Threshold based Cyclic Polling (TbCP): An Uplink Scheduling Algorithm for Mobile WiMAX Systems

D. M. Ali and K. Dimyati

***Abstract*—In this paper, we propose an uplink (UL) scheduling algorithm for Mobile WiMAX (IEEE 802.16e) system that satisfies the throughput and delay for the real and non real time application taking the adaptive modulation and coding scheme (MCS) into consideration. The proposed algorithm works by adjusting the threshold which is imposed on the nrtPS queue. The threshold value here represents the number of bandwidth request messages in the nrtPS queue. The algorithm then allocate the resources in two stages: The inter-class scheduling allocates the resources to different classes of service in accordance to the threshold based priority while the intra-class scheduling allocates the resources within the same class with the exhaustive service strategy. Finally, the simulation results validate the propose algorithm, and show that higher system throughput as well as lower delay and delay jitter can be achieved compared to other existing approaches.**

***Index Terms*—Mobile WiMAX, QoS, Throughput, Uplink Scheduling Algorithm.**

I. INTRODUCTION

WiMAX is a promising technology for a broadband wireless access which is based on wireless transmission methods defined by the standard IEEE 802.16. WiMAX is developed to substitute the broadband cable networks. It offers both fixed and mobile broadband wireless accesses, intended for the wireless metropolitan area networks. Theoretically, WiMAX can provide a broadband wireless access (BWA) up to 50 km for fixed stations and 5 - 15 km for mobile stations. It is an access technology that allows higher data rates over longer distances, more efficient use of bandwidth, avoids interference almost to a minimum with the issue of interference lessened. Initially, the standard allows the transmission in the range of 10 - 66 GHz. In 2004, the 802.16d was updated for lower frequencies in the range of 2 - 11 GHz, thus, resulting in less attenuation, an improved range and a better coverage within buildings. Then in 2005, the Mobile WiMAX 802.16e introduced the use of orthogonal frequency-division multiple access (OFDMA) as opposed to the orthogonal frequency-division multiplexing (OFDM) version with 256 subcarriers.

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The Mobile WiMAX quality of service (QoS) depends heavily on the 802.16e physical (PHY) and medium access control layers (MAC), as these layers deal with the base-station (BS) and mobile station (MS) access. Since the 802.16e standard works for different applications, fixed or mobile terminals respectively, there have been found to be some significant differences between them, where the technology is concerned.

The MAC layer of Mobile WiMAX provides a medium-independent interface to the PHY layer. It is designed to support and manage different kinds of traffic and applications through five defined service classes: Unsolicited Grant Service (UGS); Extended Real Time Polling Service (ertPS); Real Time Polling Service (rtPS); Non Real Time Polling Service (nrtPS) and Best Effort (BE). The standard does provide the QoS requirements for the service classes; however it leaves the performing algorithm undefined. In such a heterogeneous traffic, the UL scheduling algorithm serves as an important mechanism that affects the performance of the QoS. Thus, there are a number of scheduling algorithms that have been proposed to deal with the above mentioned considerations. Among these, Yi-Neng Lin et al [1] propose Highest Urgency First (HUF) which uses an urgency parameter of a maximum latency. For each service flow, bandwidth requests which deadlines are equal to one is serviced first and then to others until the minimum reserved rate is complemented. A combination of three algorithms; Priority Queuing (PQ), Weighted Round Robin (WRR) and Round Robin (RR) are evaluated in [2]. PQ is applied to the scheduling classes in which the UGS gets the highest priority followed by the rtPS, nrtPS and BE. The UGS is allocated with a fixed bandwidth; the rtPS and nrtPS are served through WRR, while the BE with the RR discipline. In [3], a Red-based Deficit Fair Priority Queuing (Red-based DFPQ) is proposed which dynamically sets the deficit counter of the rtPS service class based on the current queue length. Zeynep Yurdakul et al [4] have modified the algorithm in [3] to include the intra-class scheduling which uses the signal to noise ratio to calculate the weights of each MS. However, all above schemes do not consider the delay jitter of the UGS or ertPS as well as rtPS. Furthermore, delay jitter is the main QoS parameter for those service classes.

In this paper, we propose a threshold-based priority which is called as Threshold based Cyclic Polling (TbCP) algorithm to study the delay jitter of the real-time traffic applications (which are represented by UGS, ertPS and rtPS) as well as to satisfy the throughput and delay guarantee for various non real-time (which are represented by nrtPS and BE) and real time applications respectively.

The rest of the work is organized as follows: The next section summarises PHY and MAC layers of the IEEE 802.16e. The proposed algorithm with detailed explanation is presented in section 3 while the performance analysis is discussed in section 4. Finally, section 5 concludes this work as well as determines our future directions in this particular area of study.

II. PHY Layer and MAC Layer of the Mobile WiMAX Review Stage

Orthogonal Frequency Division Multiple Access (OFDMA), which is originally based on the Orthogonal Frequency Division Multiplexing (OFDM), has been specified as the accessing technique for the IEEE 802.16e. OFDMA subcarriers are divided into subset of subcarriers and each subset represents one subchannel. The number of subcarriers changes with the channel bandwidth in order to provide adaptive frequency bandwidth and data rate [5]. The IEEE 802.16e also supports Time Division Duplexing (TDD) as the duplexing technique. A TDD frame has fixed duration and contains one downlink (DL) and UL subframes. Fig. 1 illustrates the two-dimensional mapping between the frequency and time domains. A slot is the minimum unit of frequency-time resource with which a user may be granted, which contains 48 data sub-carriers and the amount of data carried varies with different modulations and coding schemes.

The MAC protocol is connection-oriented. Upon entering the network, each MS creates one or more connections over which its data are transmitted to and from the BS. The MAC layer schedules the usage of the air link resources and provides QoS differentiation for the different types of applications through the five defined service classes. The BS is responsible for coordinating the communication in the network since there is no direct communication between the MS in the Point-to-Multipoint mode. The transmission from the MS to BS is called UL whereas DL is from BS to MS.

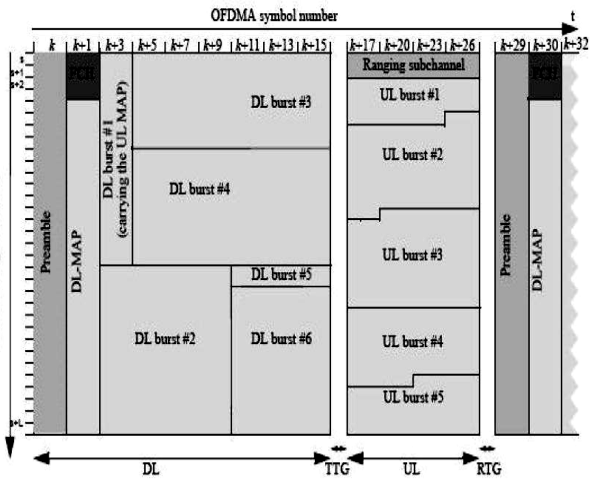


Fig. 1 The OFDMA frame structure [6]

A. System Model

The architecture of mobile WiMAX consists of a number of MSs and a BS. The BS manages the communication between the MSs in the network which take place in two directions, UL and DL. The MS that attempts to communicate with the BS will be identified with a unique connection identifier (CID). Due to the centralized architecture, the BS scheduler controls the system parameters. The transmission in DL is uncomplicated since the BS is the only one that transmits and schedules all the connections.

It should be noted that for an UL transmission, packets are queued at the MSs and specifically the UL scheduler works on a request-grant basis. For this purpose, each MS will send a bandwidth request message to the BS. Several bandwidth requests mechanisms are available such as unicast polls, broadcast and multicast polls, contention and piggybacking. Subsequently, after acquiring the bandwidth request messages, the messages are then classified according to the service class and QoS parameter in the scheduler for the scheduling process. An information element (IE) is created in the UL-MAP to indicate the control region and new resource assignments that MSs should transmit [7]. The UL-MAP is placed at the beginning of the DL subframe of each frame and the completed MAP is then broadcast to all MSs in the network. Each MS listens to the broadcasted MAP message for their CID and decodes the UL-MAP IE so that packets are transmitted in accordance to the allocated slots.

After the initialization process takes place, the MS will send a Dynamic Service Addition – Request message to demand for an UL request opportunity. The Dynamic Service Addition – Request contains the service flow (SF) parameter for the request connection and MAC address. The BS that receives the message will call the admission control to decide whether to accept or reject the request based on the available capacity. If there are several SF arriving simultaneously, the SF is handled according to the priority of each class. Then the available bandwidth is updated accordingly, following the acceptance of the SF. The available bandwidth can be calculated as:

$$C_{avail} = C_{total} - \sum_{i=0}^{I} \sum_{j=0}^{J_i} SF \tag{1}$$

where C_{total} represents the total bandwidth. I, represents the total number of SF or connections and J_i is the total number of service classes of the ith connection. Assume that each MS carries a single SF which eliminates the effect of packet scheduling at MSs. Each SF is fixed to maximum sustained traffic rate (MSTR) for UGS and ertPS. As for rtPS and nrtPS, it is set to a minimum reserved traffic rate (MRTR), and BE is given the available capacity after all the service classes are considered. A Dynamic Service Addition – Response will be sent from BS to inform the acceptance of the connection and a CID will then be assigned to the MS.

Five service classes are supported in the IEEE 802.16e. The UGS is appropriate for the applications that generate constant bit rate traffic hence it requires a fixed bandwidth at periodic intervals. The BS pre-allocates periodic slots to such MS prior to the QoS agreement (observing the MSTR) negotiated at the connection setup. The ertPS is formed through the combination of UGS and rtPS. It suits the real-time application with varying bit rates and delay requirements. In this case, the BS will provide unicast grants in an unsolicited manner to the MSs with the ertPS flows. Applications such as audio and video streaming are classified under the rtPS service class. These types of applications generate real-time packets stream which are variable in size. Due to that, the BS will provide unicast request opportunities in which the MS will stipulate the size of the grant required. The nrtPS is designed for a delayed tolerant variable size applications such as file transfer. The MS with such flow will be polled by the BS on a regular basis. Finally, BE is suitable for the application with no throughput or delay guarantees. MSs with the BE connections have to contend for their bandwidth allocation.

III. THRESHOLD BASED CYCLIC POLLING

In order to initiate a graceful trade-off between the real-time and non-real time traffic, a threshold based priority algorithm is presented. This is to ensure that the delay property of the real-time traffic is properly satisfied without sacrificing the throughput of the non-real time traffic. A priority based algorithm seems capable of guaranteeing the QoS parameters of the different service classes. However, the direct outcome of the priority concept is the starvation problem of the lower priority class. Thus, to mitigate with this issue, a threshold-based priority is proposed.

Our approach aims at dynamically adjusting the threshold value, n which represents the number of bandwidth request messages in the nrtPS queue. Furthermore, our proposed scheduling algorithm is simple and does not require computational complexity.

The scheduling scheme starts with the scheduler visits to the UGS queue. The UGS queue is served until no more grants are available (exhaustive service strategy). Once the queue is empty, the scheduler will move to the ertPS queue which is also attended until the queue is empty. The rtPS queue is served next and attended until the queue is empty. Before continuing the service to nrtPS, the scheduler will check on the amount of bandwidth requests available at the nrtPS queue. If the amount of bandwidth request message exceeds the threshold assigned, n, then the service will be carried out to nrtPS and subsequently the BE. On the other hand, the scheduler will return to serve UGS if the amount of the bandwidth request is less than the threshold assigned.

Immediately after each queue is served, the grants and bandwidth request messages are then sent to the staging buffer area for the mapping of IE to the UL-MAP process (to decide on the time slots that should be allocated to each MS subscriber). However, the mapping of IE to the UL-MAP, DL-MAP and data bursts to the OFDMA frame is not included in the scope of this work. The staging buffer is emptied as needed (if there are spaces in the UL-MAP) to fill up the empty slots in the UL-MAP. In order to achieve full frame utilization, a fragmentation process is supported. Fragmentation allows a grant and bandwidth request message to be partitioned in case there are some left-over spaces in the UL-MAP. The UL scheduler explained above is depicted in fig. 2 and 3, respectively.

The exhaustive service strategy which is employed to the second stage of the algorithm is to ensure that the queues are being allocated fairly despite their service classes. Each queue attended by the scheduler is served until the buffer is completely vacant and if the current queue to be serviced is empty, the scheduler will then move to the next available queue.

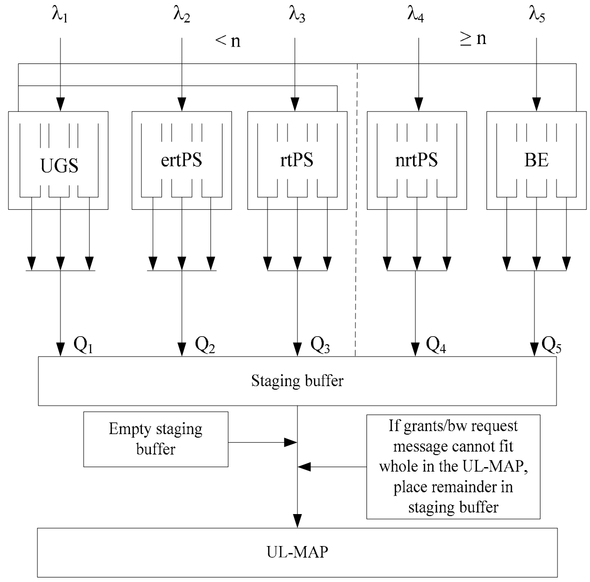


Fig. 2 The architecture of the TbCP algorithm

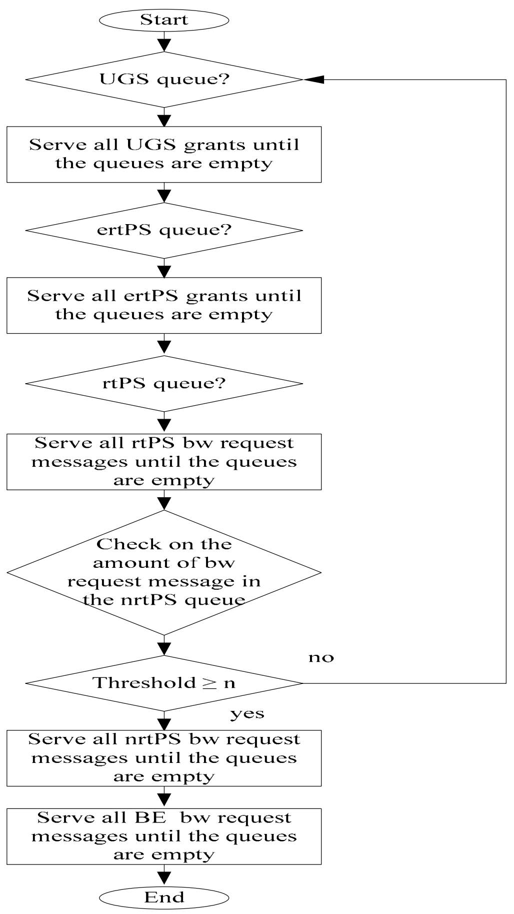


Fig. 3 The TbCP algorithm flowchart

IV. SIMULATION ENVIRONMENT AND PERFORMANCE ANALYSIS

The task of the Mobile WiMAX UL scheduler is a complex one. The scheduler must be able to support different types of application [8] as well as delivering the available bandwidth of different service classes [9] while satisfying the

QoS requirements. The UL scheduler has no information on the packets and the queue status information of each flow. Thus, most of the proposed algorithm assumed an estimate of the delay information which is less accurate and timely [10].

The IEEE 802.16e standard enables optimization of each MSs data rate by allowing the BS to set the MCS according to the channel condition [11]. Thus, to account for the adaptive MCS, each service flow is translated to the coding rate and bits per symbol carried for each modulation. Table 1 summarizes the coding rate and bits per symbol supported in the Opnet software. The simulation parameters and traffic parameters are summarised in Tables 2 and 3 respectively. Fragmentation is supported in order to achieve full frame utilization and allow packets to be partitioned in case there are some left-over spaces in the MAP.

TABLE 1 CODING RATE AND BITS PER SYMBOL FOR DIFFERENT MCS

MCS	QPSK		16-QAM		64-QAM		
Code rate	1/2	¾	1/2	3/4	1/2	2/3	3/4
Bits per symbol	2		4		6		

TABLE 2 TRAFFIC PARAMETERS

Application	Parameters
Voice	G711 Encoder $T_{on} = 0.352$ $T_{off} = 0.65$ Size = 10 ms Coding rate = 64 kbps
Video Conference [10,12]	Frame size: - Lognormal distribution - Average : 4.9 bytes - Standard deviation : 0.75 bytes Inter-arrival time - Normal distribution - Mean : 33 msec - Standard deviation : 10 msec
File Transfer Protocol (FTP)	Inter request time : - constant distribution - 30 seconds File size : - constant distribution - 10000 bytes
Web browsing (HTTP)	Page interarrival time: - exponential (exp) distribution (30 sec) Page properties: - objecsize:exp 1000 bytes - object per page: exp 4

TABLE 3 SIMULATION PARAMETERS

Parameters	Value
PHY Profile	OFDMA
Bandwidth	10 MHz
Base Frequency	2.5 GHz
TTG (Transmit-receive Transition Gap)	106 µs
RTG (Receive-transmit Transition Gap)	60 µs
OFDMA symbol duration	100.8 µs
Frame preamble	1 symbol
FFT size	1024
Frame duration	5 ms
Subframe ratio (DL/UL)	1:1
MSTR UGS	96000 b/s
MRTR rtPS	20000 b/s
MRTR nrtPS	80000 b/s
Polling time (rtPS)	2 msec
Polling time (nrtPS)	10 msec

A. *Performance Analysis*

The performance of the algorithms as explained in section 3 is evaluated using the Opnet simulator to show the correctness and the performance of the algorithm, respectively. To select the suitable value for the threshold, we have performed the simulation rigorously, varying the threshold. Through this, we observed the optimal value of the threshold to be equal to 10 (th10) and the smallest to be 1 (th1).

Fundamentally, the network operators in the real environment do not provide all the five service classes simultaneously (as define theoretically by the standard) to the end users so as not to burden the scheduler. Thus, to reduce the complexity [8], mainly two or the maximum of three QoS classes are assigned. The UGS and ertPS classes are designed for the VoIP traffic while the nrtPS and BE both represent the non-real time traffic, thus choosing either one to represent the traffic is considerably sufficient. Therefore, we have chosen the UGS, rtPS and nrtPS for scenario 1 while the rtPS, nrtPS and BE are selected for scenario 2. Through this, we highlight the importance of the threshold setting which represents the number of bandwidth request message in the nrtPS queue.

To guarantee the QoS of different service classes, a priority-based scheme is applied for the inter-class scheduling in a Mobile WiMAX scheduler. UGS is set to have the highest precedence followed by the ertPS, rtPS, nrtPS and BE. The direct negative effect of the priority-based algorithm is that, it may starve the connections of lower priority service classes (BE) which cause the throughput to be lower. Thus, to mitigate with this problem, DFPQ with a counter is introduced to maintain the maximum allowable bandwidth for each service class. However, determining the correct value of the counter is crucial and if not configured properly, the delay and throughput of the service classes traffic might suffer.

We compare the performance of the proposed algorithm with reference to [2,3] for the purpose of evalution and comparison. We have chosen the work of [2,3] with the condition that our proposed algorithm falls into the category of priority-based algorithms. Red-based DFPQ [3] is also a type of priority-based algorithm where the counter is set to the MSTR/MRTR value to maintain the maximum allowable bandwidth for each service class. As for the intra-class scheduling, the fixed bandwidth implementation, WRR [2] and RR (BE service class) [2] are involved. Fixed bandwidth implementation is specified in the IEEE 802.16e as the algorithm for the UGS and ertPS service classes.

1) Scenario1

In scenario 1, ten MSs are configured to web browsing using the nrtPS while an increasing number of MSs performing VoIP and video conferencing which are associated with the UGS and rtPS service classes respectively are adopted for the purpose of expanding the load. The VoIP server, video conferencing server and HTTP server are linked up to the BS via the core network as shown in figure 4.

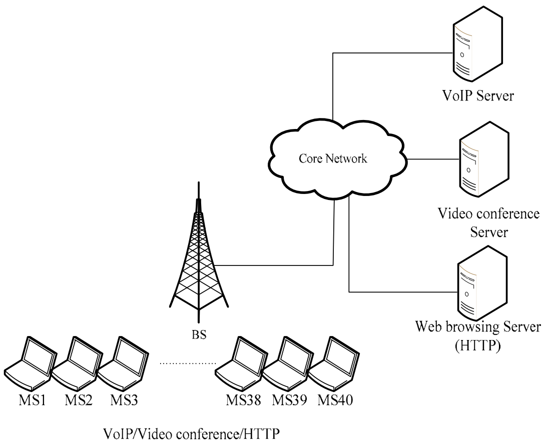
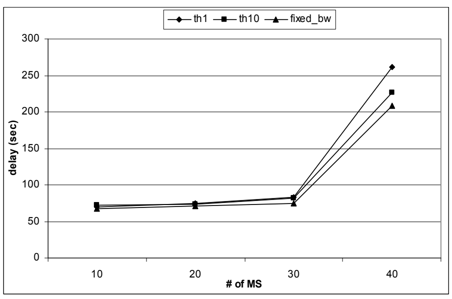
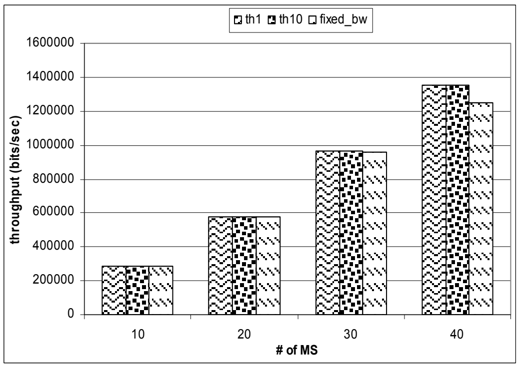


Fig. 4 Simulation topology for scenario 1

We observe that TbCP of threshold 1 and 10 has shown an increase in delay as compared to fixed bandwidth especially when MS approaches 40. This is due to the additional delay introduced in the scheduler which is the time that the grants have to wait (waiting time or queuing delay) to be serviced (as shown in Table 4). However, TbCP with threshold of 10 has reduced the mean waiting time as compared to threshold of 1. In UGS, the BS provides fixed-size grants (IE) in the UL-MAP to the MSs at periodic intervals, thus as the number of MS increases, higher amount of packets are transmitted which is then translated into higher system throughput. This can be further examines as in fig. 5 (b) where TbCP of threshold 1 and 10 outperforms the fixed bandwidth implementation when the number of MSs reaches 40.



(a)



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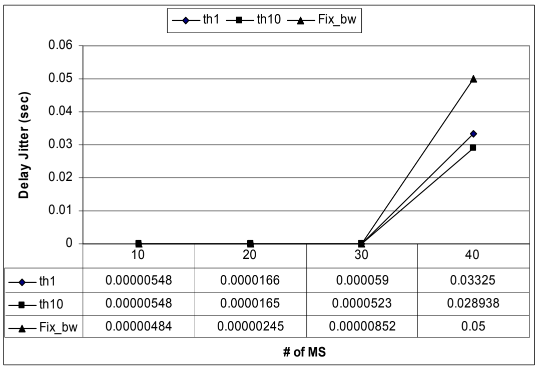


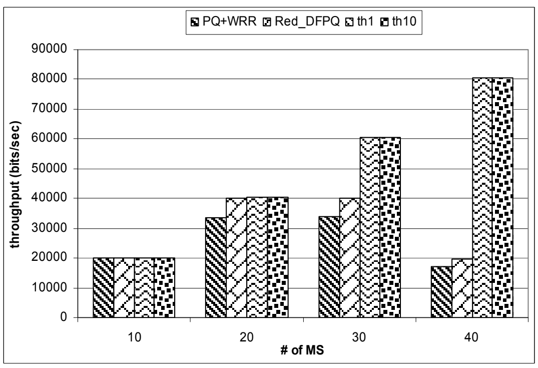
(c)

Figure 5 (a) UGS throughput (b) Average delay (c) Average delay jitter

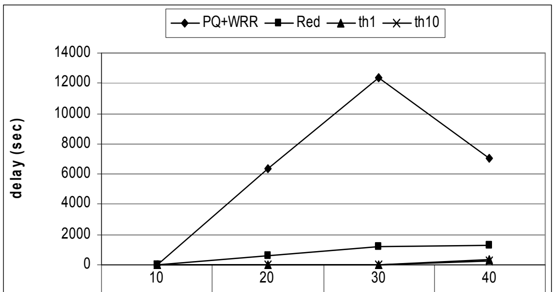
Fig. 5 (c) shows the average delay jitter for the proposed algorithm with the threshold of 1 and 10 which is compared against the fixed bandwidth implementation. Delay jitter is defined as the amount of variance in the arrival of packet to the destination. We observe that the average delay jitter value increases when the MS approaches 40. This is because when the queue that holds the UGS packets is full, most of the incoming packets will be discarded due to the overflow. As a result, the delay jitter is very small in a low-loading condition because most packets are immediately served without dropping and increases steeply when the queue is about to be full. The TbCP of threshold 10 of the proposed algorithm has reduced the delay jitter further.

TABLE 4 MEAN WAITING TIME (SEC) OF THE GRANTS IN THE UGS QUEUE

# of MS	th1	th10	fixed_bw
10	0.001907	0.001906	-
20	0.005241	0.005239	-
30	0.010257	0.010244	-
40	0.019586	0.019002	-



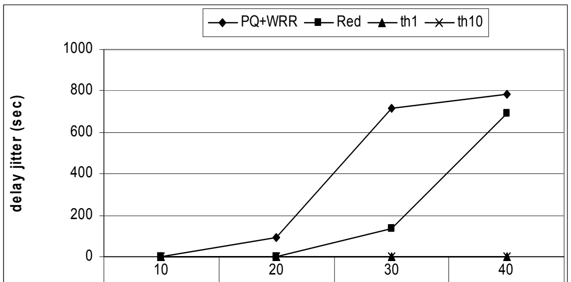
(a)



	10	20	30	40
PQ+WRR	9.06	6384.1067	12336.84	7084.713
Red	8.289	638.4655	1210.882	1283.562
th1	13.652	20.591	28.824	340.2183
th10	13.574	20.28	28.678	300.3536

# of MS

(b)



	10	20	30	40
PQ+WRR	0.0000982	90.627	715.4395	785.4935
Red	0.0000262	1.100653	138.405	693.1039
th1	0.0000705	0.000168	0.00038	0.169804
th10	0.0000577	0.000147	0.00037	0.163527

# of MS

(c)

Fig. 6 (a) rtPS throughput (b) Average delay (c) Average delay jitter

Fig. 6 (a) examines the rtPS throughput of the algorithms. In the WRR, each MS is allowed to be allocated with the amount of slots which is equal to the weight assigned for each connection. In this case, the weight is assigned and calculated as:

$$weight\ [\ j\ ] = \frac{MRTR\ (\ i, j\ )}{\sum_{j=0}^{J_i} MRTR\ (\ i, j\ )}\ x\ Total\ \ Capacity \tag{2}$$

where *weight [j]* is the weight of the j^{th} connection of the i^{th} service class and $\sum_{j=0}^{J_i} MRTR(i, j)$ is the total of the minimum reserved traffic rate of the i^{th} service class.

As for the Red-based DFPQ, the deficit counter of each service class is fixed to the MRTR. The rtPS is given higher transmission opportunities (the size of the deficit counter is adjusted adaptively based on the number of bandwidth request message of the rtPS service class) when the number of bandwidth request messages of the rtPS service class increases. Therefore, the Red-based DFPQ transmits more bandwidth request message which effectively increases the system throughput and decreases the delay of rtPS as shown in fig. 6 (a) as compared to PQ+WRR. The PQ+WRR algorithm starts to deliver the lowest throughput when the number of MS advances to 20 because the algorithm cannot perform well in the presence of variable size packets.

The exhaustive service strategy employed as the intra-class scheduling of the TbCP algorithm helps to further improve on the throughput of rtPS. This is because, the BS is only capable of estimating the UL traffic [9] through the bandwidth request messages sent from the MS and the MRTR assigned for each service class. The actual packets reside in the MS in which there is no mechanism that serves to transmit the information of the individual packet sizes to the BS. Thus, the use of MRTR in determining the values of weight and deficit counter thus affects the throughput, delay and delay jitter.

Fig. 6 (b) demonstrates the average delay in which the PQ+WRR displays the highest delay while TbCP produces the lowest delay. The TbCP with the threshold of 10 has reduced the delay as compared to threshold of 1.

Fig. 6(c) shows the performance of the delay jitter of the algorithms. The lowest delay jitter is observed from the TbCP with the threshold of 10. Jitter occurs when packets are sent and received with timing variations which can degrade the user's perceived quality of voice or video applications. To ensure a quality user experience, the WiMAX forum has specified the delay and delay jitter guideline for the VoIP and video conferencing application as <160 msec and < 50 msec.

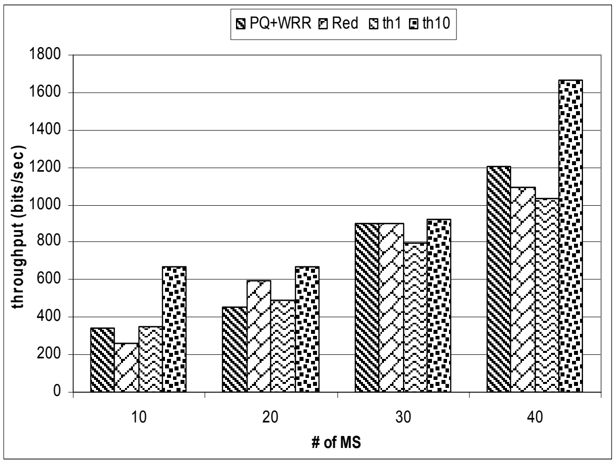


Fig. 7 nrtPS throughput

The throughput of nrtPS is presented in fig. 7. The TbCP of threshold 10 delivers the highest throughput among the algorithms. When the threshold value is set to 1, the scheduler will service the nrtPS queue after serving the rtPS queue, that is when there is ≥ 1 bandwith request message available in the buffer. At this point in time, the number of bandwidth request messages that is serviced is basically less than the number of bandwidth request message when the threshold value is set to 10 because the non-real time traffic load is configured to be lower than the real-time traffic load and the MS with nrtPS connections is polled every 10 msec which is much later than the MS with rtPS connections thus, resulting in smaller number of incoming bandwidth request messages directed to the nrtPS queue.

2) Scenario 2

In scenario 2 eight MSs are configured to the FTP and HTTP server using the nrtPS and BE service classes respectively while an increasing number of MSs performing video conferencing using rtPS are employed to enlarge the load. The video conferencing server, FTP server and HTTP server are linked up to the BS via the core network. The network topology is illustrated as in fig. 8.

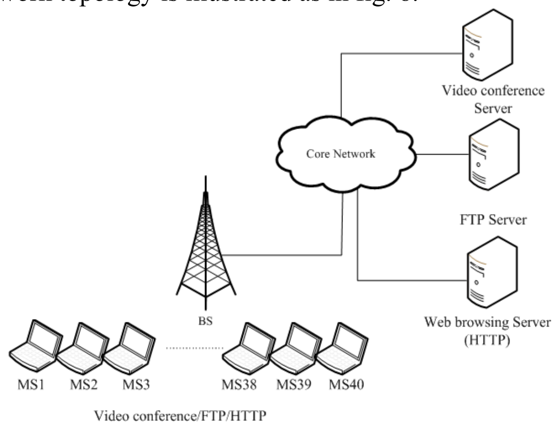
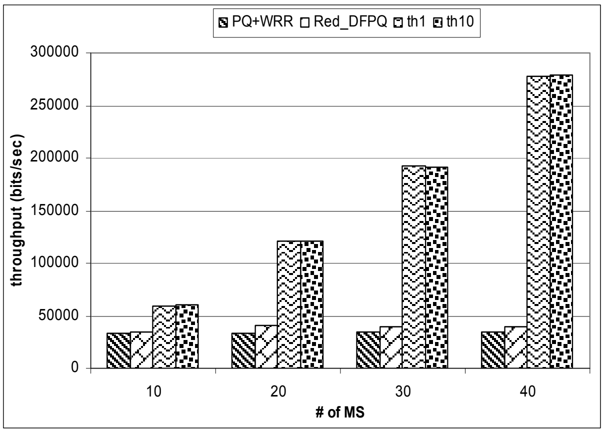
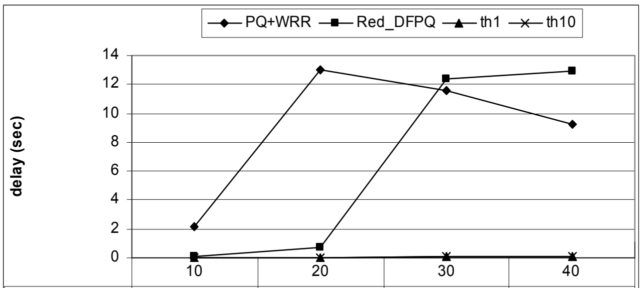


Fig. 8 Simulation topology for scenario 2



(a)



	10	20	30	40
PQ+WRR	2.137314	13.00195	11.61668	9.217369
Red_DFPQ	0.08933	0.688149	12.34865	12.947
th1	0.018863	0.031872	0.046961	0.065572
th10	0.018830	0.031804	0.046816	0.065435

# of MS

(b)

Fig. 9 (a) rtPS throughput (b) Average delay

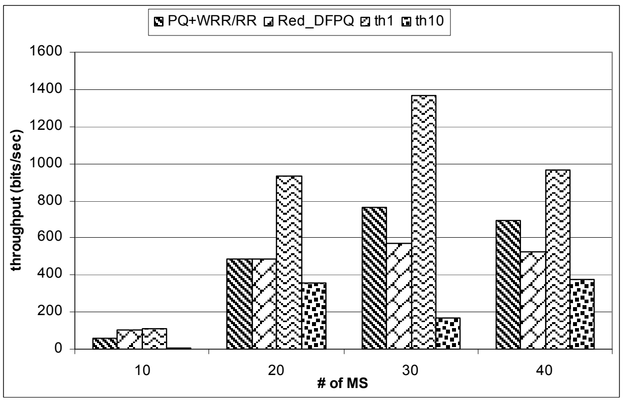


Fig. 10 BE throughput

Fig. 9 (a) shows the throughput of the rtPS with the TbCP of threshold 1 and 10 deliver the highest for the QoS class. In spite of the fact that the delay as in fig. 9 (b) of the rtPS of threshold 1 are not the lowest, it is still rendered low and bounded as compared to PQ+WRR and Red DFPQ algorithms.

Fig. 10 compares the throughput for the BE service class. The comparison shows that our proposed algorithm with the threshold of 1 delivers the highest throughput for the BE. However, for the threshold of 10, the scheduler needs to satisfy the higher priority service class as long as the bandwidth request messages are less than the threshold assigned, causing the throughput to be the lowest.

The Red-based DFPQ delivers a lower average throughput as compared to PQ+RR when the MS approaches 30 because the rtPS is allowed higher transmission opportunities when the number of bandwidth request messages of rtPS increases thus, reduces the throughput of BE. The combination of the PQ and RR algorithm reduces the throughput of the BE class because the scheduler needs to satisfy the higher priority class before the BE is served. Furthermore, the RR scheduling algorithm would allow only one bandwidth request message to be served in each round.

V. Conclusions

In this paper, we have proposed a threshold based cyclic polling algorithm for the ULscheduler of IEEE 802.16e. The algorithm supports diverse QoS as defined by the standard. In order to meet the delay property of the real-time traffic, we have manipulated the threshold value set upon the nrtPS bandwidth request message to take advantage of the delay-tolerant feature demonstrated by this service class. We have discovered the importance of the threshold value and its undeniable influence towards the configuration of the service classes. Results from simulations show that when the network is configured with a higher concentration of real-time QoS classes (UGS,ertPS and rtPS), then the threshold value should be set higher which is 10. This can be demonstrated from the results of scenario 1 with low bounded delay, lowest delay jitter and higher throughput delivered from the UGS; lowest delay and delay jitter of the rtPS and highest throughput achieved of the nrtPS. On the other hand, results from scenario 2 of the simulations prove that the threshold of 1 is the most appropriate value (when the network is configured with higher concentration of non-real time service classes). This is confirmed through the results of higher rtPS throughput, low bounded delay and delay jitter and highest throughput delivered by the BE.

Scheduling algorithm which is employed to the DL scheduler cannot be applied to UL scheduler directly. This is because the BS schedules the bandwidth requests arriving from the various MS and has no information about the packet. The BS estimates the UL traffic according to the MSTR and MRTR sent by the service flow of the MS.

As for our future direction, we recommend an admission control scheme that adaptively admits the connection, by taking into account the threshold setting and the QoS parameters. Furthermore, the threshold value should be self-configuring, implying that the adjustment of the value should be done intelligently based on the configuration of the network instead of being adjusted manually.

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