TCP-aware Cross Layer Scheduling with Adaptive Modulation in IEEE 802.16 (WiMAX) Networks

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Abstract—In this paper, we propose Transmission Control Protocol (TCP)-aware cross layer scheduling algorithms in a multipoint-to-point network such as the uplink of an IEEE 802.16 (WiMAX) network. Inadequate bandwidth allocation to a TCP flow may lead to timeout and since TCP source drops its congestion window ($cwnd$) immediately after a timeout, it may affect the average throughput adversely. On the other hand, since the TCP source increases its $cwnd$ only linearly upon the availability of bandwidth, any excess assignment of bandwidth may remain underutilized. The proposed scheduling algorithms address this by allocating the resources based on $cwnd$ and TCP timeout. Moreover, since we focus on uplink scheduling, we consider that only flow level resource requirement is communicated to the Base Station (BS) instead of per packet information. The schedulers also take into account the wireless channel characteristics and are thus cross layer in nature. Through exhaustive simulations, we demonstrate that the proposed schedulers exhibit enhanced throughput and fairness properties when compared to that of Round Robin (RR) scheduler under different shadowing. We demonstrate a gain between 3.5\% to 15\% in throughput and 15\% to 25\% in channel utilization over RR scheduler under different shadowing.

Index Terms—Cross Layer, TCP, TCP-aware, Fair Scheduling, IEEE 802.16, WiMAX.

I. INTRODUCTION

Most of the current Internet applications can be broadly divided into two types: real-time applications and non real-time applications. Real-time applications require Quality of Service (QoS) guarantees in terms of minimum bandwidth and maximum latency from the network. Typically, these applications employ Universal Datagram Protocol (UDP) as the transport layer protocol. Non real-time Internet applications which constitute a significant percentage (80-90\% applications are TCP based) employ Transmission Control Protocol (TCP) as the transport layer protocol. We term these applications as TCP-based applications. Unlike real-time applications, the TCP-based applications do not operate within the strict QoS guarantee framework. Instead, a TCP source adapts its rate of transmission based on the feedback received from the sink. In this paper, we focus on scheduling algorithms that take into account the characteristics of a TCP flow and adjusts its bandwidth allocation accordingly. Specifically, we consider the setting of IEEE 802.16 based WiMAX network [3].

Due to the recent technological developments, Broadband Wireless Access (BWA) [3], [4] based services turn out to be advantageous than the traditional wired services in terms of fast deployment, flexible architecture, scalability, nomadic access and low cost. IEEE 802.16-2004 [3], is a fixed BWA standard for both multipoint-to-point and mesh mode of operation. The standard prescribes WirelessMAN-SC air interface in 10-66 GHz bands based on a single-carrier modulation scheme and WirelessMAN-OFDM, WirelessMAN-OFDMA air interfaces in the band of 2-11 GHz. Along with the fixed BWA, mobile BWA is also supported through the IEEE 802.16e-2005 [5] amendment.

IEEE 802.16 standard does not prescribe any particular scheduling algorithm, and thus network elements are permitted to implement their own algorithms at the Base Station (BS) for both uplink and downlink. We note that the requirements of uplink and downlink flows are different. In the downlink of IEEE 802.16, the BS has knowledge of the queues assigned to each Subscriber Station (SS), the arrival time of each packet and the individual channel condition of each SS. Hence, the BS can employ a scheduler similar to that of traditional scheduling schemes like Weighted Fair Queuing (WFQ) [6], Self-Clocked Fair Queuing (SCFQ) [7], Worst-case Fair Weighted Fair Queuing (WF^2Q) [8]. However, in the uplink transmission, the BS does not have packet arrival time and queue state information of SSs. Since communicating this information at packet level has significant overhead, these scheduling algorithms are not scalable and hence not suitable for uplink. From scalability perspective, Round Robin (RR) or its variants are suitable candidates for uplink scheduling.

In this paper, we consider a variant of RR scheduling. Since the channel state of SSs varies randomly across the frames, a RR scheduler would result in unfairness. Moreover, a RR scheduler does not allocate the resources based on TCP characteristics. If the scheduler does not assign the adequate bandwidth to a TCP flow, then it may lead to TCP timeout. Since a TCP flow drops its congestion window ($cwnd$) immediately after a timeout, inadequate bandwidth allocation and the consequent scheduling delay may affect the average throughput adversely. On the other hand, since the TCP source increases its $cwnd$ only linearly upon the availability of bandwidth, any excess assignment of bandwidth...
may remain underutilized. The proposed scheduling algorithm addresses this by allocating the resources based on $cwnd$ and TCP timeout. The scheduler also takes into account the wireless channel characteristics and is thus cross layer in nature. Since we focus on uplink scheduling, we consider that only flow level resource requirement is communicated to the BS instead of per packet information. We consider a polling based approach where the SSs are required to communicate their resource requirements with the BS once every few frames (called polling epoch). Since $cwnd$ size of a SS does not change for one Round Trip Time (RTT), we consider the polling epoch to be the minimum of $RTT$ of all flows.

### A. Related Work

Cross layer scheduling algorithms have been extensively studied in the literature [9], [10]. Optimal algorithms can be formulated as constrained optimization problems within the framework of Markov Decision Processes where the objective is to maximize a given utility subject to some QoS constraints. However, optimal algorithms are often computationally inefficient and several suboptimal algorithms have been proposed [11]. In this section, we review many such algorithms specifically proposed within the setting of IEEE 802.16 network. Most of these algorithms have been proposed for real-time traffic with QoS guarantees - the only exception being [12] where the authors have proposed a contention based TCP-aware uplink scheduling for IEEE 802.16 network. However, in [12], SSs do not transmit any bandwidth (BW) requests for scheduling, instead the BS measures the send rate of each individual flow dynamically and assigns resources based on the measured send rate. This kind of dynamic send rate measurement of all TCP flows at the BS in every frame can lead to scaling problem as the BS has to keep track of the states of all TCP flows along with the SSs’ requirements. Moreover, the scheme in [12] does not consider the time varying nature of wireless channel, the effect of $RTT$ variation on the requirement, and the effect of TCP timeouts. By assigning resources based on the send rate only, some flows might get starved resulting in frequent TCP congestion window ($cwnd$) drops and throughput degradation. On the other hand, in this paper, we propose a scheduling algorithm that not only takes $cwnd$ and TCP timeout into account but also the time varying wireless channel.

In [13], [14], the authors have analyzed the QoS support by providing differentiated services to applications such as Voice over IP (VoIP) and web services. They have employed Weighted Round Robin (WRR) for uplink and Deficit Round Robin (DRR) [15] for downlink scheduling. In [16], [17] have proposed fair uplink scheduling schemes for Multiclass Traffic in Wi-Max. The authors have also considered delay guarantee along with fairness. Scheduling based on dynamic weights of the IEEE 802.16 (WiMAX) flows have also been proposed in the literature [18]–[20]. In [18], the authors determine the weights of various flows based on the ratio of average data rate of the individual flows to the average aggregate data rate. [19] determines the weights based on the size of bandwidth requests, whereas [20] determines the weights of the individual flows based on the minimum reserved rate. Scheduling based on the delay requirements of Real Time Polling Service (rtPS) and Non Real Time Polling Service (nrtPS) have also been proposed in the literature [21], [22]. In [21], the authors propose a Delay Threshold Priority Queuing (DTPQ) scheduling scheme, which determines urgency of rtPS flows based on the delay of the Head of the Line (HoL) packet and a fixed delay threshold. The authors also consider adaptive delay threshold-based priority queuing in [22]. In [23], [24], the authors propose Deficit Fair Priority Queue (DFPQ) scheduling algorithm. [23] uses a deficit counter to maintain the maximum allowable bandwidth for each service flow. Based on the value of the deficit counter, it determines the priority of each flow. In [24], the authors have exploited the use of deficit counter for inter-class scheduling in IEEE 802.16 multipoint-to-point as well as mesh network. In [25], the authors propose an adaptive queue aware uplink bandwidth allocations scheme for rtPS and nrtPS services. The bandwidth allocation is adjusted dynamically according to the variations in traffic load and/or the channel quality. Researchers have also exploited the Opportunistic scheduling [26] in IEEE 802.16 (WiMAX) networks. Though the Opportunistic scheduling improves aggregate capacity of the network, performance of TCP-based application is degraded due to variable rate and delay, leading to unfairness among the flows.

In [27], the authors have proposed a Token Bank Fair Queuing (TBFAQ) [28] based scheduler for the downlink flows of an IEEE 802.16 network. It considers location dependent channel errors while scheduling and employs credit behavior of a flow to determine a priority index. Though this scheme provides fairness, it does not guarantee any delay while scheduling. [29] proposes an adaptive selective Automatic Repeat reQuest based scheduling scheme for nrtPS applications and uses an analytical model for parameter manipulation. Though it provides a trade-off between utilization and throughput, it is more suitable for the downlink scheduling in WiMAX networks.

In [30], the authors have proposed a QoS based uplink scheduling scheme in IEEE 802.16d/e (WiMAX) networks. It considers end-to-end QoS, both for real-time and non real-time applications and proposes a hybrid uplink scheduling algorithm, which is a combination of Priority (P) and Earliest Due Date (E) scheduling schemes. Even though it improves the utilization of the radio resources, normalized throughput drops substantially and access delay increases exponentially as the the number of system cells increase.

In [31], the authors have proposed a two-phased mechanism in which resource allocation and QoS scheduling are considered separately for OFDMA-based WiMAX networks. It considers system throughput optimization and QoS implementation for various types of traffic flows and ensures QoS by a priority-based bandwidth management scheme. Further, it provides admission control to provide QoS at the individual session level. Though this scheme provides QoS guarantee, it does not ensure fairness and high system utilization.

In [32], the authors have illustrated the performance of TCP
and UDP based applications through rigorous experiments conducted in an IEEE 802.16 deployed network as well as in test-beds. It has been observed that TCP applications suffer significantly as compared to UDP applications if the scheduling scheme does not consider the nature of TCP (TCP parameters). This key observation has encouraged us to work on scheduling schemes which are TCP-aware.

In this paper, we propose scheduling algorithms for the multipoint-to-point network that adapts its resource allocation based on TCP parameters - congestion window and timeout. The resource requirements are communicated during polling at flow level. The algorithms also exploit the wireless channel characteristics while maintaining fairness.

II. SYSTEM MODEL

We consider a multipoint-to-point network where multiple SSs are connected to one BS as shown in Fig. 1. This scenario may correspond to a single cell IEEE 802.16 (WiMAX) system. BS is the centralized entity responsible for scheduling the TCP flows. We assume that the SSs are TCP traffic sources. Each packet is associated with a TCP flow (source-sink pairs) and each flow is associated with a SS. Though this can be generalized to multiple flows per SS, we consider a single flow per SS. TCP acknowledgement (ACK) packets traverse from the sink to the source in the downlink direction. We assume that the ACK packets are very small in size and the downlink scheduler at the BS schedules these ACK packets without any delay.

Time is divided into frames. Each frame (of duration \(T_f\) in turn is composed of a fixed number of slots of equal duration \(T_s\)). We assume time varying wireless channel between a SS and the BS. We further assume that the channel gains between SSs and the BS are independent and identically distributed (i.i.d.) random variables and remain constant for a frame duration and change from frame to frame.

We assume that the individual channel state information is available at the BS in every frame. Let \(SNR_i\) denote the Signal power to Noise density Ratio (SNR) measured between SS and the BS. Packets can be successfully received if \(SNR_i\) is greater than a certain threshold \(SNR_{th}\). The value of \(SNR_{th}\) depends upon the modulation and coding scheme employed at the Physical (PHY) layer. In this paper, we consider both fixed as well as adaptive modulation schemes at the PHY layer.

We assume that a set \(I\) of TCP flows shares a network of \(I\) unidirectional links through the BS. The maximum possible data rate at link \(i\) denoted by \(R_i\), for \(i = 1, 2, 3, ...I\), is a function of \(SNR\) of the corresponding link. Since the channel state varies from frame to frame, \(R_i\) also varies from frame to frame.

III. UPLINK SCHEDULING WITH ADAPTIVE MODULATION

Before discussing the scheduling algorithm, we define the following terms.

- Connected Set: The set of SSs that has been admitted into the system through an admission control and connection set up phase is called connected set \((L_{connect})\). Let \(N\) be the cardinality of the connected set.
- Polling Epoch: It is defined as the interval that the BS chooses to poll the connected SSs. In the proposed scheduling algorithm, the polling is performed by the BS only once after every \(k\) frames.
- Schedulable Set: A SS is schedulable, if at the beginning of a polling epoch, it has a non zero \(cwnd\) and the \(SNR\) of its wireless link to the BS is above a minimum threshold \(SNR_{th}\). The set of such SSs constitute a schedulable set \(L_{sach}\). This set may change dynamically across the polling epochs. Let \(M\) be the cardinality of the schedulable set in a given polling epoch.
- Active Set: A schedulable SS is said to be an active SS in a frame of the polling epoch if its \(SNR\) is above \(SNR_{th}\) in the frame. The set of such SSs constitutes an active set \(L_{active}\). During a given frame of a polling epoch, the BS schedules traffic only from the active set. The membership of an active set may change dynamically across the frames of a polling epoch, whereas the membership of a schedulable set changes only across the polling epochs.

We divide the proposed scheduling algorithm into two phases: polling and slot assignment. BS polls all connected SSs once in every \(k\) frames and determines the schedulable set. In each of the subsequent \(k\) frames, the BS determines the list of active SSs and schedules only active SSs on a frame-by-frame basis. For slot assignment, the BS determines the weight of each active SS based on the values of its \(cwnd\), TCP timeout and accumulated deficit (as explained in the next section) and assigns slots based on its weight. At the end of \(k\) frames, the BS polls the connected SSs again and the above process is repeated.

The relationship between polling epoch and frame-by-frame scheduling is illustrated in Fig. 2. We discuss the slot assignment algorithm in the next section.

A. Slot Allocation

Consider a polling epoch. In the proposed algorithm, the BS maintains an indicator variable \(Flag_i\) for each SS; \(Flag_i(n)\) is 1, if \(SS_i\) is scheduled in frame \(n\) of the polling epoch. Let \(N_i(n)\) be the total number of slots assigned to \(SS_i\) in

\[\text{We assume one to one mapping between the flows and the links.}\]
frame \( n \). Let \( R_i(n) \) be the rate of transmission between \( SS_i \) and the BS in frame \( n \). If the underlying PHY layer employs fixed modulation scheme \( R_i(n) \) is considered to be fixed in a polling epoch, else, it varies on a frame by frame basis.

Let \( PL \) denote the length of a packet in bits. The amount of data (in bits) remaining to be transmitted by \( SS_i \) at the beginning of frame \( n \), \( D_i(n) \), is given by,

\[
D_i(0) = cwnd_i \times PL,
\]

\[
D_i(n) = D_i(n-1) - Flag_i(n-1) \times N_i(n-1) \times R_i(n-1) \times T_s, \quad \forall i \in L_{sch}, \, \forall n \geq 1.
\]

The number of slots actually required by \( SS_i \) in frame \( n \) will be \( \frac{D_i(n)}{R_i(n)} \). However, in the proposed algorithm, the slots are allocated in proportion to the weight of an \( SS \). Let \( W_i(n) \) denote the weight of \( SS_i \). After the determination of weights, the BS assigns slots to \( SS_i, \forall i \in L_{active} \) in frame \( n \) using:

\[
W_i(n) = \frac{1}{T_s} \times \min \left\{ \frac{W_i(n) \times T_f}{\sum_{j \in L_{active}} W_j(n)^{1/R_j(n)}}, D_i(n) \right\}, \quad \forall i \in L_{active}, \, \forall n \geq 1.
\]

The first term in the braces of (2) corresponds to the number of slots in proportion to the weight \( W_i(n) \), while the second term corresponds to the number of slots in proportion to the actual resource requirement \( D_i(n) \) of \( SS_i \). By using the \text{min} function in the above equation, the BS restricts the maximum number of slots assigned to any \( SS \) by its requirement. This ensures maximum slot utilization.

We now outline the determination of weight of each \( SS \) in the following subsections.

1) Weight Determination in TCP Window-aware Uplink Scheduling with adaptive modulation (TWUS-A): To determine a fair allocation of slots, we define the notion of quantum size \( Q \). The quantum size \( Q(n) \) in each frame of the polling epoch corresponds to the number of bits transmitted per schedulable subscriber. Specifically, the quantum size is updated as:

\[
Q(0) = \frac{R_{min} N_s T_s}{M},
\]

\[
Q(n) = \frac{1}{M} \sum_{i \in L_{sch}} \left( Flag_i(n-1) \times R_i(n-1) \right) \times N_i(n-1) \times T_s, \quad \forall i \in L_{sch}, \, \forall n \geq 1,
\]

where \( N_s \) is the total number of uplink slots and \( R_{min} \) is the minimum rate of transmission among all modulation schemes.

Fig. 2. Polling and Frame by Frame Scheduling

To keep a track of the number of slots assigned with respect to the quantum size \( Q \), we introduce the notion of a deficit counter \( DC_i \), similar to that of DRR [33]. The deficit counter \( DC_i(n) \) is updated in every frame as:

\[
DC_i(0) = 1,
\]

\[
DC_i(n) = DC_i(n-1) + Q(n) - Flag_i(n-1) \times R_i(n-1) \times N_i(n-1) \times T_s, \quad \forall i \in L_{sch}, \, \forall n \geq 1.
\]

From the above, we note that the deficit counter of \( SS_i \) in each frame is updated by the difference of the number of bits transmitted per schedulable subscriber and the actual number of bits transmitted by it. Thus, the deficit counter corresponds to the accumulated credit by a schedulable \( SS \). Since \( DC_i(n) \) can take negative value, we define the scaled deficit counter \( dc_i \) as follows:

\[
dc_i(0) = 1, \quad \forall i \in L_{active},
\]

\[
dc_i(n) = DC_i(n) + \min_{j} DC_j(n), \quad \forall i, j \in L_{active}, \, \forall n \geq 1.
\]

The weight of a \( SS \) is then determined in proportion to not only its resource requirements in terms of the number of slots as indicated by \( \frac{D_i(n)}{R_i(n)} \), but also to the accumulated credit in terms of the number of slots as indicated by \( \frac{dc_i(n)}{R_i(n)} \), i.e., the weight \( W_i(n) \) is determined using

\[
W_i(n) = \frac{D_i(n)}{R_i(n)} \times \frac{dc_i(n)}{R_i(n)}, \quad \forall i \in L_{active}, \, \forall n \geq 1.
\]

The slots are then assigned using (2). Inclusion of transmission rate \( R_i(n) \) in weight computation ensures in providing fair opportunity for the amount of data transmissions to each user, irrespective of its channel quality and transmission rate. In the next section, we incorporate TCP timeout information along with the congestion window size to determine the weight.

2) Weight Determination in Deadline based TCP Window-aware Uplink Scheduling with adaptive modulation (DTWUS-A): The basic idea in determining the weight is that an active \( SS \) whose TCP flow is approaching TCP timeout would be given a higher weight. Let \( TTO_i \) denote the time left to reach TCP timeout of \( SS_i \) at the beginning of a polling epoch. Note that the maximum value of \( TTO_i \) is the TCP timeout associated with the TCP flow of \( SS_i \).

For each schedulable \( SS \), we define \( deadline \) \( d_i \) to indicate the urgency of scheduling. At the beginning of a polling epoch, \( d_i \) of \( SS_i \) is initialized to \( TTO_i \). If \( SS_i \) is scheduled in frame \( n \), then \( d_i(n) \) remains unchanged, i.e., it takes the value of \( d_i(n-1) \). Otherwise, \( d_i(n) \) is decremented by one frame duration from its previous value. BS updates the deadlines of the schedulable flows as follows:

\[ ^2TCP \] flows generally start at random and hence different flows have different residual times to reach TCP timeout.
\(d_i(0) = \text{TTO}_i\), \(\forall i \in L_{\text{connected}}\),
\(d_i(n) = d_i(n-1) - T_f, \forall i \in (L_{\text{sch}} \setminus L_{\text{active}}), \forall n \geq 1, \tag{7}\)
\(d_i(n) = d_i(n-1), \forall i \in L_{\text{active}}, \forall n \geq 1. \tag{8}\)

If \(T_f\) exceeds \(d_i(n)\), then the deadline \(d_i(n)\) of \(SS_i\) is initialized to TCP timeout (\(\text{TTO}_i\)) of that \(SS\). In that case, TCP flow experiences a timeout before getting scheduled, resulting in reduction of \(cwnd\) to one. Thus, the \(BS\) incorporates the urgency measure \(d_i(n)\) in computing weight \(W_i(n)\) for \(SS_i\) as:

\[W_i(n) = \frac{\frac{D_i(n)}{R_i(n)} \times \frac{dc_i(n)}{R_i(n)/d_i(n)}}{\sum_{j \in L_{\text{active}}} \frac{D_j(n)}{R_j(n)} \times \frac{dc_j(n)}{R_j(n)/d_j(n)}}, \quad \forall i \in L_{\text{active}}, \forall n \geq 1. \tag{9}\]

Note that (9) is similar to (6) except for incorporating \(d_i(n)\). The use of the deadline in weight determination ensures that the weight of a \(SS\) with a smaller deadline is higher as compared to that of another \(SS\) which has a larger deadline. After the determination of weights, the number of slots assigned to \(SS_i\), \(\forall i \in L_{\text{active}}\) in frame \(n\) is determined using (2).

The pseudo-code of the proposed schedulers TWUS-A and DTWUS-A is presented in Algorithm 1. We have combined both schedulers by using \(\text{Flagdeadtime}\) which is set to one for DTWUS-A and is set to zero for TWUS-A.

IV. IMPLEMENTATION OF TCP-AWARE SCHEDULING

We consider Time Division Duplex (TDD) based IEEE 802.16 (WiMAX) network, in which each frame of duration \(T_f\) is divided into uplink and downlink subframes of durations \(T_{ul}\) and \(T_{dl}\) respectively. We consider adaptive modulation scheme at the PHY layer and employ Quadrature Phase Shift Keying (QPSK) modulation, 16-Quadrature Amplitude Modulation (QAM) and 64-QAM schemes. Let \(B\) denote the channel bandwidth. The maximum data rate \(R\) attainable for an Additive White Gaussian Noise (AWGN) channel can be expressed as:

\[R = B \times \log_2(1 + MI \times SNR), \tag{10}\]

where \(MI\) is the modulation index, which depends upon the desired Bit Error Rate (BER) and spectral efficiency of the modulation scheme. As discussed in [34], for a target BER \(p_b\) and spectral efficiency \(\frac{B}{f}\), modulation index can be expressed as:

\[MI = \left\{ \begin{array}{ll}
\frac{\ln(5 \times p_b)}{1.5}, & \text{if } \frac{B}{f} < 4,
\frac{\ln(0.5 \times p_b)}{1.5}, & \text{if } \frac{B}{f} \geq 4.
\end{array} \right. \tag{11}\]

Using (9) and (10), we determine the minimum \(SNR\) required \((\text{SBR}_{th})\) as:

\[\begin{align*}
\text{SNR}_{th} &= \frac{2^\frac{B}{f} - 1}{MI} \\
&= \frac{1}{\frac{1}{1.5} \times \ln(5 \times p_b)}, \quad \text{if } \frac{R}{B} < 4 \\
&= \frac{1}{\frac{1}{1.5} \times \ln(0.5 \times p_b)}, \quad \text{if } \frac{R}{B} \geq 4,
\end{align*} \tag{12}\]

For target BERs of \(10^{-5}\) and \(10^{-6}\), a channel bandwidth \((B)\) of 25 MHz, and for the data rates of 40, 80 and 120 Mbps (for QPSK, 16-QAM and 64-QAM modulation schemes respectively), we determine \(\text{SNR}_{th}\) using (12). These values are given in Table I.

In the proposed scheme, \(SSs\) are required to maintain a queue (per flow) at their interfaces. Packets residing in the queue of a \(SS\) is served in a first-come-first-serve basis. We assume that the \(BS\) has the channel state information.

Algorithm 1 : TCP-aware Uplink Scheduler with Adaptive Modulation

1: while TRUE do
2: Determine \(L_{\text{sch}}\) for the current polling epoch
3: \(\text{Flag}(0) \leftarrow 0 \forall i \in L_{\text{sch}}\)
4: \(D_i(0) \leftarrow cwnd, \forall i \in L_{\text{sch}}, \quad DC_i(0) \leftarrow 1, \quad dc_i(0) \leftarrow 1, \quad W_i(0) \leftarrow 0, \quad N_i(0) \leftarrow 0 \forall i \in L_{\text{sch}}\)
5: if \(\text{SchedulerType} = \text{TWUS} - A\) then
6: \(\text{Flagdeadtime} = 0, \quad d_i(0) \leftarrow 1 \forall i \in L_{\text{sch}}\)
7: else
8: \(\text{Flagdeadtime} = 1, \quad d_i(0) \leftarrow \text{TTO}_i, \forall i \in L_{\text{sch}}\)
9: end if
10: \(M \leftarrow |L_{\text{sch}}|\)
11: if \(n = 1\) then
12: \(Q(0) \leftarrow \sum_{i \in L_{\text{sch}}} N_i \times T_s\)
13: end if
14: \(k \leftarrow \min\{\text{RTT}_i\}, \quad T \leftarrow kT_f\)
15: Frame number \(n \leftarrow 1\)
16: while \(T > 0\) do
17: \(L_{\text{active}} \leftarrow \emptyset\)
18: for all \(i \in L_{\text{sch}}\) do
19: if \((\text{SNR}_{R_i}(n) \geq \text{SNR}_{th}) \land (D_i(n-1) > 1)\) then
20: \(L_{\text{active}} \leftarrow L_{\text{active}} \cup \{i\}\)
21: \(DC_i(n) \leftarrow DC_i(n-1) + Q(n-1) - R_i(n-1) \times N_i(n-1) \times T_s\)
22: if \(\text{Flagdeadtime} = 1\) then
23: \(d_i(n) \leftarrow d_i(n-1)\)
24: else
25: \(d_i(n) \leftarrow 1\)
26: end if
27: \(R_i(n) \leftarrow D_i(n-1), \quad DC_i(n) \leftarrow DC_i(n-1) + Q(n-1)\)
28: if \(\text{Flagdeadtime} = 1\) then
29: \(d_i(n) \leftarrow d_i(n-1) - T_f\)
30: else
31: \(d_i(n) \leftarrow 1\)
32: end if
33: end if
34: if \(d_i(n) \leq 0\) then
35: \(W_i(n) \leftarrow 0, \quad N_i(n) \leftarrow 0\)
36: end if
37: \(W_i(n) \leftarrow \sum_{j \in L_{\text{sch}}} W_j(n) \times T_f, \quad D_i(n)\)
38: end if
39: end for
40: for all \(i \in L_{\text{sch}}\) do
41: \(D_i(n) \leftarrow D_i(n-1) - N_i(n-1) \times R_i(n-1) \times T_s\)
42: \(dc_i(n) \leftarrow DC_i(n) + \sum_{j \in L_{\text{active}}} \text{dc}_j(n)/d_j(n)\)
43: Map \(R_i(n)\) to \(\text{SNR}_{R_i}(n)\) in Table I
44: \(W_i(n) \leftarrow \frac{\sum_{j \in L_{\text{active}}} W_j(n) \times T_f}{\sum_{j \in L_{\text{active}}} W_j(n)} \times \frac{D_i(n)}{R_i(n)}\)
45: \(N_i(n) \leftarrow \frac{1}{T_f} \times \min \left(\frac{W_i(n) \times T_f}{D_i(n)}, \frac{W_i(n) \times T_f}{R_i(n-1) \times N_i(n-1) \times T_s}\right)\)
46: end for
47: \(T \leftarrow T - T_f, \quad n \leftarrow n + 1\)
48: end while
49: end while
of each connected SS. This information is used by the BS to determine the schedulable set at the beginning of a polling epoch and to update the active set in every frame. At the beginning of every polling epoch, each SS conveys its requirements in terms of its current congestion window (cwnd(i)) size and time left to reach TCP timeout (TTO(i)) to the BS. BS in turn, determines the number of slots to be assigned based on the resource requirement, deficit counter and deadline values of each schedulable SS and determines the modulation scheme to be used on a frame by frame basis. BS conveys this information to each SS through the uplink map (UL_MAP) [3].

BS also determines the polling epoch k. Since cwnd of each flow remains fixed for one RTT\(^2\), the overall resource requirement of each SS also remains fixed for one RTT. Hence, the BS should choose a polling epoch of the order of one RTT. If it polls once per multiple RTTs (more than one), then there is a chance of TCP timeout resulting in cwnd reduction. In this paper, we choose polling epoch k to be the minimum RTT among all the flows. This enables each SS to convey its resource requirement at least once in every RTT.

The block diagram of the proposed uplink scheduler is shown in Fig. 3

![Block Diagram of TCP-aware Uplink Scheduler](image)

### V. EXPERIMENTAL EVALUATION OF TCP-AWARE SCHEDULERS WITH ADAPTIVE MODULATION

In this section, we describe simulation experiments that have been performed to evaluate TCP-aware scheduling. All the simulations have been conducted using implementation of TCP-aware scheduling within IEEE 802.16 (WiMAX) setting in MATLAB [35]. We consider a multipoint-to-point WiMAX network where 10 SSs are connected to a centralized BS as shown in Fig. 1. We simulate one TCP flow per SS. Each TCP flow starts randomly. The RTTs of the flows are updated using exponential averaging. Each SS is assumed to have a large enough buffer at its interface, such that the probability of buffer overflow is negligible. The frame duration \(T_f\) is set equal to 2 msec.\(^3\) The uplink subframe \(T_{ul}\) consists of 500 data slots (we assume that the number of control slots used is negligible). We consider both equal and unequal distances between SSs and the BS. For equal distances, the distances of all SSs from the BS are 1 km each and for unequal distances, the distances between SSs (\(SS_1 - SS_{10}\)) and the BS are 0.857 km, 1.071 km, 0.910 km, 1.230 km, 1.113 km, 0.956 km, 1.122 km, 0.884 km, 0.970 km and 1.216 km respectively.

We consider \(BER = 10^{-6}\) for the applications and use the \(SNR_{th}\) for selecting an appropriate modulation scheme as shown in Table I. The path loss exponent due to distance is set as \(\gamma = 4\). We simulate both shadowing as well as fast fading in our experiments. We also consider AWGN with Power Spectral Density (PSD) \(N_0 = 0.35\) (4.5 dB/Hz). The shadowing is modeled as Log-normal with mean zero and standard deviation (\(\sigma\)) of 8 dB. In each simulation run, the channel gain due to Log-normal shadowing is kept fixed for a duration of 50 frames. For fast fading, we consider Rayleigh fading model. The channel gain due to fast fading is modeled as complex Gaussian random variable or equivalently the power gain is an exponential random variable with mean \(\beta\). The coherence time of the channel is considered to be equal to one frame duration, i.e., the channel gain due to fast fading changes from frame to frame. The values of \(\beta\) and transmission power are chosen such that the expected \(SNR\) received at the cell edge is more than \(SNR_{th}\) required for transmission. We also repeat the experiments with different values of \(\sigma = 4, 6, 8, 10\) and 12 dB.

We conduct eight sets of experiments based on distance (equal and unequal) and the proposed schedulers with fixed and adaptive modulations. Note that TWUS and DTWUS are the fixed modulation versions of TWUS-A and DTWUS-A. In TWUS and DTWUS, we employ QPSK modulation scheme only, whereas in TWUS-A and DTWUS-A, we adapt the modulation scheme as shown in Table I. The system parameters used for simulations are presented in Table I. The value of each performance parameter observed has been averaged over 50 independent simulation runs, with the warm up frames (approximately 200 frames) being discarded in each run, to ensure that the values observed were steady-state values. We also implement Round Robin (RR) scheduler with both fixed and adaptive modulation schemes and compare the performance of TCP-aware schedulers with that of RR schedulers.

#### A. Simulation Results

1) Impact of \(cwnd_{Max}\): Since \(cwnd_{Max}\) value controls the TCP throughput, choosing its correct value in simulations is very important. A very high \(cwnd_{Max}\) will cause more

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### TABLE I

| Modulation Scheme | Data Rate | \(\frac{R}{BW}\) (bps/Hz) | \(SNR_{th}\) (dB) \(BER = 10^{-5}\) | \(SNR_{th}\) (dB) \(BER = 10^{-6}\) |
|-------------------|-----------|-------------------------|-----------------|-----------------|
| QPSK              | 40        | 1.6                     | 11.27           | 12.18           |
| 16-QAM            | 80        | 3.2                     | 17.35           | 18.23           |
| 64-QAM            | 120       | 4.8                     | 23.39           | 24.14           |

\(^2\)Typical TCP RTTs are in the range of 100 msec - 200 msec, whereas the frame length \(T_f\) in IEEE 802.16 is 0.5 msec, 1 msec or 2 msec.

\(^3\)Frame duration \((T_f)\) is equally divided between uplink subframe \((T_{ul})\) and downlink subframe \((T_{dl})\).
TABLE II
SUMMARY OF SYSTEM PARAMETERS

| Simulation Parameter | Value |
|----------------------|-------|
| Channel Bandwidth    | 25 MHz |
| Adaptive Modulation Schemes | QPSK, 16-QAM, 64-QAM |
| Bit Error Rate       | 10^{-10} |
| Path Loss Exponent (\(\gamma\)) | 4 |
| Frame Length \(T_f\) | 2 msec |
| Uplink/Downlink Frame Length | 1 msec |
| Number of Data Slots per \(T_u\) | 500 |
| Number of Frames Simulated | 40000 |
| TCP Type             | TCP Reno |
| Number of Independent Runs | 50 |
| Number of SSs        | 10 |
| Packet Size          | 8000 bits |

congestion and packet drops due to buffer overflow, whereas a small \(cwnd_{Max}\) will under-utilize the network. The value of \(cwnd_{Max}\) should be selected depending upon the PHY layer capacity, such that buffer overflow is minimized and network is appropriately utilized. Therefore, before conducting the experiments to verify the performance of the proposed schedulers, we perform experiments to determine the value of \(cwnd_{Max}\) at which the TCP throughput saturates and plot the results Fig. 4. For completeness, we plot the results of both TWUS and TWUS-A with equal distances in this figure. From this figure, we observe that TCP throughput remains constant (reaches saturation) once the \(cwnd_{Max}\) reaches 70 packets for TWUS-A, and 60 packets for TWUS. We choose \(cwnd_{Max} = 70\) and 60 packets for the TCP-aware schedulers with adaptive modulation and fixed modulation schemes respectively in the rest of our experiments.

\[
\text{Fig. 4. Average TCP Throughput vs. } cwnd_{Max}
\]

B. Comparison with Round Robin Scheduler
To compare the performance of TCP-aware schedulers with that of RR schedulers, we determine the average \(cwnd\) size, average TCP throughput, slot utilization and Jain’s Fairness Index (JFI) [36] achieved by each of the schedulers.

1) Average \(cwnd\) size and Throughput Comparison: In Fig. 5 and 6 (Fig. 7 and 8), we plot the average \(cwnd\) size and TCP throughput respectively under different shadowing with adaptive (fixed) modulation in consideration. From these figures, we observe that the average \(cwnd\) size as well as TCP throughput achieved by the TCP-aware schedulers are higher than that of RR schedulers under different standard deviation (\(\sigma\)) of Log-normal shadowing. Moreover, the average \(cwnd\) and throughput are higher in case of adaptive modulation. We also observe that as the \(\sigma\) of Log-normal shadowing increases, the average \(cwnd\) size as well as TCP throughput achieved by both RR and TCP-aware schedulers decrease. However, the rate of decrease of \(cwnd\) and TCP throughput is more in adaptive modulation than that in fixed modulation. Moreover, the gain in \(cwnd\) size of TWUS-A over RR varies between 5.5% to 16.5%, whereas the gain in TCP throughput varies between 3.5% to 16.5% for equal distance experiments.

Though we have illustrated the results for equal distance experiments, similar comparisons are also valid for unequal distance experiments and for fixed modulation experiments.

\[
\text{Fig. 5. Avg. } cwnd\text{ for TCP-Aware Schedulers vs. RR Scheduler}
\]

\[
\text{Fig. 6. Avg. Throughput for TCP-Aware Schedulers vs. RR Scheduler}
\]

2) Jain’s Fairness Comparison: In Fig. 9 we plot the variation of Jain’s Fairness Index (JFI) for the number of slots assigned to each SS for the TCP-aware and RR schedulers with different values of \(\sigma\) of Log-normal shadowing. From this
Log-normal Shadowing ($\sigma$ in dB)

Transport layer by the proposed scheduler, $\varsigma_i$ denote the Transport layer throughput received for user $i$ by the RR scheduler, $u$ is the total number of users and $\mathcal{M}$ is a positive real-valued function defined as:

$$\mathcal{M}(\vartheta) = \begin{cases} \vartheta & \text{if } 0 \leq \vartheta \leq 1 \\ 1, & \text{otherwise.} \end{cases}$$  \hfill (14)

Values of WCTFI and TFI varies between 0 and 1; 0 for a completely unfair system and 1 for a completely fair system at the Transport Layer. Since the TCP-aware schedulers provide higher throughput than RR scheduler at all shadowing conditions (Fig. 6 and 8), the values of both TFI and WCTFI of TCP-aware schedulers are 1. Therefore, TCP-aware schedulers are also Transport layer fair.

4) Slot Utilization: We also investigate the slot utilization of TWUS-A and DTWUS-A and compare it with RR scheduler. In Fig. 10 and 11 we plot slot utilization of TCP-aware and RR scheduler with different value of $\sigma$. From these figures, we observe that the slot utilization of TCP-aware schedulers is more than that of RR scheduler. In addition, utilization of TWUS-A (TWUS) scheduler is more than that of DTWUS-A (DTWUS) scheduler. Note that even though TWUS-A has higher channel usage as compared to that of DTWUS-A (c.f., Fig. 10), throughput achieved by DTWUS-A scheduler is more than that of TWUS-A scheduler. This is due to the fact that the chance of TCP timeouts in DTWUS-A is lesser than TWUS-A and hence less retransmission of packets, resulting in higher throughput achieved (c.f., Fig. 6).

We also observe that the slot utilization of TWUS-A and DTWUS-A scheduler varies between 70% to 85%, whereas that of RR varies between 54% to 70%. Moreover, as the value of $\sigma$ increases, the slot utilization decreases. This is because, when the channel is under heavy shadowing, the probability of not being scheduled in a frame is very high. This results in reduction in cwnd size thereby resulting in low utilization. Similar results are also observed for unequal distance experiments and for fixed modulation experiments.
delay in the proposed scheduling. Let \( p \) be the probability that \( SNR_i \geq \min\{SNR_{th}\} \), for any \( SS_i \). Hence, the expected number of polling epochs \( L \) which a \( SS \) needs to wait before becoming schedulable is,

\[
E[L] = \sum_{L=1}^{\infty} Lp(1-p)^{L-1} - 1. \tag{15}
\]

The expected number of frames that a \( SS \) waits is \( E[L] \times k \times T_f \). This corresponds to the average polling delay in the proposed scheduling algorithm. As discussed in [38], the average TCP send rate can be expressed as:

\[
B_w \approx \min\left( \frac{cwnd_{Max}}{RTT_w}, \frac{1}{RTT_w \sqrt{\frac{2p_w}{3}} + TO \min\left(1, 3 \sqrt{\frac{3p_w}{8}} \right) p_w (1 + 32p_w^2)} \right), \tag{16}
\]

where \( B_w \) is the TCP send rate or end-to-end throughput (in packets per unit time), \( b \) is the number of TCP packets acknowledged by one ACK, \( RTT_w \) is the average TCP round trip time, \( p_w \) is the packet loss probability and \( TO \) is the average TCP timeout value. In TCP, since the congestion window size can grow up-to \( cwnd_{Max} \), maximum TCP send rate is bounded by \( \frac{cwnd_{Max}}{RTT_w} \).

The end-to-end TCP throughput or send rate for our case can be modified by incorporating the polling delay into the round trip time. Accordingly, \( RTT_{wr} \) in the above expression can be modified to include the polling delay in scheduling. The new round trip time \( RTT_{wr} \) is expressed as:

\[
RTT_{wr} = RTT_w + E[L] \times k \times T_f. \tag{17}
\]

By replacing \( RTT_w \) by \( RTT_{wr} \) in (16), TCP send rate for WiMAX network \( (B_{wr}) \) can be expressed as:

\[
B_{wr} \approx \min\left( \frac{cwnd_{Max}}{RTT_{wr}}, \frac{1}{RTT_{wr} \sqrt{\frac{2p_w}{3}} + TO \min\left(1, 3 \sqrt{\frac{3p_w}{8}} \right) p_w (1 + 32p_w^2)} \right). \tag{18}
\]

A. Validation of TCP Throughput

We compare the average TCP send rate obtained in [18] with our simulation results. We determine the probability of loss \( (p_w) \) similar to that of [38]. We consider both triple-duplicate ACKs and TCP timeouts as loss indications. Let \( p_w \) be the ratio of the total number of loss indications to the total number of packets transmitted. From simulations (with \( \sigma \) of Log-normal shadowing as 8 dB and other parameters as shown in Table II), we observe that the average probability \( (p) \) that \( SNR_i \geq \min\{SNR_{th}\} \) is 0.87, \( \forall i \in I \). Using (17), we determine \( RTT_{wr} \). Then by using the value of \( RTT_{wr} \)},
obtained using (17) and \( p_w \) obtained above, we determine the average TCP send rate using (18).

To verify our model, we determine average TCP send rate for all four sets of experiments (TWUS-A with equal and unequal distances, DTWUS-A with equal and unequal distances). We plot the analytical and experimental TCP throughput at different \( cwnd_{Max} \) in Fig. 12-15. From these figures, we observe that the theoretical send rates determined using (18) and the send rate obtained by our simulations match very closely.

VII. SUMMARY AND OBSERVATIONS

In this paper, we have proposed scheduling algorithms for a multipoint-to-point network that adapts resource allocation based on TCP parameters-congestion window and timeout. The resource requirements are communicated during polling at flow level. The scheduler also takes into account the wireless channel characteristics and is thus cross layer in nature. Further, we have performed exhaustive simulations to investigate fairness and throughput behavior in WiMAX network setting. We have compared the performance of our scheduling schemes with that of RR scheduler under different shadowing. The proposed TCP-aware scheduling schemes perform better than RR scheduler in terms of slot utilization, fairness and throughput.

Though we have assumed that the downlink does not have any bandwidth constraint, in practice this assumption may not hold true. Hence, the effect of downlink congestion and the possible drop of ACK packet on the TCP throughput needs to be analyzed. In addition to this, as discussed before, the impact of scheduling of other class of traffic on scheduling of TCP traffic needs further investigation. Further, there is also a scope of extending this work for a high speed broadband mobile network such as IEEE 802.16m [39] based network. With mobility in place, scheduling of users to provide high data rates with hand-off margins is another area for investigation.

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