approach is that the bandwidth provided to the real-time traffic is based on both its latency and queue size. At the same time, the nrtPS traffic can obtain additional bandwidth when the real-time traffic is less i.e. during inactive burst periods. This would also take care of traffic fluctuations between the bandwidth request periods at the SS. Since the bandwidth request/grant will take some time to be sent by SSs to the BS along with scheduling time at the BS, the rtPS traffic may actually vary within this period. However, the proposed algorithm would not be affected because the granted bandwidth is always redistributed as per the QoS requirements of real-time and non-real-time traffic at SSs at the granting instant. It should also be noted that each individual real-time and non-real-time flow is allotted bandwidth effectively based on the weights of its queues, which is analogous to that of a WFQ strategy.

3.1.2. ANALYSIS OF FIXED SCHEDULING ALGORITHM

To analyze the proposed scheduling scheme, we created a Markov chain based analytical model. Since UGS is a constant bit rate service and its impact can be easily accounted for, we only study the performance of the rtPS and nrtPS scheduling. We consider a PMP WiMAX network and model all the individual rtPS connection queues in the system into an equivalent global rtPS queue and the individual nrtPS connection queues into a global nrtPS queue. This is a simplifying approximation needed for computational tractability of the analytical mode used to study the gated cyclic queues of real-time and non-real-time queues. For these two global queues, we take the total number of MPDU's in the individual rtPS and nrtPS queues as the state

descriptor and define a set of service rules according to the scheduling algorithm outlined earlier. A brief description of this analytical approach is also available in [44].

The end-to-end delay experienced by a MPDU consists of two components. One is the queuing delay at the SS and the other is the access delay caused by the frame cycle. The queuing delay at the BS is a constant as the MPDU's received in the UL frame by the BS will be transmitted on the immediate DL incurring only access delay, if equal bandwidths are assumed for the UL and DL frames. For simplicity, we also assume a single α value for all the real-time flows. In order to calculate the queuing delay, the arrivals and departures from the global queues are modeled as follows.

1. MPDU arrivals are continuous and occur in both uplink (UL) and downlink (DL) frames. For each real-time and non-real-time queue, we assume that the actual average arrival rates used in subsequent calculations are known. Using these, the arrival probabilities may be calculated for each frame period comprising of one UL and one DL slot.
2. MPDU arrival process is assumed to be a Poisson process with probabilities of MPDU arrivals in a frame period given by

$$P_{rt}(n) = (\lambda_{rt} T_f)^n e^{-\lambda_{rt} T_f} / n! \quad (11)$$

$$P_{nr}(m) = (\lambda_{nr} T_f)^m e^{-\lambda_{nr} T_f} / m! \quad (12)$$

where $P_{rt}(n)$ is the probability of n arrivals at the real-time queue, $P_{nr}(m)$ is the probability of m arrivals at the non-real-time queue, λ_{rt} is the arrival rate of real-time MPDUs, λ_{nr} is the arrival rate of non-real-time MPDUs and T_f is the total frame time. The arrival processes to the real-time and non-real-time queues are assumed to be independent. Hence the total probability $P_{tot}(n, m)$ will be

$$P_{tot}(n, m) = P_{rt}(n)P_{nr}(m) \quad (13)$$

3. The servicing of the queues is assumed to be done from the beginning of every UL frame period as shown in Fig. 17.

Figure 17: MPDU Arrival and Departure Process at the SS

The total number of MPDU's de-queued from both queues during an UL frame period will be in proportion to the fraction of the bandwidth allotted to both the queues in the UL frame, i.e. if total bandwidth granted to real-time queue is B_{rt} and non-real-time queue is B_{nr} , then the total number of MPDU's de-queued in the corresponding UL frame is

$$N_u = \left\lfloor \frac{(B_{rt} + B_{nr})F_t^{UL}}{S_{MPDU}} \right\rfloor \quad (14)$$

where F_t^{UL} is the UL frame time in seconds, S_{MPDU} is the average MPDU size in bits and B_{rt} and B_{nr} are in bps. Though in practice the MPDU's may not be of fixed size of S_{MPDU} , we make this simplifying assumption for ease of analysis.

The service distribution between the real-time and non-real-time queues is based on the following rules:

- (i) No de-queuing is done if both queues are empty.

(ii) If only one of the queues is empty then the number of MPDU's de-queued from the non-empty queue is $\min(N_u, \text{number of MPDUs in that queue})$.

(iii) If both queues are non-empty then the number of MPDU's de-queued from the real-time queue is $\min(N_{rt}, n_{rt})$ where N_{rt} is given by

$$N_{rt} = \left\lfloor \frac{N_u \alpha n_{rt}}{(\alpha n_{rt} + n_{nr})} \right\rfloor \quad (15)$$

For simplicity, we assume a single UL real-time connection from each SS sharing the same α value with other SSs. Here, n_{rt} is the number of MPDUs in the global rtPS queue and n_{nr} is the number of MPDUs in the global nrtPS queue at the beginning of a UL period. It should be noted that (15) follows from (7) in the algorithm given earlier. From (8), the number of MPDUs de-queued from the non-real-time queue will then be $\min(N_{nr}, n_{nr})$ where

$$N_{nr} = N_u - \min(N_{rt}, n_{rt}) \quad (16)$$

Let (i, j) define the state of the Markov chain of the global queue, just prior to the start of the UL frame where i is the number of MPDUs in the rtPS queue and j is the number of MPDUs in the nrtPS queue at that instant. The state just prior to the beginning of the next UL frame can then be found based on the number of rtPS and nrtPS MPDU's arriving in the intervening frame time and the number (out of i and j) served from the respective queues during the UL period. The respective transition probabilities can then be calculated depending on the number served (given by (14)-(16)) and the arrival probabilities as given by (11) and (12). A simple illustration of these states and their transitions is given in Fig 18.

Figure 18: State Transition Diagram of Markov Chain

We produce a computer program to generate the Markov chain and find the corresponding state transition probabilities $p_{ij,kl}$ as the probability of transition from state (i,j) to (k,l) . (Details of this computer program have been presented in Appendix I.) These transition probabilities are then used to compute the equilibrium state probabilities π_{ij} $i \leq N$, $j \leq M$. To ensure a proper choice of N and M , we subsequently perform the computation again to ensure that increasing the limits (i.e. N and M) do not substantially affect the average number N_R and N_N in the rtPS and nrtPS queues.

$$N_R = \sum_{i=0}^N \sum_{j=0}^M i \pi_{ij} \quad (17)$$

$$N_N = \sum_{i=0}^N \sum_{j=0}^M j \pi_{ij} \quad (18)$$

The average queuing delays D_R and D_N in the rtPS and nrtPS queues are then respectively evaluated as

$$D_R = \frac{N_R}{\lambda_{rt}} \quad (19)$$

$$D_N = \frac{N_N}{\lambda_{nr}} \quad (20)$$

The average access delay also consists of two parts. The first part arises because an MPDU arriving at a queue at a SS will have to wait until the next UL frame to transmit even if there are no MPDUs ahead of it in the queue; this average delay τ_{SS} will be

$$\tau_{SS} = 0.5(T_U + T_D) \quad (21)$$

Here T_U and T_D are the respective lengths in time of the UL and DL sub-frames where the MPDU is assumed to arrive at any moment during the frame. Similarly, the MPDU may be assumed to arrive at the BS anytime during the UL sub-frame and will have to wait until the next DL sub-frame. The corresponding average delay τ_{BS} will be

$$\tau_{BS} = 0.5T_U \quad (22)$$

Assuming the average access delays to be the same for both real-time and non-real-time flows, the average end-to-end delay D_{rt}^{MPDU} experienced by a real-time flow MPDU and the delay D_{nr}^{MPDU} will be

$$D_{rt}^{MPDU} = D_R + \tau_{SS} + \tau_{BS} \quad (23)$$

$$D_{nr}^{MPDU} = D_N + \tau_{SS} + \tau_{BS} \quad (24)$$

3.2. A Threshold-Based Adaptive Scheduling Algorithm

The analytical model in Section 3.1.2 indicates that the average MPDU delay depends on the current state of both the rtPS and nrtPS queues. However, using fixed α 's, as proposed in Section 3.1, would not be able to take into account the transient traffic fluctuations in these flows and adapt to them dynamically. Ideally, α should be varied dynamically to reflect these transient conditions so that the delay guarantees to rtPS flows may be adequately supported while still serving nrtPS flows based on their buffer occupancy. The approach described in this section adjusts α based on the relative queue occupancies of the rtPS and nrtPS queues to keep the delays of rtPS flows within their maximum delay limit. It may be noted that here α is used as a tuning parameter to adapt to dynamic changes and is not computed in a static manner as in (4).

To illustrate the philosophy of this approach, consider the scheduling algorithm in Section 3.1 with a single value of α for all the rtPS flows. Let R denote the ratio of the current number of MPDUs in all the rtPS and all the nrtPS flows at the beginning of scheduling at an SS

$$R = N^{rt} / N^{nr} \quad (25)$$

where $N^{rt} = \sum_{j=1}^l N_j^{rt}$ and $N^{nr} = \sum_{j=1}^m N_j^{nr}$. Using R , (7) and (8) may be rewritten as

$$F_{tot}^{rt} = \frac{F_{poll} \alpha R}{(\alpha R + 1)} \quad (26)$$

and

$$F_{tot}^{nr} = \frac{F_{poll}}{(\alpha R + 1)} \quad (27)$$

Note that the ratio of frame lengths available to all rtPS and all nrtPS flows is $F_{tot}^{rt} / F_{tot}^{nr} = R\alpha$, i.e. the division of available bandwidth among real-time and non-real-time flows is based on the ratio of MPDUs in the respective flows and the parameter α . The primary objective of the scheduling algorithm is to give more bandwidth to rtPS flows when needed to guarantee their QoS in terms of their delay bound. If the rtPS flows have relatively more MPDUs in their queues than the nrtPS flows, they should indeed receive more bandwidth. However, if there is a transient burst in the nrtPS flows then R decreases. This causes the amount of bandwidth provided to rtPS queues to decrease and may cause an increase at least temporarily in their delay in case of fixed α if α is small. The rtPS flows may be protected against such burstiness by adjusting α based on the relative numbers in the queue so that α can be increased when R decreases. When R is high, α can be reduced to lower values since rtPS flows already have more bandwidth. Decreasing α when R is large would enable the algorithm to be fair to the nrtPS by protecting it against the burstiness of rtPS flows. However a minimum value of α would also need to be set so that even when rtPS flows have more MPDUs, the bandwidth available should still be enough to support their delay guarantee.

We propose to adjust α based on the current value of R at the SS so that the scheduler would be able to provide enough bandwidth for rtPS flows to guarantee QoS in terms of delay while allocating fair resources to nrtPS flows based on their queue size. R and α values will be computed during every UL at the SS. This would

enable the scheduler to take care of the transient fluctuations in the rtPS and nrtPS flows. Since we aggregate all the rtPS and nrtPS flows within a SS for scheduling, the algorithm will also be more scalable.

We propose a range-based approach for doing this instead of a continuous variation of α as the former would be more practical to implement in a real system. Specifically, α is adjusted as in (28), where the threshold values of R are $(R_1 > R_2 > R_3 \dots)$, α_0 is the initial value of α and δ is a small increment. (These parameters need to be appropriately chosen, as discussed subsequently.)

$$\alpha = \begin{cases} \alpha_0 & R > R_1 \\ \alpha_0 + \delta & R_1 > R > R_2 \\ \alpha_0 + 2\delta & R_2 > R > R_3 \\ \vdots & \vdots \end{cases} \quad (28)$$

Following the selection of α , the actual bandwidth division among real-time and non-real-time flows is done as before. If F_{poll} is the total number of bytes available to polling service then the total number of bytes allocated to rtPS flows is

$$F_{tot}^{rt} = F_{poll} \frac{N^{rt} \alpha}{N^{rt} \alpha + N^{nr}} \quad (29)$$

and the total number of bytes allocated to non-real-time flows is

$$F_{tot}^{nr} = F_{poll} \frac{N^{nr}}{N^{rt} \alpha + N^{nr}} = F_{poll} - F_{tot}^{rt} \quad (30)$$

The individual rtPS flows $v \in (1, 2, 3, \dots, l)$ may be allotted frame length (bytes) as in (31), based on their respective queue sizes

$$F_v^{rt} = F_{tot}^{rt} * \frac{N_v^{rt}}{\sum_{j=1}^l N_j^{rt}} \quad v \in (1, 2, 3, \dots, l) \quad (31)$$

while the individual nrtPS flows $w \in (1, 2, 3, \dots, m)$ are allotted frame lengths (bytes) as in (32)

$$F_w^{nr} = F_{tot}^{nr} * \frac{N_w^{nr}}{\sum_{j=1}^m N_j^{nr}} \quad w \in (1, 2, 3, \dots, m) \quad (32)$$

A Markov Chain based analytical model for this can also be constructed in the same way as in Appendix I. Note that here α will depend on the state (n, m) but otherwise the approach will be similar to that of Section 3.1.2 and (14) - (24) can be applied.

The values of R and δ may be iteratively computed so as to keep the rtPS delay less than the maximum latency specification of rtPS flows. Initially, the upper and lower ranges of threshold R can be coarsely computed by increasing α in small steps while decreasing R to achieve the objective of keeping the delay below a maximum limit. Once a coarse range of R is found, finer adjustments can be made by reducing the step size δ and decreasing the difference between the subsequent thresholds R_1, R_2, R_3 etc. Initially, R_1 can set as unity i.e. $R > 1$ and the corresponding value of α is computed such that decreasing its value further will not cause the rtPS flows to exceed the maximum delay limit. For this threshold, α_0 should be chosen large enough to provide adequate bandwidth to the rtPS flows to meet their delay requirements even if the queue occupancy or rtPS flows is less than or equal to that of the nrtPS flows. Subsequently, the value of δ is increased as R decreases till the point when the rtPS delay does not show any significant decrease thus finding the upper bound of α . It should be noted that α is increased only when the ratio of rtPS to nrtPS queue occupancies decreases. When the ratio increases, the scheduling parameter α_i is decreased to give relatively more bandwidth to the nrtPS flows. This would

implement an adaptive value of α , unlike the fixed value used in the basic scheduling algorithm of Section 3.1.

3.3. Delay Feedback Based Adaptive Scheduling Algorithm

Another alternative approach is to use a delay-feedback based scheduling which we proposed in [44]. It would be more logical to use a higher value of α only when the dynamic conditions are such that rtPS delay is high. When the rtPS flow's delay requirements are being adequately met, a lower value of α should be chosen to see if better performance may be provided for the nrtPS flows. This dynamic algorithm also needs to consider the queue occupancy of nrtPS flows so that (appropriately chosen) buffer bounds of the nrtPS flows are not exceeded when a higher value of α is chosen to meet rtPS delay requirements. This motivates a delay feed-back based adaptive algorithm proposed in this section where α is varied based on the feedback delay from the destination.

The delay feedback needed for this adaptive approach would come from the application layer of the destination SS. This calls for a cross layer approach similar to that of [24]. One way of sending delay feed back is through the primary management connection provided to each SS. The packet sent from the source will carry a time stamp. The receiver at the other end will calculate the time delay using the time stamp information and current network time and report the delay to the MAC layer. At the MAC layer the delay information is mapped to the received CID and will be attached to a DSC (Dynamic Service Change) message as shown in Fig. 19.

Figure 19: Delay Feed Back in IEEE 802.16 PMP Systems

This DSC message will then be sent to the BS over the primary management connection as described in [1] and [24]. At the BS, the CID is mapped to the source CID and the DSC message is forwarded to the source through the primary management connection of the source.

We can use this delay feedback to choose the appropriate value of α . Instead of defining α as in (4), we would like to define α as a dynamically varied parameter that would be assigned to rtPS flows. At the beginning of an uplink frame period, if the delay feed back obtained by the source SS is d_{f-1} then the candidate α to be used for scheduling in the subsequent frame period is decided based on

$$\alpha_c = \begin{cases} \alpha & d_{f-1} < D_l \\ \alpha + \delta & D_l < d_{f-1} \leq \max_rtPS_Delay \\ \alpha + 2\delta & d_{f-1} > \max_rtPS_Delay \end{cases} \quad (33)$$

where D_l is an appropriately chosen threshold, and α and δ are appropriately chosen value. However, choosing α as in (33), it should not cause the nrtPS buffer to overflow in the subsequent frame. Let Q_f^{nr} be the number of bytes in the nrtPS queue at the beginning of scheduling frame f , N_f^{nr} , N_f^{rr} be the respective number of packets in rtPS

and nrtPS queues and F_{poll} be the number of bytes granted to the polling service.

Using the known average arrival rate of nrtPS flows, we can estimate the probability of A or fewer nrtPS packets arriving during the subsequent frame as $P(k \leq A)$

$$P(k < A) = \sum_{k=0}^A P(k) \quad (34)$$

Choosing $P(k \leq A)$ appropriately (e.g. as 99.8% in our studies in Section 3.4), we estimate the value of A to meet this requirement. This is used in (35) to estimate the worst case target Q_{f+1}^{nr} as the predicted size of the nrtPS queue in the next frame.

$$Q_{f+1}^{nr} = Q_f^{nr} - F_{poll} * \frac{N_f^{nr}}{N_f^{nr} * \alpha_c + N^{nr}} + A * P_{avg}^{nr} \quad (35)$$

Here P_{avg}^{nr} is the average nrtPS packet length. If Q_{max}^{nr} is the maximum allowed buffer occupancy for the nrtPS queue then the value of α chosen during the current scheduling is decided by

$$\alpha_f = \begin{cases} \alpha_c & Q_{f+1}^{nr} \leq Q_{max}^{nr} \\ \alpha_{f-1} & otherwise \end{cases} \quad (36)$$

where α_{f-1} is the parameter α_f used in the previous frame. This is done so that

choosing a higher value of α following (33) still tries to ensure that the nrtPS flow buffer limit is not crossed.

The Flow Chart for the Delay Feedback based Algorithm in Section 3.3 is given in

Fig. 20

Figure 20: Flow Chart for the Delay Feedback based Algorithm

3.4. Results and Discussions of Scheduling Algorithms

Algorithms

Figure 21: WiMAX PMP Network for Analysis and Simulation

For analytical and simulation studies of the scheduling algorithms, we consider a WiMAX PMP network as shown in Fig. 21. The network consists of six SS nodes with one BS. Each SS operates three pairs of connections, i.e. rtPS, nrtPS and UGS, for transmission and reception. The start and end points of the connections are shown in Fig. 21 where a single line indicates a pair of connections for transmission and reception via the BS. All the nodes use a single channel in TDMA fashion and the frame is divided equally into UL and DL frames. The number of time slots (bytes) allocated to polling service during a frame is taken as the product of the average data rate and the frame duration. Table 2 gives the traffic parameters used for analysis and simulation.

**TABLE 2: Traffic Parameters for Simulation and Analysis rtPS Data Rate twice
that of nrtPS**

Traffic Intensity	Average Data Rate of rtPS (Poisson arrival)	Average Data Rate of nrtPS (Poisson arrival)
0.33	512Kbps	256Kbps
0.45	700Kbps	350Kbps
0.51	800Kbps	400Kbps
0.64	1.0Mbps	500Kbps
0.77	1.2Mbps	600Kbps
0.83	1.3Mbps	650Kbps

A constant bit rate traffic of 128Kbps was used for UGS flows. The proposed queue aware scheduling approach was simulated using the QualNet WiMAX simulator where our proposed algorithm was implemented and the end-to-end delays obtained for different traffic intensities.

3.4.1. Results for Fixed Scheduling Algorithm

For the fixed scheduling algorithm, a single value of α is used for all the real-time flows in the system. The total bandwidth available for of UL and DL is set at 30 Mbps with one channel. The TDMA frame size is set as 10ms and divided equally between UL and DL. The number of slots (bytes) allocated to polling service during a frame is taken as the product of the average data rate and the frame time. An MPDU of 300 bytes (fixed) is used for UGS traffic (CBR 128Kbps) and one with mean of 1000 bytes (exponentially distributed) is used for the polling services. The up-link has 15Mbps net bandwidth out of which 14 Mbps (PS) + 128*6 Kbps (UGS) is set aside for data traffic and rest is made available for management connections. 14Mbps is available on the up-link for polling services. The traffic intensity is calculated as the ratio of net data rate of PS flows on the up-link to the total bandwidth available for PS on the up-link. E.g. For rtPS flow of 512Kbps, nrtPS 256Kbps and six subscriber stations, the traffic intensity is calculated as $((512 \times 6 + 256 \times 6) \text{ Kbps} / 14 \text{ Mbps} = 0.33)$. The frame length allotted to each flow is calculated based on (2), (9) and (10). The respective queues are de-queued as long as the sum of the MPDU lengths de-queued is less than or equal to the frame length allotted to that flow. The analysis and simulation results for the end-to-end delays are compared in Fig. 22 for various traffic intensities. We use $\alpha = 2$ for the analysis and simulation (i.e., $\max_mpdu_delay = 22\text{ms}$ and $\max_latency_of_rtps_flow = 11\text{ms}$ for all the real time flows).

Figure 22: Comparison of rtPS, nrtPS Delay (Simulation and Analysis)

For the analysis, the number of MPDUs serviced per frame period is derived from the number of bytes allocated to the polling service queues in the simulation as given in (14). We used a state space of (11, 11) to generate the Markov chain. The end-to-end delay is computed as the above mentioned procedure. The simulations run for 3, 5, 10 and 15 minutes and the average end-to-end delays are computed on different traffic intensities for each SS. Then the average delay for rtPS and nrtPS for the entire network is computed and compared with the analytical result. For the given system, the average delay obtained through simulations and analysis has been shown in Fig. 22 with varying system traffic intensities. It can be seen that there is a close correspondence between the analytical and simulation results confirming the validity

of the analytical model and its approximations and assumptions. But at some points the analytical and simulated delay has small variation, this may be due to the fact that we truncated the infinite Markov chain. Another reason may be that we computed the average delays of rtPS, nrtPS over the whole network in the simulation and compared it against the analytical result where we consider a global queue.

Fig. 23 shows the comparison of UGS, rtPS and nrtPS delays against system traffic intensities. Here, the UGS traffic is a constant bit rate flow and is served at its maximum sustained rate without any bandwidth constraint. Therefore, it has the minimum end-to-end delay.

Figure 23: Delay Comparison of rtPS, nrtPS and UGS

The real-time flow has a higher bit-rate, twice that of non-real-time flow, hence it causes more occupancy in the queue. By our proposed queue-aware scheduling algorithm, service of queues is based on their buffer state and the real-time flow is effectively scaled up by $\alpha = 2$. The real-time flow experiences significantly lower delays than the non-real-time flow, which has delays almost similar to that of the UGS flows. Fig. 23 shows that, as desired, the delay of rtPS flow is within the maximum delay limit for the flow, which is 11ms. It may also be observed that the nrtPS traffic does not exceed its time limit to stay in the queue hence no MPDU is lost. The UGS jitter performance which is an important factor in VoIP voice communications has been shown in Fig. 24. It can be seen that the observed jitter is only a small fraction of the corresponding value of the average delay shown in Fig. 23.

Figure 24: UGS Traffic Delay Jitter

Figure 25: Comparison of Queue Based Scheduling Scheme and Bandwidth Request based Scheme.

Figure 25 compares the rtPS and nrtPS delays of the proposed queue based scheduling scheme and a basic bandwidth request based scheduling scheme outlined by [1]. In the bandwidth request based scheduling scheme the polling services (rtPS and nrtPS) are allotted up-link bandwidth based on what they had requested when they were polled during the previous frame. This bandwidth request was based on the current queue occupancy (number of bytes) of the rtPS and nrtPS flows during the previous frame. We used the same input data rates as shown in Table 1 to simulate the bandwidth request based scheduling. Figure 25 shows that the rtPS delay for the queue based scheduling is much lower than that of the bandwidth request scheme. In the bandwidth request scheme, the rtPS and nrtPS delays are almost the same and in some cases rtPS delay exceeds nrtPS delay showing no difference in QoS. This is because of the fact that there is no bandwidth sharing between rtPS and nrtPS like in

the case of queue based scheduling. Also rtPS and nrtPS delays in the bandwidth request scheme are generally higher compared to the queue based scheme due to the request-grant delay. Hence it can be seen that the queue based scheduling scheme provides better QoS especially for real-time traffic.

TABLE 3: Traffic Parameters for Simulation- rtPS Data Rate half that of nrtPS

Traffic Intensity	Data Rate of UGS (CBR)	Average Data Rate of rtPS (Poisson arrival)	Average Data Rate of nrtPS (Poisson arrival)
0.33	128Kbps	256Kbps	512Kbps
0.45	128Kbps	350Kbps	700Kbps
0.51	128Kbps	400Kbps	800Kbps
0.64	128Kbps	500Kbps	1.0Mbps
0.77	128Kbps	600Kbps	1.2Mbps
0.83	128Kbps	650Kbps	1.3Mbps
0.89	128Kbps	700Kbps	1.4Mbps

Figure 26: Delays of rtPS, nrtPS and UGS (Data Rate of nrtPS twice that of rtPS)

We also examined a scenario where the nrtPS traffic rate is twice that of rtPS. Table 3 gives the traffic parameters for this scenario and Fig. 26 shows the corresponding UGS, rtPS and nrtPS delays. It can be seen that while the average rtPS delay has increased over UGS delay, which is roughly similar for both scenarios, the average nrtPS delay is lower. This is because the queue occupancy of nrtPS is now larger and according to (7) and (8), a larger portion of the bandwidth is allotted to the nrtPS flows. However, this does not drastically affect the performance of the rtPS flow as their delays marginally cross the specified maximum delay limit only at the higher values of the traffic intensity. If desired, this may also be overcome by setting α to a

somewhat higher value to increase the bandwidth available to the real-time flow. Fig. 27 provides a comparison of rtPS and nrtPS delays for the two scenarios.

Figure 27: Delay Comparison of rtPS, nrtPS with $\alpha=2$ for Different Traffic Rates

The effect of the design parameter α is illustrated in Fig 28. It compares α values of 2, 3 and 4 for the scenario where nrtPS data rate is twice that of rtPS. It can be seen that if α is increased, the rtPS delay is considerably reduced while nrtPS delay increases only marginally. This is because the queue occupancy of the nrtPS flow is higher since its data rate is higher than rtPS. However, this effect is offset by increasing α which increases the bandwidth available for rtPS according to (7) and (8). It can also be observed that the nrtPS traffic delay does not increase by too large an amount and is still less than the maximum time that a MPDU can stay in the queue.

Figure 28: Comparison of rtPS, nrtPS Delays for $\alpha=2, 3$ and 4 (nrtPS Data Rate is twice that of rtPS)

Fig. 28 examines a scenario where the data rate of rtPS flows is twice that of nrtPS flow (Table 2). We can observe that there is a very negligible change in rtPS delay between $\alpha=2, 3$ and 4, while the nrtPS delay increases sharply for higher value of α . This can also be explained from (7) and (8). Since the data rate of rtPS is higher than that of nrtPS, its queue occupancy is also higher which means that more bandwidth has already been allocated to rtPS flows. Here, increasing α which increases the bandwidth available to rtPS flows will effectively starve the nrtPS flows for bandwidth while providing only a marginal improvement in performance for the rtPS flows.

Figure 29: Comparison of rtPS, nrtPS Delays for $\alpha = 2, 3$, and 4 (rtPS Data Rate is twice that of nrtPS)

The results of Fig. 28 and 29 also provide the motivation to suggest that dynamically changing α based on the queue occupancies of rtPS and nrtPS flows should lead to good overall system performance. This motivates the adaptive scheme presented earlier in Section 3.2.

3.4.2. Results for Threshold Based Scheduling

For the adaptive scheme of Section 3.2, based on relative rtPS and nrtPS queue sizes, we first need to compute the initial threshold R_l . The approach followed for this is outlined next. For $R > 1$, generally, when queue occupancy of rtPS is greater than that of nrtPS, initial value of α_0 is computed such that by decreasing α below α_0 , the rtPS delay will cross its maximum limit. From Fig. 29, where rtPS data rate is twice the nrtPS data rate, it can be seen that for $\alpha = 2$, the delay of the rtPS flows only marginally cross their maximum delay limit. Hence, we set $\alpha_0 = 2$. Subsequently, the value of δ is increased while decreasing R to the point when the rtPS delay does not show any significant decrease thus finding the upper bound of α and minimum value of R . Then we reduced the step size between the thresholds and the δ value in order to smoothen the variation of parameter α over the threshold range. It can be seen from Fig. 30 that having minimum threshold of R below 0.6 and increasing $\alpha > 3$ will only marginally improve the delay of rtPS compared to the case $R_{min}=0.7$ and $\alpha=3$, where the delay is already less than the maximum limit. It can also be seen that with $R_{min}=0.8$ and $\alpha=2.75$, the rtPS delay significantly increases beyond the maximum limit. Selecting $R_{min}=0.8$ and $\alpha=3$ is still able to keep the rtPS delay below the maximum limit. However, to have a smooth variation of α over the range of R , we chose the thresholds to be $(R_1=1, R_2=0.9, R_3=0.8, R_4=0.7)$, $\alpha_0=2$, $\delta=0.25$ and obtained the end-to-end delay following the procedure given in Section 3.2.

Figure 30: Comparison of Average Delays for Various Thresholds and α Values

We have used the traffic parameters in Table 2 to obtain the MPDU delay performance of the adaptive approach of Section 3.2. Fig. 31 shows the results obtained through simulations along with those obtained by using the approximate analytical approach. The analytical and simulation results using the same set of parameters for R , α_0 and δ have a close correspondence as can be seen from Fig. 31.

Figure 31: Comparison of rtPS, nrtPS Delays (Analysis and Simulation)

Fig. 32 and 33 show end-to-end delay comparison for rtPS and nrtPS delays between the basic scheme of Section 3.1 and the adaptive scheme of Section 3.2. It can be observed that the adaptive scheme works much better than the fixed scheduling scheme for both scenarios where rtPS data rate is twice that of nrtPS and vice versa. In both scenarios, the rtPS delay stays within maximum limit of 11ms. Fig. 32 shows that the adaptive scheme is much fairer to nrtPS flows providing them the excess bandwidth whenever the rtPS flows are less active even though this slightly increases the delay of the rtPS flows as compared to the case where $\alpha = 4$.

Figure 32: Comparison of rtPS, nrtPS Delays for Fixed and Adaptive Scheduling (rtPS Data Rate is twice that of nrtPS)

The adaptive algorithm also keeps the rtPS delay always below the maximum limit compared to the fixed one with $\alpha=2$ where the delay crosses the threshold marginally for high traffic intensity. In Fig.33, even though the nrtPS flows have higher queue occupancies by virtue of their higher arrival rates, the adaptive scheduling algorithm is able to provide good service to the rtPS flows keeping their delays below the maximum limit. Note that for fixed scheduling with $\alpha=2$, the delay does reach beyond the limit for the highest traffic value as shown in Fig. 33. The throughput of the fixed and adaptive schemes has been compared in Fig 34. It shows that the throughput of the two schemes is similar. Hence the adaptive scheme keeps

the rtPS delay below the maximum delay limit while providing the same throughput as the fixed scheme.

Figure 33: Comparison of rtPS, nrtPS Delays for Fixed and Adaptive Scheduling (nrtPS Data Rate is twice that of rtPS)

Figure 34: Comparison of Throughput for Fixed and Adaptive Scheduling Schemes

To show the performance of the algorithm for more practical real-time application, we generated a VoIP ON-OFF traffic for rtPS and Poisson traffic for nrtPS. The traffic model for rtPS has a voice activity period of 40%, during which it generates exponentially distributed traffic. The delay performance of the system for VoIP traffic is compared with the regular Poisson traffic in Fig 35. The abscissa shows the average data rate used for Poisson traffic and during the active period of VoIP traffic.

Figure 35: Comparison of System Delay Performance between VoIP and Poisson Traffic for rtPS

It can be seen from Fig. 35 that the nrtPS delay decreases drastically when VoIP is used as the rtPS traffic. This is due to two reasons. First since the voice activity of rtPS traffic is only 40%, excess bandwidth is available which is given to nrtPS traffic thereby reducing its delay. The second reason is that the adaptive algorithm is fair to nrtPS traffic providing it with the excess bandwidth available during the inactive periods of rtPS flows. It can also be seen that the rtPS delay is kept well within its delay bound. This scenario supports the suitability of the algorithm for real-world bursty traffic like VoIP with silence suppression.

3.4.3. RESULTS FOR DELAY FEEDBACK SCHEDULING

The delay feedback based adaptive scheme of Section 3.3 is implemented using $P(k < A) = 99.8\%$ and $Q_{\max}^{nr}$ as the maximum nrtPS buffer occupancy obtained for $\alpha = 4$. D_1 is set as 95% of max_rtPS_delay. Initial value of α is chosen as 2 and $\delta = 1$. The comparison of average delays using fixed and adaptive α is shown in Fig. 36. It shows that the rtPS delay can be decreased by using adaptive α compared to fixed value of 2. The important inference is that nrtPS average delay has also reduced as reasoned previously.

Figure 36 Comparisons of Delays for Adaptive and Fixed Scheduling

Fig. 37 and 38 compare the maximum queue occupancy of rtPS and nrtPS flows for different α and adaptive schemes. Both figures show that adaptively changing α considerably reduces the maximum buffer occupancy compared to using fixed scheduling, at the same time rtPS delay is kept within limit. Hence adaptive scheme is better in terms of delay and buffer occupancy than fixed scheduling. The adaptive scheme is also fairer to non-real time flows since it provides additional bandwidth to nrtPS flows if the delay of rtPS flow is being met. The buffer occupancy of rtPS flow is also considerably less than that of nrtPS though data rate of rtPS traffic is more. This indicates that the algorithm provides more resources to rtPS on an average to reduce its delay.

Figure 37: Comparison of Maximum Buffer Sizes for Adaptive and Fixed Scheduling for non-real time flows

Figure 38: Comparison of Maximum Buffer Sizes for Adaptive and Fixed Scheduling for Real Time Flows

We also simulated the scenario where rtPS data rate is half that of nrtPS using traffic parameters from Table 3 and obtained end-end delay values. The Fig. 39 shows the comparison of fixed and adaptive scheduling for this scenario. It can be seen that rtPS delay for adaptive scheduling is still under the maximum delay limit whereas it had exceeded the limit for fixed scheduling with $\alpha = 2$. The nrtPS delay has also decreased by using adaptive scheduling when compared to fixed scheduling with $\alpha = 4$.

Figure 39: Comparison of Delays for Adaptive and Fixed Scheduling for nrtPS Data Rate Twice that of rtPS

Chapter 4

802.16j Multi-Hop Scheduling

The literature review on multi-hop networks motivates the research for scheduling algorithm in WiMAX relay network. The scheduling algorithm should be capable of providing QoS to different traffic classes and also should be compatible with the PMP mode. This motivated our research to design a multi-hop scheduling algorithm for QoS in WiMAX networks. The details are given in the following sections.

The Wireless Metropolitan Area Network (WMAN) air interface standard IEEE 802.16 [1] specifies a low-cost “last-mile” Point to Multi Point (PMP) wireless broadband access technology. However the broadband coverage is limited by low SNR and shadowing at the cell boundaries. Hence there is a need to extend the coverage of the traditional PMP network. Though the Mesh network with inter Subscriber Station (SS) communication was part of the initial documentation for IEEE 802.16[1], it was not made compatible with the PMP architecture. Hence a new architecture is needed to extend the coverage of the IEEE 802.16 BS. The IEEE 802.16j task group [32] is working on the construction of such a multi-hop tree based relay network that is compatible with the legacy standard [1]. It aims at achieving coverage extension without any functional change to the SS. The base line document includes support for QoS classes like UGS, rtPS, nrtPS and BE but does not have any implementation details. The broad band wireless access (BWA) standard lays strong emphasis on quality of service (QoS) and defines different types of service classes to handle different types of multimedia traffic. Hence the multi-hop relay network must

also be able to support these QoS services, i.e. Unsolicited Grant Service (UGS), Real-time Polling Service (rtPS), Non Real-time Polling Service (nrtPS) and Best Effort (BE).

The IEEE 802.16 standard supports a frame-based TDMA system for the PMP network. The authors in [34] proposed both centralized and de-centralized schemes to divide the TDMA frame between different hops but do not consider multiple traffic classes or a tree based implementation. Motivated by prior literature, we propose a hybrid controlled scheduling architecture to allocate bandwidth to UGS, rtPS and nrtPS traffic classes in compliance with the legacy standard. The hybrid scheduling architecture will provide bandwidth on the up-link and down-link for different flows based on their individual QoS requirements, and will not need any change in the protocols used by the regular mobile station defined as per the IEEE 802.16e standard. The hybrid architecture will significantly improve the up-link scheduling which requires bandwidth request-grant mechanism that has not been considered in earlier literature. To the best of our knowledge, this is a novel approach addressing QoS issues in WiMAX relay networks with a new proposed solution for QoS provisioning in such networks within the framework specified by the IEEE standard.

The legacy IEEE 802.16 PMP system specifies different resource allocation mechanisms to handle different kinds of traffic. Here, a SS initiates the flow creation and sends information about the flow to the BS. The BS decides to permit flows based on the QoS information and admission control criteria and provides a Connection Identifier (CID) for the flow to the SS, i.e. each flow is registered at the BS. The BS further schedules UL and DL traffic according to the QoS requirements. For UGS

flows on the UL, the BS provides a fixed number of slots on each frame. For rtPS and nrtPS services, the BS implicitly polls each SS for bandwidth request information and allots slots in the subsequent frame based on the bandwidth request.

Figure 40: IEEE 802.16 PMP Frame Format

Fig. 40 shows the PMP frame format. The broadcast part of the frame is heard by all the SSs. It contains the DL_MAP which specifies the location and duration of DL data burst in the frame for a particular SS. The UL_MAP specifies the location and number of time slots allotted in the frame for a particular SS. Hence all SSs listen to the broadcast portion of the frame and receive or transmit data during the slots in the frame which have been designated to them by the BS.

To extend this frame structure for multiple hops, [34] proposes a multi-hop sub-frame in addition to the regular UL and DL sub-frames. Each BS or relay station (RS), after allocating transmission time slots to UL and DL sub-frames also provision a multi-hop sub-frame which can be used by the child RS in the next hop for scheduling. However, the details on the implementation of the frame division and how different service classes will be handled in a multi-hop tree structure have not been given. It is important to specify proper rules for handling QoS services in multi-hop network for it to be interoperable with the PMP network. Two types of RS control, namely

centralized and de-centralized, are given in [34] which have been explained earlier in Section 2.3.3. In the centralized case, the BS decides the frame slot allocation to every RS and SS at each hop. In the de-centralized case, each RS has full functionality of a BS and forms its own frame using the multi-hop sub-frame space provided to it. Both architectures have their own limitations in handling different types of traffic. In a centralized control, each SS must send its bandwidth request (for rtPS, nrtPS) to the BS, which may be H hops away. Here it will take $2H$ frames for a SS to receive a bandwidth grant for its bandwidth request from the BS. If a SS generates MAC Protocol Data Units (MPDU) at a fixed rate where this rate information is available to the BS, then bandwidth may be pre-allocated for this on different hops. This will reduce the request-grant delay. However, if the traffic streams have variable rate and are bursty, then the de-centralized scheme works better since an RS can then directly give bandwidth grants to its immediate child nodes using the available multi-hop sub-frame.

4.1. Hybrid Scheduling for Tree Based Multi-Hop WiMAX Network

Different service classes defined by the 802.16 standard have to be handled differently in a multi-hop network so that their QoS requirements can be met in a manner which remains compatible with the existing PMP network. To do this efficiently, we propose a hybrid control scheme; combining features of both the centralized and de-centralized schemes of [34]. Decentralized control is used for DL and UL of rtPS and nrtPS traffic, while centralized control is used for UGS traffic. The BS will directly handle the bandwidth allocation for UGS flows (centralized),

while the RSs will handle rtPS and nrtPS bandwidth allocations by themselves based on bandwidth requests. Since UGS is a fixed rate flow, the BS can predict the amount of resources needed in each layer (hop) to handle the UGS traffic. The rtPS and nrtPS flows have variable bit rates and generate bandwidth requests as per the standards [1]. These can be handled by their respective RSs reducing the bandwidth request grant delay. BE uses whatever bandwidth remains unused after servicing the UGS, rtPS and nrtPS flows.

Figure 41: Multi-hop Relay Topology

To illustrate the scheduling scheme, we may consider a multi-hop topology such as the one given in Fig. 41. In such a multi-hop network, each RS at layer n either connects to SSs which are served directly by it or to RSs at layer $n+1$, which may in turn serve other SSs or RSs at layer $n+2$. The BS is the root node of the tree and the branches may contain RSs and SSs. DL data is carried from the root node to the SSs and vice versa for the UL. (The intermediate RSs do not generate any data on their own [31].) Each SS has to register with the RS (or BS) which it hears the loudest [34]. When a new data traffic flow is created, the SS communicates this to its RS which is in turn responsible for registering the flow with the BS. SS mobility in this architecture can be accommodated by allowing the SS to deregister from a RS and re-

register with another RS with a stronger signal. The BS and the RSs are all aware of their respective sub-network (i.e. below them in the tree) since all flow establishments will pass from the bottom of the tree to its root. Based on its admission control criteria, the BS allows the flow establishment and provides a unique CID to each admitted flow. If the flow is UGS, the BS uses the flow rate specified by the SS, to pre-allocate time slots in the respective layers for fast transmission of data. For rtPS and nrtPS traffic, we assume that the RSs and SSs are able to send their bandwidth

Figure 42: Frame Format for IEEE 802.16j Multi-Hop Network

requests during each frame to their respective parent (BS or the higher layer RS). The bandwidth request referred to here is the queue size of the UL buffer at the start of the frame. The Bandwidth Request sent by a child node during frame k can only be scheduled by its parent node in frame $k+1$. As shown in Fig. 42, the RSs are assumed to construct their own frame from the multi-hop sub-frame provided to them. The scheduling algorithm needs to consider both the UL and DL scheduling for each of the traffic classes (UGS, rtPS, nrtPS). The DL time slot can be allotted simply based on the DL queue size of each of the traffic classes with strict priority (SP) provided to rtPS over nrtPS. Since the DL needs to be faster to support traditional internet streaming, it is served before the UL. The scheduling strategy also needs to consider

framing and allocation of multi-hop sub-frame time slots at each layer. In our proposed approach, we consider the scheduling of UGS, rtPS and nrtPS traffic in detail as given below.

At the beginning of each frame, a fixed frame size is available to the BS, say F time slots.

1. Each MPDU is assumed to occupy one time slot in a frame.
2. For each frame, the scheduling starts from the BS and continues to subsequent layers.
3. The BS first handles DL UGS. It assigns a fixed number of slots on each link to transmit a fixed chunk of UGS data in every frame. If a subscriber station SS_i is H_i hops away from the BS and the BS generates b_i^{DL} MPDUs (fixed) per frame for this SS, then the BS allocates b_i^{DL} slots on all the links between the BS and SS_i . If a link l falls in the route between the BS and m SSs, then the BS will allocate $\sum_{i=1}^m b_i^{DL}$ time slots on every frame to that link. Therefore, if there are totally n SSs in the multi-hop tree having active DL UGS connection each generating b_j^{DL} slots per frame, then the BS pre-allocates $F_{pre_allocate}^{UGS_DL}$ time slots for DL UGS as given by (37).

$$F_{pre_allocate}^{UGS_DL} = \sum_{j=1}^n b_j^{DL} H_j \quad (37)$$

This is indicated in the DL_MAP message and forms a part of the multi-hop sub-frame. This fixed allocation cannot be used by the RS or SSs for any other traffic class.

4. The BS pre-allocates UL UGS time slots in a similar fashion. If there are n SSs with active UL UGS connection each generating b_j^{UL} MPDUs per frame, then the

BS will pre-allocate $F_{pre_allocate}^{UGS_UL}$ time slots for UL UGS data transfer as given by (38).

$$F_{pre_allocate}^{UGS_UL} = \sum_{j=1}^n b_j^{UL} H_j \quad (38)$$

This is indicated in the UL_MAP and the reserved slots form a part of the multi-hop sub-frame. This allocation cannot be used by the RS or SS for any other purpose.

5. The UGS 2 and UGS 3 in Fig. 42 refer to aggregate UL and DL time slots that are pre-allotted to layers 2 and 3. This is passed on by the BS to its child RSs (e.g. RS1 and RS2 in Fig. 41) which control scheduling in layer 2. Since the BS and RS know the topology of their respective sub-trees, the BS can put the information in the MAP message about which time slots are allocated to which SSs and will pass on these slots only to the respective RS. For example, in Fig. 41, the BS assigns the slots reserved for nodes SS3, SS5, SS6 to RS1 and the slots reserved for SS4, SS7 and SS8 to RS2 on their respective multi-hop sub-frames. The BS also informs the RS about which SS these slots belong to in the MAP message so that the RS may allot them correctly. Since RS1 and RS2 know the topology of their sub-trees, they follow the same procedure followed by the BS.
6. The remaining time slots $F_{DL_avail}^{PS}$ in the frame are given by (39).

$$F_{DL_avail}^{PS} = F - \sum_{j=1}^n b_j^{UL} H_j - \sum_{j=1}^n b_j^{DL} H_j \quad (39)$$

These $F_{DL_avail}^{PS}$ time slots are used by the BS to handle rtPS and nrtPS flows in strict priority on the DL in layer 1. If the queue sizes of rtPS and nrtPS exceed $F_{DL_avail}^{PS}$, the BS first allocates available bandwidth to the rtPS queues and then allocates the remaining to the nrtPS queues.

7. Steps 4, 5, 6 and 7 will decide the frame allocation on the DL and UGS (UL & DL) in layer 1, as shown in Fig. 42. The BS will send this information to the immediate child nodes on the DL_MAP.

8. The BS will schedule the UL polling services using the remaining bandwidth

$F_{UL_avail}^{PS}$ given in (40).

$$F_{UL_avail}^{PS} = F_{DL_avail}^{PS} - F_{DL_alloc}^{PS} \quad (40)$$

Here $F_{DL_alloc}^{PS}$ is the actual number of time slots allocated to DL rtPS and nrtPS

flows in step 7. For UL data transfer, the BS uses the aggregate polling service

(PS) bandwidth request B_c^{PS} of the h_l child nodes in the immediate hop (Layer 1)

$c \in (1, 2, 3, \dots, h_l)$ during frame $k-1$, i.e. the BS receives a total bandwidth request of

$\sum_{c=1}^{h_l} B_c^{PS}$ from its h_l child nodes. If the time slots available $F_{UL_avail}^{PS}$ is greater than

this total bandwidth request, then the BS will allot bandwidth to each child node

equal to its bandwidth request. If the bandwidth requested is more than what is

available, then the bandwidth F_c^{PS} is given to each child node c in proportion to its

bandwidth request as given in (41).

$$F_c^{PS} = \left\lfloor F_{UL_avail}^{PS} * B_c^{PS} / \left(\sum_{c=1}^{h_l} B_c^{PS} \right) \right\rfloor, (c=1, 2, 3, \dots, h_l) \quad (41)$$

The bandwidth allocation information is sent to the child nodes through the

UL_MAP.

9. The time slots that is left, as $F_{multihop_avail}$ given by (42) will be available for the multi-hop sub-frame.

$$F_{multihop_avail} = F_{UL_avail}^{PS} - \sum_{c=1}^{h_l} B_c^{PS} \quad (42)$$

If there are r RS nodes in the next hop, and if each RS has v_r SSs with active rtPS or nrtPS connections in its sub-tree, then each of the r RSs is allotted $F_{multihop}^r$ time slots in proportion to the SSs they have in their sub-tree as given in (43).

$$F_{multihop}^r = F_{multihop_avail} * v_r / \sum_{i=1}^r v_i \quad (43)$$

This information will be encoded in the UL_MAP field and broadcast to all the child nodes.

10. The actual data transfer on Layer 1, between the BS and the immediate child nodes, occurs in UGS DL HOP1, DL HOP 1 for rtPS and nrtPS, UGS UL HOP1, UL HOP 1 for rtPS, and nrtPS as shown in Fig 42. The immediate child nodes are informed of their multi-hop sub-frames and the pre-allocated UGS for different layers on the UL_MAP.
11. After data transfer in the 1st layer, there is a small intermediate transmit-receive gap (ITG) for the RSs in the 1st hop to switch their transceivers. The RSs use the MAP information provided by the BS and the multi-hop frame allocated to them to form their own MAC frames in order to communicate with their respective child nodes in Layer 2. The RSs in the second and subsequent layers use the UL and DL UGS time slots allocated to them and their child nodes by the BS to perform UGS scheduling. They pass the information obtained from their parents about the UGS allocations on the next layer through their UL_MAP message which is broadcast to their child nodes (RSs).
12. The RSs construct their DL, UL and multi-hop sub-frames using the same strategy as those given for the BS in Steps 6-10. They do this in the multi-hop frame allotted to them, except that they only schedule rtPS, nrtPS DL and UL data traffic among their immediate child nodes. The RS can also downlink the MPDUs it has

received in the same frame. For this, the RS must be able to compute bandwidth allocations within the ITG period. The RS use the bandwidth requests obtained from their immediate child nodes in frame $k-1$ to do the UL scheduling in frame k .

13. If there are multiple RSs in Layer 2 with child RS nodes of their own (or in any subsequent layer), then there will be multiple UGS, UL, DL, multi-hop sub-frames which will be non-overlapping in time. For example in Fig. 41, Layer 2 has two RSs, RS1 and RS2, communicating with their respective child nodes RS3, RS4. Hence each parent (RS1 and RS2) will create its own separate frame consisting of UGS DL HOP2, DL HOP 2, UGS UL HOP2, UL HOP 2 and multi-hop sub-frames. As RS1 and RS2 create new frames from the non-overlapping slots provided to them by the BS, the newly created time slots will also not overlap.
14. The scheduling and framing will continue in the subsequent layers until all the time slots are exhausted and the BS starts a new frame. The SSs in each layer will not be allotted any multi-hop sub-frame. At the last hop, the remaining time slots, if any, available to the servicing RS will be provided to the SSs for UL. If there are no UL data available, the SSs will pad the UL transmission with NULL indicators.

From the algorithm given above, the minimum frame size needed to pre-allocate time

slots to all the UGS flows as given in (37) and (38) is $\sum_{j=1}^n b_j^{DL} H_j + \sum_{j=1}^n b_j^{UL} H_j$. This is

also the admission control criteria for UGS flows. If the total UGS flows in the network are such that (44) is satisfied, then all the UGS flows can be promised delay guarantees equal to the respective number of hops the desired SSs are from the BS in units of frame time (H_j frames).

$$\sum_{j=1}^n b_j^{DL} H_j + \sum_{j=1}^n b_j^{UL} H_j \leq F \quad (44)$$

This is because we assume that UGS MPDUs traverse one hop per frame, which makes the UGS delays dependent on the position of the corresponding SS in the multi-hop tree.

From the above description it can be seen that the SSs actually perceive the same frame format in Fig. 40 as prescribed by the IEEE 802.16 legacy standard and that the SSs also issue bandwidth requests and obtain grants from its immediate controlling RS (which it perceives as its BS) in a manner analogous to the PMP mode. This multi-hop relay network will therefore be interoperable with the PMP mode. The scheduling and framing scheme given above will also be able to handle different QoS classes specified by the standard [1]. Pre-allocation of time slots is essential to provide QoS guarantee to VoIP UGS connections. This proposed hybrid control scheme combines the advantages of both the centralized and decentralized control. The above algorithm is illustrated by flow-charts in Fig. 43 and Fig. 44. The Fig. 43 describes the scheduling operation at the BS while Fig. 44 describes the scheduling at the RS. Though the algorithm specified above (Strict Priority (SP) scheme) is simple to implement, the proposed strategy is such that as traffic intensity increases, the DL may monopolize the resources and unfairly starve the lower layer UL flows. Moreover, SSs which are farther away from the BS will suffer if the traffic intensity at the layers closer to the BS is high. This indicates the need to consider the fairness of allocating resources to lower layers and also take their bandwidth requirements into consideration while scheduling flows at the upper layers. In the following section, we provide a scheduling architecture and a modified EDD based solution to provide fair bandwidth to lower layers of the multi-hop tree.

Figure 43: Flow-Chart for Strict Priority Multi-Hop Scheduling Algorithm at the BS

Figure 44: Flow-Chart for Strict Priority Multi-Hop Scheduling Algorithm at the RS

4.2. QoS Scheduling Architecture for Tree Based Multi-Hop WiMAX Network

A detailed scheduling architecture of the RS is provided in Fig. 45. It uses the same frame format as shown in Fig. 42. The scheduling architecture considers both the UL and DL scheduling for each of the traffic classes (UGS, rtPS and nrtPS). Each RS has to deal with UL, DL and next hop scheduling. It also has to handle two-tier scheduling for the down-link and up-link with its child nodes. The first tier involves bandwidth allocation to UL and DL using the available resources and the second tier involves QoS scheduling among different flows as shown in Fig. 45.

Figure 45: Relay Station MAC Architecture

The scheduling strategy also considers framing and allocation of multi-hop sub-frame time slots at each layer. The details of scheduling of UGS, rtPS and nrtPS traffic are given next.

1. At the beginning of each frame, a fixed frame size is available to the BS, say F time slots.
2. Each MPDU is assumed to occupy one time slot in a frame.
3. For each frame, the scheduling starts from the BS and continues to subsequent layers.
4. The BS first handles DL UGS. It assigns a fixed number of slots on each link to transmit a fixed chunk of UGS data in every frame. If a subscriber station SS_i is H_i hops away from the BS, and the BS generates b_i^{DL} MPDUs (fixed) per frame for this SS then the BS allocates b_i^{DL} slots on all the links between the BS and SS_i . If a link falls in the route between the BS and m SSs, then the BS will allocate $\sum_{i=1}^m b_i^{DL}$ time slots on every frame to that link. Therefore, if in total there are n SSs in the multi-hop tree with active DL UGS connection where each generates b_j^{DL} slots per frame, then the BS pre-allocates $F_{pre_allocate}^{UGS_DL}$ time slots for DL UGS as given by (45).

$$F_{pre_allocate}^{UGS_DL} = \sum_{j=1}^n b_j^{DL} H_j \quad (45)$$

This is indicated in the DL_MAP message and forms a part of the multi-hop sub-frame. This fixed allocation cannot be used by the RS or SSs for any other traffic class.

5. The BS pre-allocates UL UGS time slots in a similar fashion. If there are n SSs with active UL UGS connection each generating b_j^{UL} MPDUs per frame, then the BS will pre-allocate $F_{pre_allocate}^{UGS_UL}$ time slots for UL UGS data transfer as given by (46).

$$F_{pre_allocate}^{UGS_UL} = \sum_{j=1}^n b_j^{UL} H_j \quad (46)$$

This is indicated in the UL_MAP and these reserved slots form a part of the multi-hop sub-frame. This allocation cannot be used by the RS or SS for any other purpose.

6. The *MAP decoder* at the RS shown in Fig. 45 assigns the pre-allocated bandwidth to the respective resource allocation units. The time slots reserved for UGS on subsequent layers is encoded by the *MAP generator* and passed to the next hop RS. The UGS 2 and UGS 3 in Fig. 42 refer to aggregate UL and DL time slots that are pre-allotted to layers 2 and 3. This is passed on by the BS to its child RSs (e.g. RS1 and RS2 in Fig. 41) which control scheduling in layer 2. Since the BS and RS know the topology of their respective sub-trees, the BS can encode the information in the MAP message about which time slots are allocated to which SSs and pass on these slots only to the respective RS, e.g. in Fig. 41, the BS assigns the slots reserved for nodes SS3, SS5, SS6 to RS1 and the slots reserved for SS4, SS7 and SS8 to RS2 on their respective multi-hop sub-frames. The BS also informs the RS about which SS these slots belong to in the MAP message so that the *MAP generator* at the RS may allot them correctly. Since RS1 and RS2 know the topology of their sub-trees, they follow the same procedure followed by the BS.

7. The remaining time slots $F_{DL_avail}^{PS}$ in the frame are given by (47) and are used by the BS to handle rtPS and nrtPS flows.

$$F_{DL_avail}^{PS} = F - \sum_{j=1}^n b_j^{UL} H_j - \sum_{j=1}^n b_j^{DL} H_j \quad (47)$$

The BS and the RSs use a modified EDD scheme to schedule down-link rtPS. Each real-time MPDU is assumed to be time stamped. A rtPS MPDU j waiting in the queue at the BS or RS is assigned its time slot if (48) is satisfied.

$$t^c - t_j^o + t_j^e \geq \max_Delay_Spec \quad (48)$$

Here t^c is the current time when the scheduling is taking place, t_j^o is the originating time of the MPDU at the BS and t_j^e is the estimated time the MPDU will take to reach its destination if it is provided a time slot now. The rtPS MPDU's are provided time-slots as long as the above condition holds or until there are no more time-slots available (i.e $F_{DL_avail}^{PS} = 0$). It should be noted that, unless the EDD scheme is used, an MPDU can traverse multiple hops in a single frame time and giving priority to rtPS traffic can strangle the nrtPS and subsequent layers. This is the first tier of DL rtPS scheduling taking place at the *downlink bandwidth allocator*. The second tier scheduling uses the allotted rtPS timeslots to schedule multiple rtPS flows based on EDD.

8. The resource allocation at the upper layers should be fair to the lower layers, otherwise the upper layer SSs can hog the bandwidth and cause buffer over flows at the lower layers. In order to take this into account we need to consider the bandwidth requirement of the lower layers while scheduling the nrtPS DL. In the legacy IEEE 802.16 system, bandwidth request are sent by all the SS to their serving BS. In the multi-hop system the BS or RS will be similarly able to obtain

bandwidth requests from all its child nodes along the tree but with an associated delay. The RS will receive a bandwidth request from a node h hops away from it with a delay of h frames as, on the up-link, a message can travel at-most one hop per frame. If the amount of bandwidth available F_{avail} after rtPS scheduling is as given by (49) then the bandwidth F_{DL}^{nr} allotted to nrtPS flows is given by (50).

$$F_{avail} = F_{DL_avail}^{PS} - F_{DL}^{rt} \quad (49)$$

$$F_{DL}^{nr} = \begin{cases} F_{avail} * N_{DL}^{nr} / \sum_{j=1}^n BWreq_j^{t-h_j} & (if..N_{DL}^{nr} > F_{DL}^{nr}) \\ N_{DL}^{nr} & (otherwise) \end{cases} \quad (50)$$

Here N_{DL}^{nr} is the total number of MPDUs in the nrtPS queue, $BWreq_j^{t-h_j}$ is the bandwidth request received from node j which is h_j hops away from the RS or BS which is obtained after h_j frames. In order to represent the resource requirement at the lower layers of the tree, we also include their down-link queue size in addition to the up-link queue size in the bandwidth request. A second tier scheduling allocates the bandwidth among multiple nrtPS flows based on WFQ method. The rest of the bandwidth $F_{UL_avail}^{PS}$ will be made available for the next layer up-link and then for multi-hop frame.

$$F_{UL_avail}^{PS} = F_{avail} - F_{DL}^{nr} \quad (51)$$

9. Steps 4 to 8 will decide the frame allocation on the DL and UGS (UL & DL) by a BS or SS. The bandwidth allocation is sent to the *MAP generator* module which will send this information to the immediate child nodes on the DL_MAP.
10. The BS will schedule the UL polling services using the remaining bandwidth $F_{UL_avail}^{PS}$. For UL data transfer, the BS or RS uses the aggregate polling service (PS) bandwidth request B_c^{PS} of the child nodes in the immediate hop

$c \in (1, 2, 3, \dots, h_1)$ during frame $k-1$; the BS/RS receives a total bandwidth request of

$\sum_{c=1}^{h_1} B_c^{PS}$ from its h_1 child nodes. If the time slots available $F_{UL_avail}^{PS}$ are greater than

this total bandwidth request, then the BS will allot bandwidth to each child nodes equal to its bandwidth request. If the bandwidth requested is more than what is available, then the bandwidth F_c^{PS} given to each child node c will be in proportion to its bandwidth request as given by (52).

$$F_c^{PS} = \left\lfloor F_{UL_avail}^{PS} * B_c^{PS} / \left(\sum_{c=1}^{h_1} B_c^{PS} \right) \right\rfloor, (c=1, 2, 3, \dots, h_1) \quad (52)$$

The bandwidth allocation information is sent to the child nodes encoded in the UL_MAP.

11. If the child node which receives the UL-MAP happens to be a RS, the *UL-MAP decoder*, sends the bandwidth allocation information to the *up-link bandwidth allocator*. The UGS bandwidth is pre-allocated and the queues are served in a FIFO fashion. The up-link rtPS is allocated bandwidth according to the modified EDD. Bandwidth is provided to all the real time MPDUs which might cross their deadline unless they are scheduled during the current frame until there is enough bandwidth available. The uplink nrtPS is provided with rest of the bandwidth and it is distributed in a WFQ fashion among different nrtPS queues.

12. The parent BS/RS allocates the remaining bandwidth $F_{multihop_avail}$ given by (53) to the multi-hop sub-frame.

$$F_{multihop_avail} = F_{UL_avail}^{PS} - \sum_{c=1}^{h_1} B_c^{PS} \quad (53)$$

If there are r RS nodes in the next hop, and if each RS has v_r SSs with active rtPS or nrtPS connections in its sub-tree, then each of the r RSs is allotted $F_{multihop}^{r}$ time

slots in proportion to the SSs they have in their sub-tree as given by (54).

$$F_{multihop}^r = F_{multihop_avail} * v_r / \sum_{i=1}^r v_i \quad (54)$$

This information will be encoded in the UL_MAP field and broadcast to all the child nodes. This multi-hop frame slots are utilized by the child RSs to communicate with their child nodes.

13. As an example, consider the actual data transfer on Layer 1 between the BS and the immediate child nodes. This occurs in UGS DL HOP1, DL HOP 1 for rtPS and nrtPS, UGS UL HOP1, UL HOP 1 for rtPS, and nrtPS as shown in Fig 42. The immediate child nodes are informed of their multi-hop sub-frames and the pre-allocated UGS for different layers on the UL_MAP. After data transfer in one layer, there is a small intermediate transmit-receive gap (ITG) for the RSs in the next hop to switch their transceivers. The RSs, use the MAP information provided by their parent BS/RS and the resource allocation on DL and UL from the *QoS Resource allocation unit* to form their own MAC frame in order to communicate with their child nodes. The RSs in the second and subsequent layers use the UL and DL UGS time slots allocated to them and their child nodes by the BS to perform UGS scheduling. They pass the information obtained from their parents about the UGS allocations on the next layer through their UL_MAP message which is broadcast to their children (RSs).
14. If there are multiple RSs in Layer 2 with child RS nodes of their own (or in any subsequent layer), then there will be multiple UGS, UL, DL, multi-hop sub-frames which will be non-overlapping in time. For example, in Fig. 41, Layer 2 has two RSs, RS1 and RS2, communicating with their respective child nodes RS3, RS4. Hence each parent (RS1 and RS2) will create its own separate frame consisting of UGS DL HOP2, DL HOP 2, UGS UL HOP2, UL HOP 2 and multi-

hop sub-frames. As RS1 and RS2 create new frames from the non-overlapping slots provided to them by the BS, the newly created time slots will also not overlap.

15. The scheduling and framing will continue in the subsequent layers until all the time slots are exhausted and the BS starts a new frame. The SSs in each layer will not be allotted any multi-hop sub-frame. At the last hop, the remaining time slots if available to the servicing RS will be provided to the SSs for UL. If there are no UL data available, the SSs will pad the UL transmission with NULL indicators.
16. In steps 4 and 5, it is assumed that the admission control decision is made such that equation (55) is satisfied.

$$\sum_{j=1}^n b_j^{DL} H_j + \sum_{j=1}^n b_j^{UL} H_j \leq F \quad (55)$$

This will provide delay guarantee to the UGS flows as H_j frame times for UL and DL transfer. Thus UGS delay can be guaranteed both on the up-link and the down-link.

The above scheduling algorithm will be able to overcome the shortcomings of the hybrid algorithm (SP scheme) provided in Section 4.1 since it takes into consideration the bandwidth requirements of lower layers. It also uses a modified EDD scheduling for real-time flows which will only provide just enough resources to real-time flows. This is unlike the earlier algorithm which followed a strict priority. The excess bandwidth from this is made available to non-real time flows and lower level RSs. The above EDD algorithm is illustrated by flow-charts in Fig. 46 and Fig. 47. The Fig. 46 describes the scheduling at the BS while Fig. 47 describes the scheduling at the RS.

Figure 46: Flow-Chart for EDD Based Multi-Hop Scheduling Algorithm at the BS

Figure 47: Flow-Chart for EDD Based Multi-Hop Scheduling Algorithm at the RS

4.3. Results and Discussions of Multi-Hop Scheduling Schemes

We simulated the tree based IEEE 802.16j network of Fig. 41 using MATLAB to study its performance. First we studied the hybrid scheduling with strict priority (SP scheme) given in section 4.1 for down-link rtPS over down-link nrtPS and up-link rtPS over nrtPS without EDD for rtPS. A static scenario is used to show the working of the scheduling scheme and track the bandwidth allocation and data transfer at each layer. Here the BS injects DL data for each SS and each SS injects UL data for the BS at the beginning of the simulation and the proposed scheduling is applied. We consider a frame with 48 time slots where each MPDU occupies one time slot. For UGS, at the beginning of each frame, each SS injects one UL MPDU for the BS and the BS injects one DL MPDU for each SS. Similarly, for both rtPS and nrtPS, two MPDUs each are injected both for UL (from each SS) and DL (from BS to each SS). For this static traffic scenario, Fig. 48 shows the UL and DL queue sizes for each node and their time slot allocation at the beginning of their respective frames. Fig. 48a and 48b shows the DL scheduling in which bandwidth allocation is easily carried out since each node knows its DL queue size and available resource to be allocated. DL scheduling has highest priority and the DL data transfer is done within 3 frames. Fig. 48c shows the UL queue size of each node at the beginning of the respective frames and Fig. 48d the number of UL slots allocated to the respective nodes in that frame. In Fig. 48d, it can be seen that the scheduling mechanism pre-allocates slots to different nodes for UGS traffic to minimize the number of frames taken for UGS UL. In the first frame, no slot is allocated for UL polling service (PS) because no bandwidth request has yet been received. After the third frame all the bandwidth is available for UL PS. Note that since the RS and BS use the previous bandwidth

request to allocate resources, the allocation may not always exactly match the current queue lengths at the RSs and SSs, e.g. RS1 and RS2 sometimes get less resources than what they require while SS 5, 6, 7, and 8 get more.

Figure 46a: DL Queue Size

Figure 46c: UL Queue Size

Figure 46b: DL Slot Allocation

Figure 46d: UL Slot Allocation

Figure 48: UL, DL Queue Sizes and Time Slot Allocation

A more typical random traffic loading scenario has also been simulated with Poisson arrivals for, rtPS and nrtPS, UL and DL, in every frame. The algorithm outlined in Section 4.2 is used with the modified EDD strategy applied to rtPS scheduling. Each RS shown in Fig. 41 handles a UGS, rtPS and nrtPS flows on the up-link and down-link. For the rtPS traffic on the down-link we use the *max_Delay_Spec* to be 2 frames, for the Up-link we use the *max_Delay_Spec* to be 3 frames. It should be noted that on the DL an MPDU can make multiple hops in a single frame, while an MPDU can

make only 1 hop on the up-link in each frame. The end-end delays (average number of frames required for MPDUs from BS(SS) to reach the SS(BS) for each layer) for various layers on the UL and DL are shown in Fig. 49 and Fig. 50. If a node j is h_j hops from the BS and if on an average, it generates b_j^{UL} MPDUs/frame for the BS and the BS generates b_j^{DL} MPDUs/per frame for DL to it, then the total load offered is

$$\left(\sum_{j=1}^n b_j^{UL} * h_j + \sum_{j=1}^n b_j^{DL} * h_j \right) \text{ per frame when there are } n \text{ such nodes in the network. The}$$

normalized traffic intensity used as abscissa in Fig 49 and 50 is the ratio of this nett offered load to the frame length. The traffic rates for PS were varied to obtain different traffic intensities. For example traffic intensity 0.39 in Fig 49 is calculated as

	b_j	Number of Hops	Number of Nodes in Layer	Net Traffic Load at Layer
Layer 1	10	1	2	20
Layer 2	10	2	2	40
Layer 3	10	3	4	120
		Net Traffic		180
		Frame size		465
		Network Traffic Intensity		0.39

$$b_j = (1 \text{ UGS MPDU/frame} + 2 \text{ rtPS MPDU/frame} + 2 \text{ nrtPS MPDU/frame}) \times 2 (\text{UL and DL})$$

The simulation is done for 500 frames. The pre-allocation of slots ensures that the UGS traffic had a fixed delay equal to the number of hops both on the UL and DL (this is not shown in the figure but confirmed experimentally). It can be seen from Fig. 49 and 50 that as traffic intensity increases, DL performs better than UL as its priority is effectively higher. From Fig 49 it can be seen that the rtPS flow delay irrespective of the destination layer is close to 2 frames since we use an EDD principle outlined in Section 4.2 with a maximum delay parameter of 2. Since rtPS traffic requires soft

delay guarantees, unlike UGS which requires hard delay guarantee, the flows will be able to tolerate a small delay as traffic intensity increases.

Figure 49: DL Delay Comparison for Different Layers vs. Traffic Intensity

However, such low delay is not possible on the UL since (a) each MPDU can travel only one hop in a frame, and (b) there is an inherent delay due to the bandwidth request grant mechanism and the fact that the associated bandwidth allocation may not be adequate. Even then it can be seen that the delay is comparable to that of DL delay. As the traffic intensity increases we can see that the nrtPS delay increases drastically for layer 3, this is due to the lack of bandwidth available on the multi-hop frame for lower layers as the upper layer nodes utilize available bandwidth to serve their flows. However, since the EDD scheme is used in the rtPS scheduling and the

lower layer bandwidth needs are considered, this scheme is still better than the Strict Priority (SP) scheme where strict priority is given to rtPS over nrtPS and DL over UL.

Figure 50: UL Delay Comparison for Different Layers vs. Traffic Intensity

We compare layer-3 DL and UL delay for the EDD scheme and SP scheme in Fig 51 and Fig. 52 respectively

Figure 51: DL Delay Comparison for Layer 3 between EDD and SP

Figure 52: UL Delay Comparison for Layer 3 between EDD and SP

It can be seen from Fig 51 and 52 that the UL and DL delay performance is better for EDD scheme. The rtPS delay is close to the maximum limit of 2 frames for DL and close to the set limit of 3 frames for UL. The delay performance for both rtPS and nrtPS are significantly better in case of UL. This is primarily because the EDD scheme allocates bandwidth only to urgent MPDUs on the upper layers, and thus does not hog the bandwidth. This allows the lower layers to receive more time slots on the multi-hop frame. Secondly, the upper layer scheduler also takes into account the bandwidth requirement of the lower layers, and effectively trades off bandwidth to be allocated to the nrtPS MPDUs on the upper layers with granting more time slots on the multi-hop frames. Thus the EDD scheme outlined in Section 4.2 with bandwidth request consideration performs better than the SP scheme described in Section 4.1.

Chapter 5

5.1. Conclusion

In this thesis we have analyzed resource sharing strategies and QoS provision for IEEE 802.16 PMP mode and multi-hop networks. Resource allocation and scheduling is one of the critical problems in broadband wireless networks since the limited resource should be shared efficiently among multiple users to derive maximum benefit. We have done an extensive literature survey pertaining to the two topics and have found important contributions from other research workers. We have derived motivation for our work from this prior literature and have proposed resource allocation algorithms which are shown to perform well through our analysis and simulations.

In Chapter 3 we have proposed a novel packet scheduling algorithm to be implemented at the MAC layer at the SSs in WiMAX PMP networks. The proposed scheduling scheme can provide a better resource allocation between real-time and non-real-time traffic streams within the polling service framework specified by the IEEE 802.16 standard. The algorithm provides QoS guarantees with consideration of both the queue sizes of different traffic flows and the requirements on their latencies. It can provide more bandwidth to real-time traffic to meet their delays constraints while providing excess resources to non-real-time traffic. Our simulation studies demonstrate that the performance of our proposed algorithm appears to satisfy the QoS requirements of real-time traffic while providing reasonable transmission service to non-real-time traffic. We have shown that the proposed scheme works better than

the bandwidth request based scheduling by providing better QoS to real-time traffic. We have also developed an analytical model based on Markov chains to analyze the performance of the algorithm. We have extended the algorithm to provide dynamic scheduling by changing parameter α adaptively, depending on the relative queue occupancies of rtPS and nrtPS flows. We have also produced a set of analytical and simulation results for the threshold based adaptive scheduling. We have shown that the adaptive algorithm works much better than the fixed scheduling algorithm. We have further proposed a delay feedback based adaptive scheduling scheme that can provide even better resource allocation with fairness among real-time and non-real-time traffic streams within the Polling framework of IEEE 802.16. The virtue of the proposed algorithm is that it provides just enough bandwidth for real-time traffic to meet their QoS requirements (delay) while providing excess resources to non real-time traffic. We have also evaluated the performance of the proposed feedback based adaptive algorithm through extensive simulations and found it to perform better than the fixed scheduling algorithm.

In Chapter 4 we have proposed a hybrid scheduling scheme for IEEE 802.16j multi-hop relay network. We propose a MAC scheduling architecture for the Relay Station in the multi-hop WiMAX network. This takes into consideration different traffic classes mandated by the 802.16 standard and is also kept compatible with the legacy standard. The pre-allocation of time slots will provide QoS guarantees to VoIP UGS connections which is considered to be essential for these services. The modified EDD scheme used for rtPS scheduling provides fair resource allocation to lower layers of the tree. The scheduling scheme also takes into account the bandwidth requirements of the lower layers through bandwidth requests and has been shown to perform better

than strict priority (SP) scheme. The proposed hybrid control scheme is able to combine the advantages of both centralized and de-centralized control and provides strict delay guarantee to UGS flows while soft delay bound to rtPS flows both on the down-link and up-link.

5.2. Future Research Work Recommendations

- The scheduling algorithms in Chapters 3 and 4 were designed for UGS, rtPS and nrtPS flows. Best Effort is very differently handled by WiMAX standard. Hence it will be a challenging problem to integrate best effort service into the proposed scheduling schemes.
- For any QoS scheme to work well, the incoming traffic streams need to be monitored and controlled. Hence adding admission control and traffic policing schemes to scheduling schemes in Chapter 3 and 4 will enhance the performance of the scheduling algorithms and help in guaranteeing QoS such as strict delay-bound and buffer bound.
- Multicasting and Broadcasting can also be explored for Multi-Hop WiMAX networks. These will involve scheduling changes at each hop and can be handled in both centralized and decentralized fashion.

Author's Publications

Papers Published

K. R. Raghu, S. K. Bose, M. Maode, "Queue Based Scheduling for IEEE 802.16 Wireless Broadband," in Proc. of 6th Intl. Conf. on Information Communications and Signal Processing, Dec. 2007.

K. R. Raghu, S. K. Bose, M. Maode, "A Queue Based Scheduling Approach for WMAN with Adaptive Augmentation" in Proc. of IEEE Wireless Communications and Networking Conference, IEEE WCNC Mar. 2008.

Papers in Submission

K. R. Raghu, S. K. Bose, M. Maode, "Queue Aware Strategies for Scheduling in Broadband Wireless Networks" submitted to Springer Wireless Networks Journal.

K. R. Raghu, S. K. Bose, M. Maode, "QoS Architecture for IEEE 802.16j Multi-hop Relay Networks" submitted to IEEE Transactions on Vehicular Technology.

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APPENDIX I

ALGORITHM FOR CONSTRUCTING MARKOV CHAIN

The computer program algorithm used for building Markov chain for the scheduling is given below:

1. Initialize state space for global rtPS and nrtPS queues as N, M
2. Initialize the value of α
3. For each state i, j $i \in (0, 1, 2, \dots, N)$ $j \in (0, 1, 2, \dots, M)$ Poisson arrivals of n, m are considered $n \in (0, 1, 2, \dots, N)$ $m \in (0, 1, 2, \dots, M)$ into the rtPS and nrtPS queues respectively and the arrival probability is given by

$$P_n(n) = (\lambda_n T_f)^n e^{-\lambda_n T_f} / n!, \quad P_m(m) = (\lambda_m T_f)^m e^{-\lambda_m T_f} / m!$$

4. The next state at the beginning of UL frame boundary, is determined by current state i, j , arrivals n, m and the total number of MPDUs that can be dequeued N_u given by equation (14). The next state is determined as:

If $((i==0) \ \&\& \ (j==0))$

{

$Departure_rtPS=0, Departure_nrtPS=0;$

}

else if $(i==0)$

{

$Departure_rtPS=0, Departure_nrtPS=N_u;$

}

else if $(j==0)$


```

{
     $Departure\_rtPS = N_u, Departure\_nrtPS = 0;$ 
}

else

{
     $Departure\_rtPS = \text{floor}(N_u * (\alpha * i) / (\alpha * i + j));$ 
}

 $Departure\_nrtPS = N_u - \min(Departure\_rtPS, i);$ 

 $Nxt\_rtPS\_state = i - \min(Departure\_rtPS, i) + n;$ 

 $Nxt\_nrtPS\_state = j - \min(Departure\_nrtPS, j) + m;$ 

```

5. The probability of the next state ($Nxt_rtPS_state, Nxt_nrtPS_state$) is given by

$P_{tot}(n, m) = P_n(n)P_{nr}(m)$ which is the combined probability of n, m arrivals in the rtPS and nrtPS queues.

6. Similarly for each state the next state is computed and the probabilities are represented in terms of a matrix P .

$$P = \begin{bmatrix} P_{00,00} & P_{00,01} & \cdots & P_{00,NM} \\ P_{01,00} & P_{01,01} & & P_{01,NM} \\ \vdots & \vdots & \ddots & \vdots \\ P_{NM,00} & P_{NM,01} & & P_{NM,NM} \end{bmatrix}$$

7. Then equilibrium state probability vector is

$$\pi = [\pi_{00}, \pi_{01}, \dots, \pi_{0M}, \pi_{10}, \dots, \pi_{N0}, \dots, \pi_{NM}]$$

The Markov chain equation can be written as $\pi = \pi P$. Hence the Markov chain equations are of the form

$$\pi_{00} = p_{00,00}\pi_{00} + p_{01,00}\pi_{01} + \dots + p_{NM,00}\pi_{NM}$$

$$\pi_{01} = p_{00,01}\pi_{00} + p_{01,01}\pi_{01} + \dots + p_{NM,01}\pi_{NM}$$

:

$$\pi_{N-1M-1} = p_{N-1M-1,00}\pi_{00} + p_{N-1M-1,01}\pi_{01} + \dots + p_{N-1M-1,N-1M-1}\pi_{N-1M-1}$$

$$1 = \pi_{00} + \pi_{01} + \dots + \pi_{NM}$$

Hence the equilibrium solution for $\pi_{00}, \pi_{01}, \dots, \pi_{NM}$ is obtained by solving these simultaneous equations. Then equation (17) and (18) are used to compute the average number of MPDUs in the rtPS and nrtPS queues respectively.

A sample value for the probability matrix P of $N=10, M=10$ is given here

$$P = \begin{bmatrix} 0.835008 & 0.08849 & \dots & 0.00000 \\ 0.674089 & 0.158926 & \dots & 0.00000 \\ \vdots & \vdots & \vdots & \vdots \\ 0.00000 & 0.00000 & \dots & 0.038202 \end{bmatrix}$$

ABBREVIATIONS

ARQ	Automatic Repeat Request
BE	Best Effort Service
BS	Base Station
CBQ	Class Based Queuing
CID	Connection Identifier
CS	Service specific convergence sub-layer
DHCP	Dynamic Host Configuration Protocol
DL	Down Link
DL-MAP	Downlink Map
DSA	Dynamic Service Addition
DSC	Dynamic Service Change
DSD	Dynamic Service Deletion
DSL	Digital Subscriber Line
EDD	Earliest Delay Due Date
ertPS	Extended Real-Time Polling Service
FBWA	Fixed Broadband Wireless Access
FDD	Frequency Division Duplexing
FIFO	First-in-First-Out
FTP	File Transfer Protocol
GPC	Grant Per Connection
GPSS	Grant Per Subscriber Station
H-ARQ	Hybrid Automatic Repeat Request
HSDPA	High Speed Downlink Packet Access
HTTP	Hypertext Transfer Protocol
LOS	Line Of Sight
MAC	Medium Access Control Layer
MPDU	MAC Protocol Data Unit
MS	Mobile Station
NLOS	Non-Line Of Sight
nrtPS	Non-Real Time Polling Service
OFDM	Orthogonal Frequency Division Multiplexing

OFDMA	Orthogonal Frequency Division Multiple Access
PDA	Personal Digital Assistant
PDU	Protocol Data Unit
PHY	Physical Layer
PMP	Point to Multi Point
PS	Polling Service
QoS	Quality Of Service
RS	Relay Station
RSVP	Resource Reservation Protocol
rtPS	Real-Time Polling Service
SFID	Service Flow Identifier
SNMP	Simple Network Management Protocol
SNR	Signal To Noise Ratio
SP	Strict Priority
SS	Subscriber Station
TDD	Time Division Duplexing
TDMA	Time Division Multiplexing
UGS	Unsolicited Grant Service
UL	Up Link
UL-MAP	Uplink Map
WCFQ	Wireless Credit-Based Fair Queuing
WiMAX	World-wide Interoperability for Microwave Access

SYMBOLS

D_i	Burst Duration
F_{tot}	Total Frame Length
F_{ugs}	UGS Frame Length
B_j^{ugs}	Maximum Sustained Rate of UGS
F_{poll}	Polling Service Frame length

max_mpdu_delay	Maximum Delay Tolerance of any MPDU
α	Design parameter for Scheduling
$max_latency_rtPS$	Maximum Delay Tolerance of rtPS MPDU
N^{rt}	Number of MPDU's in rtPS Queue
N^{nr}	Number of MPDU's in nrtPS Queue
F_{tot}^{rt}	Total rtPS Frame Length Allotted
F_{tot}^{nr}	Total nrtPS Frame Length Allotted
$P_{rt}(n)$	Probability of n Arrivals at the rtPS Queue
$P_{nr}(m)$	Probability of m Arrivals at the nrtPS Queue
λ_{rt}	Arrival Rate of rtPS MPDUs
λ_{nr}	Arrival Rate of nrtPS MPDUs
T_f	Total Frame Time
F_t^{UL}	UL Frame Time in Seconds
S_{MPDU}	Average MPDU Size in Bits
B_{rt}	Bandwidth granted to rtPS queue
B_{nr}	Bandwidth granted to nrtPS queue
N_u	Total Number of MPDU's De-queued in a UL Frame
n_{rt}	Total Number of MPDUs in Global rtPS Queue at the beginning of UL frame
n_{nr}	Total Number of MPDUs in Global nrtPS Queue
$p_{ij,kl}$	Probability of Transition from State (i,j) to (k,l)
D_R	Average Queuing Delay rtPS Queue
D_N	Average Queuing Delay nrtPS Queue

T_U	UL Time
T_D	DL Time
τ_{SS}	Access Delay at Subscriber Station
τ_{BS}	Access Delay at Base Station
D_{rt}^{MPDU}	Average End-End Delay of rtPS MPDU
D_{nr}^{MPDU}	Average End-End Delay of nrtPS MPDU
R	Ratio of Number of MPDUs in All rtPS and All nrtPS flows
d_{f-1}	Feedback End-to-End rtPS Delay
Q_f^{nr}	nrtPS Queue Size in Bytes
Q_{f+1}^{nr}	Predicted nrtPS Queue Occupancy in Next Frame
P_{avg}^{nr}	Average nrtPS MPDU length
Q_{max}^{nr}	Maximum nrtPS Buffer
H	Number of Hops from the BS
b	Data Rate in Number of Time Slots/Frame
$F_{pre_allocate}$	Time Slots Pre-Allocated by BS for UGS
$F_{DL_avail}^{PS}$	Time Slots Available for Downlink Polling Service
B_c^{PS}	Bandwidth Request for Polling Service from node c
$F_{multihop_avail}$	Time Slots Available for Multi-Hop Sub- Frame
v_r	Number of SSs or MSs under RS r
max_Delay_Spec	Maximum Delay Specified for rtPS Flows
$BWreq_j^{t-h_j}$	Bandwidth-Request received from node j, h_j hops away from a RS or BS