A Hierrachical Channel-Aware Scheduling Scheme Provisioning in IEEE 802.16e/Mobile WiMAX

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Abstract

One of the most significant promises in today's integrated broadband wireless networks such as WiMAX is in providing Quality of Service (QoS) guarantees. Using scheduling algorithms to provide QoS guarantees, wireless networks are able to integrate applications with a wide range of traffic characteristics. Deficit fair priority queue (DFPQ) provide directional based priority and service class differentiation. However, the environment for wireless networks is variable both in time and space so that the effectiveness of scheduling algorithm maybe invalidated by bad channels conditions. This paper proposes a channel aware deficit fair priority queue (CA-DFPQ) packet scheduling architecture for the QoS management for the Mobile WiMAX. The proposed scheduling is an extension of DFPQ found in literature, suitably modified to provide differentiated service even in non-ideal channel conditions. The modified algorithm together with a proposed mechanism for error compensation in WiMAX error-prone channels is designed to provide directional differentiation and service class priority.

Key Words: QoS, WiMAX, Scheduling, IEEE 802.16

1. Introduction

WiMAX (Worldwide Interoperability for Microwave Access) is a part of IEEE 802.16 [1] family of standards that target wide area line of sight (LOS) and non-line of sight (NLOS) broadband data access for fixed and mobile terminals. IEEE 802.16 standards specify the air interface at the physical (PHY) and the media access control (MAC) layers; the WiMAX forum defines additional specifications to support fixed, nomadic and mobile access [1]. Based on specific profiles from IEEE 802.16 standards the WiMAX forum also promotes and certifies compatibility of broadband wireless products to ensure diverse vendor interoperability [2].

802.16 WiMAX is designed to simultaneously support a diverse set of applications such as voice, video and data through a common bandwidth. Each of these applications has unique traffic requirements e.g. throughput, delay, jitter, loss rate that are collectively specified by Quality of Service Support (QoS). To successfully transmit these applications through the wireless link, their QoS requirements must be met and maintained throughout the duration of the transmission. The problem of assuring QoS then becomes how to allocate the limited bandwidth resources while at the same time ensuring an application's QoS requirement. To help in allocating bandwidth, traffic scheduling is used.

IEEE 802.16 standard does not specify how to support QoS features and requirements, such implementations are vendor specific [1], [4]. Therefore one of the most important issues in designing WiMAX networks is the choice of a scheduling algorithm. As a consequence, a number of scheduling algorithms have been proposed in the literature, most of which assume perfect channel conditions, no error losses and an unlimited power source [5]. The proposed schedulers would work well for wired networks. However, unlike wired networks, wireless links suffer from diverse impairments; they are subject to time-and location-dependent attenuation, multipath and noise that results in received message degradation. For the users on the move, there may be rapid changes in their environment. These factors together with transmission impairments due to bad channel and MAC delays greatly affect the performance of wireless traffic schedulers [5], [6], [7].

Consequently, it is difficult to predict the behavior of a scheduler in such an unpredictable wireless environment. It's crucial that the schedulers take into effect channel states when assigning resources. In this paper we present a modified scheduling algorithm for Mobile WiMAX named Channel Aware Deficit Fair Priority Queue (CA-DFPQ). It is a combination of a hierarchical scheduler proposed in [3]suitably modified with a channel compensation mechanism. Our objective is to provide bandwidth and delay guarantees to QoS sensitive applications while achieving high bandwidth utilization even in the face of poor channel environment.

2. IEEE 802 IEEE 802.16 QoS Architecture

802.16e Mobile WiMAX is an improvement over fixed WiMAX (IEEE 802.16d). It offers mobility with scalability in radio access and network architecture [1], [4]. Key features that differentiate 802.16e Mobile WiMAX from other wireless access technology include its use of (1) Orthogonal Frequency Division Multiple Access (OFDMA), (2) a scalable spectrum use which ranges from 1.25 MHz to 28 MHz, (3) advanced antenna combined with multiple input multiple output (MIMO) support, (4) adaptive modulation and coding schemes and (5) a variety of QoS service class support.

For 802.16, communication at the link layer is either time-division duplex (TDD) or frequency-division duplex (FDD). Mobile WiMAX supports TDD mode, where time is divided into frames [4] and each frame is further dynamically subdivided into an uplink (UL) sub-frame and a downlink (DL) sub-frame. TDD frame for fixed WiMAX mode is 0.5, 1 or 2 ms while for Mobile Mode it's typically fixed at 5 ms [1]. Each sub-frame is divided into subcarriers, and a number of available subcarriers are grouped together to form sub-channels. For OFDMA-PHY, the frame is partitioned both in time and frequency forming slots. The MAC layer allocates the time/frequency resources to subscriber stations (SSs) in units of slots. A slot is the smallest PHY layer resource that can be allocated to a single SS in the time/frequency domain (Fig 1).

The DL channel is a broadcast channel used by the Base station (BS) to transmit control information and data to subscriber stations (SSs), while the UL channel is time-shared among all the SSs. The DL sub-frame contains the DL-MAP and UL-MAP at the beginning of the frame. Schedule grants to SSs for each frame in the DL channel is carried by the DL-MAP while schedule request for the UL channel is carried by UL-MAP messages transmitted during one or more transmission bursts. This information must be read by each SS, since it informs each of the SS of allocated DL resources, ranging contention, registration etc. Using the UL-MAP, the SS will place data on its allotted data slots in UL sub-frame with the corresponding connection ID (CID). Each flow from and to the BS is uniquely identified by its own CID. If there is any data remaining, the SS can piggyback remaining queue request and queue information of each CID(s) in these data packet. The BS will use this information to generate slots allocation pattern for the next frame. The BS will then assign resources to the SS by granting a number of physical slots with specific burst profile. A burst profile is a combination of modulation, coding rate and FEC) [1]. For DL, the BS has all the queue information and assigned weights, and having done the slot allocation will broadcast the information to all SSs.

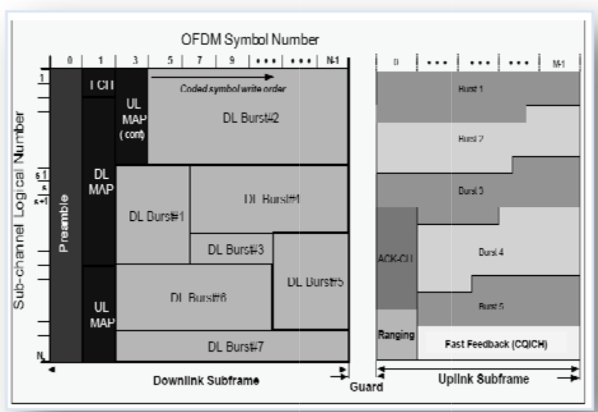


Fig 1: TDD frame structure

For 802.16 Mobile WiMAX, an SS can have a number of connections e.g. receiving a call and surfing the internet. SSs will notify the BS of its bandwidth request for each connection, but the BS grants UL bandwidth as a whole i.e. Grant Per Subscriber Station (GPSS). Hence the SS has to have its own scheduler which decides of how best to redistribute the granted resources to all of its active connections. Bandwidth requests can either be incremental or aggregated and done using unsolicited polling, unicast polls, broadcast/ multicast polling or piggy-backing. The SS will send to the BS the number of packets waiting on its connection and the sizes of the waiting packet when making bandwidth request.

2.1 Scheduling and QoS in IEEE 802.16

The principal mechanism for providing QoS in Mobile WiMAX is to associate packets traversing the MAC interface into a service flow identified by CID. A service flow is a unidirectional flow of packets that is defined by traffic behavior and specific QoS requirements [1]. Service flows are created during the process when an SS joins the network, or later on when an SS explicitly request for QoS attributes. Service flows are created, changed or deleted through the issue of Dynamic Service Addition (DSA) or Dynamic Service Change (DSC) or Dynamic Deletion (DSD) messages. The goal for the service flow is to allow the SS and BS to negotiate adequate QoS requirements. If granted, the network ensures such QoS characteristics for the life of the service flow.

Service flow uses a number of traffic attributes to describe the QoS parameters such as maximum sustained traffic rate, jitter, maximum delay etc required for a connection. These QoS parameters are grouped together to form a service class based on the need to accommodate the requirements of various applications as well as a mechanism to send bandwidth requests to the BS [1], [5]. These service classes are listed bellow;

2.2 IEEE 802.16e Service Class

Unsolicited grant services (UGS) is designed to support constant bit rate traffic (CBR) in order to support real time applications with strict delay requirements e.g VOIP with fixed packet size. These applications generate fixed size packets at periodic intervals. BS assigns sufficient UL and DL services to an SS that ensures minimum delay and jitter. Since the requested bandwidth is assured, the SS does not need to send periodic request for service; grants for UGS connections are therefore granted periodically without explicit requests from the SSs. QoS parameters used are maximum sustained rate, maximum latency and tolerated jitter.

Real time polling services (rtPS) is used to support real time Variable Bit Rate (VBR) applications that generate fixed size data packets such as MPEG Video, VOIP with silence suppression. These applications have specific bandwidth and delay requirements. Key QoS attributes similar to UGS is the minimum reserved traffic and maximum latency. Unlike UGS applications, however, the size of the arriving packets for rtPS is not fixed, and rtPS applications have to notify the BS of their current bandwidth requirements. QoS parameters used are minimum reserved traffic rate, maximum sustained rate, maximum latency and tolerated jitter.

Extended real-time polling service (ErtPS)- Extended real-time polling service (ErtPS) is designed to support applications with variable data rates that require guaranteed data rate. BS provides unsolicited unicast grants to the SS, saving the latency for continuous bandwidth requests. For ErtPS, and in contrast to UGS which has a fixed size bandwidth, it has a dynamic bandwidth allocation. No traffic is sent during the silent periods and BS polls SS during these silent periods to find out if they have ended. Thus in addition to static allocation during setup, ErtPS allows contention resolution [25]. QoS parameters used are maximum sustained rate, maximum latency and tolerated jitter.

Non-real time polling service (nrtPS); used for delay tolerant applications that do not have specific delay requirements e.g. file transfer. In this case, a certain bandwidth and delay time cannot be assured by the network. BS polls the SS to find out if bandwidth requirements are needed, and can reserve nrtPS connections having minimum amount of bandwidth or the BS grants unicast polls to nrtPS connections. In addition the BS can group several SS into a multicast UL group. The BS can then advise all the SS in the group that they can send their UL bandwidth request simultaneously instead of granting individual UL opportunities. Only minimum rate is guaranteed.

Best effort service (BE); similar to non-real-time Polling Service where the service is not ensured for bandwidth or delay. Bandwidth is granted only if there is any left over from other classes. SS uses contention and unicast request opportunities to send bandwidth request. The number of collisions, jitters etc depends on the number of SS in the cell area and the length of the contention area. The disadvantages of BE and nrtPS class of service is that a collision occurs whenever two or more stations accesses the medium in the same contention slot to send bandwidth request. To ensure minimum lost packets, that is, reduce collisions, collision avoidance scheduling schemes maybe used. Both BS and nrtPS UL connections request bandwidth by either responding to broadcast polls from BS or by piggybacking bandwidth requests on an outgoing PDU.

A number of functions are required to adequately support QoS for these service classes. Among these are a connection establishment where a connection admission control (CAC) is employed and packet scheduling for traffic prioritization. CAC algorithms are used to control packet entry during a connection establishment of new connections and reserve resources i.e. bandwidth and buffering. Packet scheduling is used to allocate resources during packet transfer. Buffering may also be required to ensure that packets without strict priority can be buffered, and provided for a means to discard packets whose QoS requirement cannot be guaranteed.

3.0 Related work

Among the scheduling schemes proposed in the literature are hierarchical and channel aware schemes. In hierarchical schemes, a two-layer approach is implemented [7]. With this approach, Deficit Fair Priority Queuing (DFPQ) is used to simultaneously schedule both UL and DL traffic in the first layer. At the lower level, a service class based priority is implemented so that as expected rtPS>nrtPS >BE. UGS is assigned fixed bandwidth allocation and thus taking precedence over all other service classes. To meet the strict delay requirements for rtPS, a variation of [7] is proposed in [8] where the DFPQ algorithm is modified to Pre-emptive DFPQ so as to give more bandwidth to rtPS flow.

Channel aware schedulers are designed to contend for the unpredictable wireless environment. The approach adopted for most of the proposed channel aware schedulers is to allocate more resources to SSs with better channel conditions at the expenses of SSs with poor channel conditions. In cases where the channel state conditions for such SSs falls below certain predetermined levels, no scheduling is done altogether. The reasoning is that for such channels, scheduled packets sent to or from the SS would be dropped anyway. A compensation scheme may also be introduced, so that SSs that had missed resource allocation will be compensated at a future time when their channel conditions improve [4], [8], [5]. In this case, channel compensation is used to swap channels between a flow that perceives a bad channel and a backlogged flow that is subject to a good channel. Additional channel access maybe granted to channels that were are bypassed once their channels quality becomes better.

Channel aware schedulers include Wireless Deficit Round Robin (WDRR) which is a modification of Deficit Round Robin (DRR) [5]. WDRR consists of an error free service model to provide service to error free channel, a leading and lag model that determines which sessions are leading, lagging or in sync, and a compensation model that compensate lagging session from leading sessions for additional services received when the lagging sessions had poor channels. All sessions in WDRR are examined serially and then allocated their quantum based on the examined channel state. Uniformly-Fair DRR (UF-DRR) [8] improves on WDRR. In UF-DRR the quantum of all leading sessions is summed up and distributed among all clean lagging sessions in proportion to their lag values. This way, the authors contend, each leading session gets an equal opportunity to relinquish its quantum and each lagging session gets equal opportunity to gain additional resources resulting in better short-term fairness.

In [4] proposes a channel aware scheduler specific to WiMAX using Worst-Case Fair Weighted Fair Queuing + (WF2Q+) algorithm enhanced with an error compensation technique. WF2Q+ provides good QoS guarantees and fairness to all service class flows, but at the expense of higher implementation complexity. This however, is not an issue; while we expect that such channel aware schedulers will provide delay and throughput bounds as well as fairness, they also introduce implementation complexity. The scheduler have to search serially for clean sessions from among N such sessions, and hence their implementation complexity will increase beyond O(N) [4].

4.0 Proposed QoS Architecture

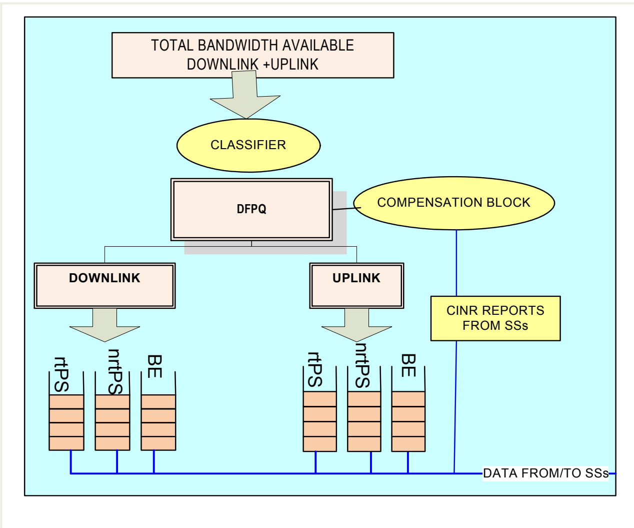
Fig. 2 shows the proposed reference channel aware DFPQ. The network model discussed consists of a cell-structured wireless network with the BS in every cell responsible for both the DL and UL communications. The wireless links between the SSs and the BS are subject to bursty errors but they are assumed to be independent. Therefore a flow for a wireless link channel can be in an error prone state-- in which a high proportion of transmitted packets are corrupted, and no transmission is possible, or in an error free state-- in which a high proportion transmitted packets will be received without being corrupted.

Our solution approach is composed of a (1) CAC admission control and packet classifier. This is used to limit the number of flows admitted into the network so that overflow and starvation for some services are preserved. (2) A hierarchical packet scheduler that is used to schedule flows based on directional differentiation (i.e. DL or UL traffic) and service class differentiation. (3) A compensation block that is able to sense the channel and aid in per flow compensation, and (4) buffering used in order to control buffer size and drop stale packets.

4.1 Admission control and Packet Classifier

We use an admission control mechanism to determine whether to admit new connections. Flows belonging to UGS and ertPS are not subject to scheduling since they receive a constant reserved bandwidth. The packet classifier will sort out the packets received at the BS and buffered into one of the per class queues according to class of service i.e. rtPS, nrtPS or BE.

Similar to [3], we shall use the minimum reserved traffic rate (r_{min}) for admission control and maximum sustained traffic rate (r_{max}) for scheduling. We use r_{min} to estimate the available bandwidth. For rtPS, both r_{min} and r_{max} are specified, while nrtPS service only r_{min} is specified. For BE service r_{min} and r_{max} are not specified. In this case since $r_{min} = r_{max} = 0$, then they can be accepted by the admission control, however, their QoS will not be guaranteed.



All the DSA requests from the SSs to the BS are summed up and compared to the estimated available bandwidth (BW_T). In this case, for the i^{th} class of services queue with j_i simultaneous connections, the available bandwidth BW_t is given by;

$$BW_t = BW_T - \sum_{i=1}^{I} \sum_{j=1}^{J-1} r\,min(i,j) \qquad (1)$$

and

$$BW_t \geq 0 \qquad (2)$$

Where $r_{min}(i,j)$ is the minimum reserved traffic rate for the j_{th} connection in the i_{th} class of service. BW_T is the total wireless link capacity. Service flow with r_{min} equal to zero i.e. BE traffic, can be admitted, but since they have low priority their QoS will not be guaranteed. Eq. (2) in this case will serve as the admission criteria.

For each connection, the admission control timestamps each arriving packet according to its arrival time. This information will also be exploited by the buffer manager to define when the time-out expires. In such a case, the packet should be discarded.

4.2 Packet scheduling

We focus on the hierarchical scheduling architecture proposed in [7], where a two-layer approach is implemented. We propose to use service class based and directional based priority using deficit fair priority queuing (DFPQ). DFPQ assigns a higher priority to DL to distribute total available bandwidth among DL and UL services. This is done in order to ensure that the BS is able to relays packets as soon as they are received thereby avoiding buffer overflow in the BS that would otherwise arise at the BS. DFPQ will then schedule service class flows in the active list in a strict priority rtPS(DL) > rtPS(UL) > nrtPS(DL) > nrtPS(UL) > BE(DL) > BE(UL). In each round highest priority queues are serviced first. In the second layer, round robin is used for BE, earliest deadline first (EDF) for rtPS and weight fair queue (WFQ) for nrtPS.

a. DFPQ

In DFPQ [7] the scheduler updates the active lists of queues, and then the variable DeficitCounter is derived from a value quantum. The value of the quantum is given such that

$$Quantum(i) = \sum_{j=0}^{J} rmax(i,j) \qquad (3)$$

Where J*i* is the total connections for the i_{th} service class flow. If r_{max} is not specified (e.g. $r_{max} = 0$ in DSA message), *rmax* will be set to r_{min}. since r_{max} and r_{min} are not defined for BE we vary the value for r_{max} arbitrarily. The scheduler will visit each non-empty queue in the active list, starting with the queue with the highest priority and determine the value of quantum. The priority of the each service class is defined bellow:

DL_rtPS	UL_rtPS	DL_nrtPS	UL_nrtPS	DL_BE	UL_BE
1	2	3	4	5	6

The DeficitCounter is initially set to the value quantum for each active queue, and then decremented by the amount of packets assigned each time a queue is visited. The process is repeated until the DeficitCounter for each queue is equal or less than zero, or there is no additional bandwidth request for that queue. The scheduler will then move to the next round. If there is no available slots on the frame i.e. $BW_t = 0$, the MAP message will be sent, and the scheduling for the current frame will end.

b. Channel Aware DFPQ

Factors such as transmission impairments due to bad channel or MAC delays greatly affect the performance of deadline-based schedulers. If an SS has poor channel quality (Fig. 3), assigning it slots will result in wastage of resources. Therefore this study will modify the DFPQ model by introducing a compensation block, as shown in fig. 2.

The compensation block makes the BS aware of the channel state. The BS gathers channel to interference and noise ratio (CINR) information whenever the SS requests bandwidth allocation for either rtPS, nrtPS, and BE traffic. The signal-to-noise ratio of the monitored channel compared to allowed signal-to-noise ratio and receiver sensitivity for each modulation and coding scheme defined by the standard for a given bit error rate (BER). The following section describes the operation of the compensation block.

The burst profile (modulation and coding rate adaptations) for each SS will be based on the channel-quality measurement i.e. received signal strength indicator (RSSI) and the carrier to interference and noise ratio (CINR) that the SS is required to provide to the BS on request. Each SS will monitor its own channel. It will then use this information to predict future channel state and send this information to the

BS using the channel quality feedback (CQI). Based on the CQI value, the BS can change the burst profile for the SS or change the power level of the associated transmissions.

We make the assumption that channel quality remains static for every frame, and that the BS is able to obtain perfect channel state information so that the CINR reports are correctly received from the SS to the BS. The compensation block will compare the CINR against the allowed values of SNR and receiver sensitivity for each burst profile specified by the standard for a BER. If the received power falls below the specified receiver sensitivity for the lowest defined transport mode i.e. the most robust modulation and coding rate (i.e. BPSK with half coding) the CID for this channel is considered bad and no packets will be transmitted.

The compensation block will

- mark a session CID as either good or banned based on CINR report
- swaps channel access between a flow that perceives a bad channel with a flow that perceives a good channel
- at a later time, when the bad channel perceives a better channel, additional channel access can be granted in-lieu of lost opportunity, while taking resources from the channel that was favored

If session i_j in service class queue i is banned as a result of bad channel, the scheduler will schedule the next packet of the same queue i to transmit, otherwise it will move on to the next queue $j+1$. The compensation block will use a counter method to mark leading flows with credit and lagging flows with debit. Thus a session i_j can be leading, lagging or in-sync and for all lagging or leading flows $\varphi(i)$

$$\sum_{i}^{N} \varphi(i) = 0$$

Where $\varphi(i)$ = credit/debit value for all session i for service class j. It follows that we can have the following three cases;

Case 1, CID session for SSi is in sync and its channel is banned

- CID session for SSi is in sync and its channel is bad, and there exists another clean lagging CID session flow for SS_{i+1} from the same service class queue with the highest credit, then CID session for packet SSi will relinquish φ(i) = Quantum (i) slots to CID session for SS_{i+1} . Also decrement the lag value for CID session for SS_{i+1} by φ(i) =Quantum(i)
- if there is no other clean channel for the service class or
- if there is no other packet in queue j then, the scheduler will move to the next priority queue $j+1$.

Case 2, CID session for SSi is in sync and its channel is good

- CID session for SSi is in sync and channel is good and there is no other leading flow, then schedule SSi
- there exist another clean leading flow, SS_{i+1}, then use part of its $\text{quantum}_{(i+1)}$ to schedule session for SS(i), debit SSi by the same amount and credit SS_{i+1}. SS_{i+1} should now be in sync
- there exists another good and lagging CID for session SS_{i+1} session flow then schedule SSi with is quanta and leave it in sync i.e. its lag counter is zero
- if there exists another lagging or leading flow that is banned, then it assigns its quanta to SSi, and schedules it. Also credit SS_{i+1} lag counter
- if there is no other packet in queue j, then the scheduler will move to the next priority queue $j+1$

Case 3, CID session for SSi is leading and its CID session is good or banned

- if the CID session for SSi is leading and its CID session is good, and there exist another clean lagging CID session flow for SS_{i+1} from the same queue with the highest credit, the CID session for packet SSi will relinquish φ(i) =Quantum (i) slots to CID session for SS_{i+1}. Also debit the lag value for CID session for SS_{i+1} and credit the lag value for SSi. CID session for SSi should now be in sync
- if there is no other packet in queue j then, the scheduler will move to the next priority queue $j+1$

The work of the compensation block is to swap channel access between a flow that perceives a bad channel with a flow that perceives a good channel. Later on, when the bad channel perceives a better channel, additional channel access can be granted in-lieu of lost opportunity, while taking resources from the channel that had been favored. Thereby, it follows that whenever a HOL packet is substituted, the substitute will be the one for the CID session with the highest debit counter. Having no substitute flow, a queue will be bypassed and removed from the active list. Of course we note that the complexity of DFPQ is raised from O(1) to O(N) since it has to search among all the sessions in a queue.

The proposed channel Aware DFPQ (CA-DFPQ) is depicted in fig. 2. Maximum slots per frame (BW_T) is the total available capacity of a TDD frame fixed at 2500 slots that can be used to schedule service flows. Consider two mobile stations ss1 and ss2 connected to a common base station. Each is able to send rtPS and BE packets that are classified to their respective service class queues. The quantum for rtPS and BE varies for each queue, but in this case is fixed at Q[rtPS]=1000 and Q[BE]=500 slots respectively.

First the scheduler updates the active list, and where a connection waiting queue is empty, it will remove it from the waiting list. The highest priority service class queue will be served first until it has a deficit, and then the scheduler will move to the next queue. In the first round, subscriber station ss1 with BW-REQ=400slots has a poor channel that is below acceptable receiver sensitivity, and therefore its channel is banned. Consequently, we shall substitute it by scheduling flows from ss2(500 slots) which is ready to be sent, and which otherwise would not have been scheduled *in this particular round.*

The compensation block will assign *ss1* with a lagging credit count $\varphi(ss1)$ equal to -400 i.e. the value of its flow that was not sent, and credit *ss2* with a leading debit $\varphi(ss2)$ equal to +500. Since the DeficitCounter is reduced to -100 (i.e. 1000-600-500=-100), the scheduler will move to the next service class queue. BW_t (unused frame slots)= BW_T -1100 =1400 slots remaining.

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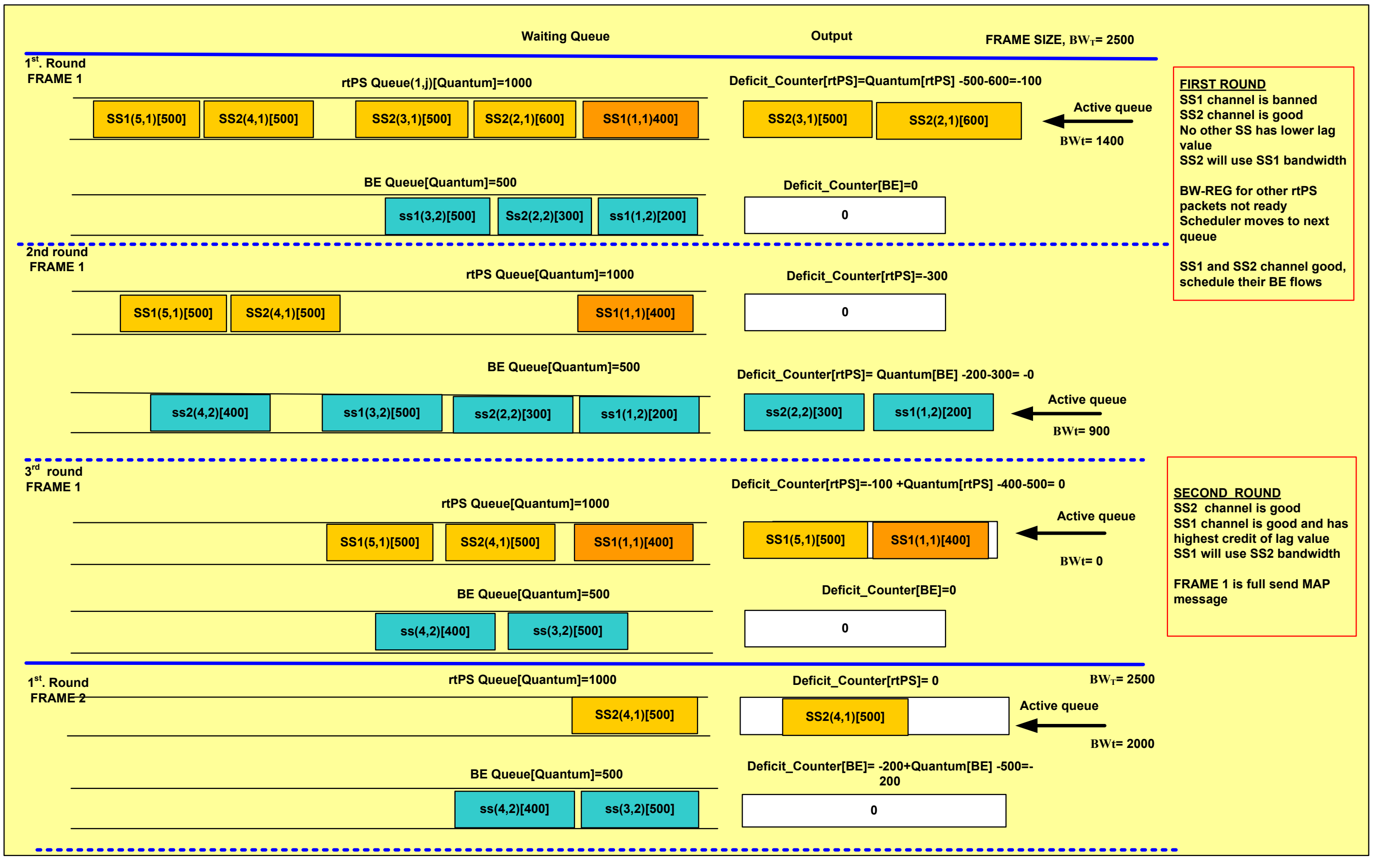

Fig. 3: Illustration of Channel Aware DFPQ

The next service class is composed of BE traffic. At this point, both ss1 and ss2 session channels are good. Both packets will be scheduled and the DeficitCounter for BE traffic service class will be reduced to zero i.e. Quantum(BE)-300-200 =0. Remain frames slots will be BW_t=1400-500=900, therefore the scheduler will move to the next queue which is rtPS.

At this point, the channel for ss1 is good, but it's lagging with $\varphi(ss1)$ =-400. In addition, there exists another flow ss2 which is leading with the highest credit $\varphi(ss2)$ =500. Since there is a another good session flow with a lagging credit, the scheduler will bypass flows for ss2 by scheduling two ss1 packets i.e. ss1(400) and ss1(500). The compensation block will update and mark ss1 as leading (i.e $\varphi(ss1)$= -400 +500 =100) while ss2 will be in sync i.e $\varphi(ss2)$ =+500-500 =0

Remaining frame slots BW_t =0, therefore there are no more slots to fill and we send the frame. The scheduler will then move to a new round with to start scheduling a fresh frame. It will move to the highest priority service class queue where the process will be repeated again.

5.0 Simulation Strategy and Future Work

We use TDD modulation where the frame is divided into DL and UL sub frame by a guard interval. The ratio of DL to UL can be varied and is typically set from 3:1 to 1:1 to support different traffic profiles. The choice for TDD is because it allows flexible sharing of bandwidth between uplink and downlink, and does not require paired spectrum. It also has a simpler transceiver design [2], [10]. However, the most important aspect in this case is that it has a reciprocal channel that can be exploited for spatial processing. The UL sub-frame has a channel-quality indicator channel (CQI) used by the SS to feed back channel-quality information that can be used by the base station (BS) scheduler to predict the channel quality.

Therefore this study will use a TDD channel bandwidth set at 5 MHz with a FFT size of 1024. Partially Used Sub-Carrier (PUSC) is used, in which the useful data subcarriers are 720 [11], 540 of which are allocated for DL and the rest are assigned to UL traffic. The frame length as suggested by [4] is fixed at 5ms. We use the International Telecommunication Union Telecommunication (ITU-T) recommendation Y.1541 for rtPS, nrtPS and BE traffic [12] that are mapped over Classes 1, 4 and 5 with maximum latency 400ms, 1s, and no limit respectively.

The classification and mapping are shown in table I. Total bandwidth BW_T = 20 Mpbs, and frame duration at 5ms so that the frame bandwidth will be 100Kbits.

For the channel model, we will use the second generation system suggested by the WiMAX working group [2] for scalable multi-cell architecture under NLOS conditions. The wireless channel is characterized by path loss resulting from shadowing, multipath delay, fading, Doppler spread, and co-channel interference etc. The median path loss (*PL* in dB) at a distance d_0 is given by,

$$PL = A + 10\gamma\log_{10}(d/d_0) + s \qquad \text{for } d > d_0,$$

$$\text{where } A = 20\log_{10}(4\pi d_0/\lambda)$$

λbeing the wavelength in m

γis the path-loss exponent and given by $\gamma = (a - bh_b + c / h_b)$ and depends on the given terrain type and the BS antenna height h_b

h_b is the height of the base station in m for h_b between 10 m and 80 m

d0 = 100m and a, b, c are constants dependent on the terrain category given in [4]

d is the distance between SS and BS antennas in meters, and

s is a log normally distributed factor that accounts for the shadow fading with a standard deviation value between 8.2 and 10.6 dB.

QoS Class	Applications (Examples)	Node Mechanisms	Network Techniques	Characteristics
0	Real-time, sensitive to jitter, interactive (VoIP)	Separate Queue Preferential Servicing, Traffic Grooming	Constrained Routing/Distance	mean delay <= 100 ms, delay variation <= 50 ms, loss ratio <= 10^{-3}
1	Real-time, sensitive to jitter, interactive (VoIP)		Less Constrained Routing/ Distance	mean delay <= 400 ms, delay variation <= 50 ms, loss ratio <= 10^{-3}
2	Transaction data, Interactive - Signaling	Separate Queue, Drop Priority	Constrained Routing/Distance	mean delay <= 400 ms, delay variation <= 50 ms, loss ratio <= 10^{-3}
3	Transaction Data, Interactive		Less Constrained Routing/ Distance	mean delay <= 400 ms, delay variation unspecified, loss ratio <= 10^{-3}
4	Low Loss (Bulk Data, Video Streaming)	Long Queue, Drop Priority	Any Route/Path	mean delay <= 1 sec, delay variation unspecified, loss ratio <= 10^{-3}
5	Traditional best effort applications	Separate Queue (Lowest Priority)	Any Route/Path	mean delay unspecified, delay variation unspecified, loss ratio unspecified

Table II: Y.1541 Guidance for IP QoS Classes and Mapping

The parameter figures chosen will depend on terrain, tree density, antenna height etc. The path loss value is calculated by the SS. The SS will then use this calculated value to estimate the channel quality and pass the same information to the BS using CQI. This simulation uses BPSK with ½ rate coding and fixes the receiver sensitivity RSS at -83.22dB as suggested in [1]. With the RSS at -83.22 dB, a BER lower than 10^{-6} can be achieved when using BPSK with half rate coding.

We compare the effect of a non-ideal channel i.e. the scheduler is unaware of channel conditions for each active flow. Since the scheduler is unaware of channel errors, it will schedule packets to active flows in poor channel conditions, thus wasting variable resources. We then introduce a channel aware scheduler which is able to schedule packets by making reference to channel quality.

6. Conclusion

This paper, proposes a channel aware DFPQ for QoS architecture. This study compares the effect of a non-ideal channel i.e. when the scheduler is unaware of channel conditions for each active flow to channel aware DFPQ i.e. one that schedules packets by making reference to channel quality. The proposed solution involves the addition of a compensation block and a buffer mechanism to support the QoS class of services specified by the IEEE 802.16 standard. The modified scheduler ensures that bandwidth resources are conserved by only allocating resources to good channels. The purpose is to provide directional based and service class based QoS guarantees even in the presence poor channel conditions.

This study has taken the practical step of proposing an efficient QoS architecture for IEEE MAC protocol. The practical issues such as performance and stability associated with the QoS architecture will further be investigated through simulations. In addition, even though the IEEE 802.16 standard specifies unsolicited grants for both UGS and ErtPS, scheduling these service classes in poor channel environments are a waste of resources. This study will also improve on the scheduler by modifying proposed architecture so that all service class flows can be scheduled.

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