A Performance Study of Downlink Scheduling Algorithms in Wireless Broadband Networks

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Abstract**—**IEEE 802.16 is also known as WiMAX was developed to produce high performance in Broadband Wireless Access (BWA) systems with a lower deployment cost than wired broadband services. Like other broadband services, IEEE 802.16 is designed to support applications such as Voice over IP (VoIP), video streaming, video conferencing and online gaming. In IEEE 802.16 standard, five types of service classes have been formed to cater the Quality of Service (QoS) needs for different applications. However, the standard does not state any specific scheduling algorithms for either uplink or downlink transmission. Therefore, scheduling algorithm implementations are depending on the vendors, service providers and researchers. In our presented work, an analysis of various available scheduling algorithms in wireless environment has been carried out. Upon the literature study and analysis, Round Robin (RR), Strict Priority (SP), Self-Clock Fair (SC) and Weighted Fair Queuing (WFQ) were tested in downlink scheduling. For each scheduling algorithm, two scenarios were created, 1) 64QAM and 2) the combination of 16QAM & 64QAM. Simulation results indicate that all the schedulers were struggling to perform as the number of Subscriber Station (SS) increases. Furthermore, the impact on having bad quality channel, which is 16QAM, is also one of the reasons that produces poor performance among all the schedulers. Hence, the traditional schedulers are not suitable for the uncertainty condition in wireless environment because they do not satisfy the QoS demand in WiMAX.

Index Terms**—**quality of service, scheduling, IEEE 802.16, WiMAX, wireless broadband network, LTE

I. Introduction

Worldwide Interoperability for Microwave Access (WIMAX) is designed for telecommunication to offer internet access. WIMAX is based on IEEE802.16 standard, which is Broadband Wireless Access (BWA) with the aim to provide broadband connectivity. The range of the wireless coverage for a city area is about few kilometers. The IEEE 802.16 idea is to attain simple deployment with high-speed data rate. A WIMAX base station (BS) can offer up to 50km in range and utmost data rate of 70 Mbps, in contrast to 802.11 with only 54 Mbps for a range of 300 meters. WIMAX offers Quality of Service (QoS) that supports five different categories of services; 1) Unsolicited Grant Service (UGS), 2) Real-Time Polling Service (rtPS), 3) Non-Real-Time Polling Service (nrtPS), 4) Best Effort (BE) and 5) Extended Real-Time Polling Service (ertPS).

UGS service class is designed to handle unsuppressed Voice over Internet Protocol (VOIP) and rtPS service class is for real-time multimedia application. Meanwhile, large non-real-time data is mapped to nrtPS traffic. BE is for general data transmission such as web surfing which does not require guarantee transmission rate. Besides that, ertPS supports real-time service flows that generate variable-size data packet.

Offering high throughput with smallest amount lost in packet by a capable scheduling algorithm is certainly a difficult task for system developers. The main problem found in this research is the difficulty in the allocation of bandwidth based on the QoS service class to satisfy the connections. There is still a clear absence of performance studies that offers an integrated platform for different algorithms. From the review of several algorithms, the positive and negative aspects of each algorithm are detected in this study. Following are some of the problems face towards finding the best scheduling algorithms in satisfies QoS guarantee in WiMAX:

- Low fairness among all queues.
- Uncertainty of wireless channel.

The main objective of this research is to study and compare existing traditional scheduling algorithms used in wireless network. Strength and weaknesses of the algorithms are identified. For evaluation, the performance metrics, such as throughput, delay and jitter are chosen with respect to the characteristic of WiMAX as specified in the IEEE 802.16 standard.

This paper is organized as follows. Section II presents the QoS Framework of IEEE 802.16 and Section III describes scheduling mechanisms for IEEE 802.16. Section IV depicts the simulation experiment environment and parameters, while the simulation results and analysis is discussed in Section V. Lastly, conclusions and future plans are presented in Section VI.

II. IEEE 802.16 QoS Framework

IEEE 802.16 protocol architecture is divided into two layers, which are MAC and PHY layer. The MAC layer is a common interface that forms the foundation of the protocol and it interprets data between physical and upper data link layer. Basically, MAC layer is connection oriented. When a Subscriber Station (SS) enters into a

Manuscript received August 8, 2013; revised October 13, 2013.
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doi:10.12720/jcm.9.1.39-47

network, the SS needs to establish a connection with a BS in order to be served. In addition, the MAC layer provides services to SSs, such as addressing and channel access protocol mechanism.

A. PHY Layer

PHY layer handles multiple specifications, frequency range and hardware recourse management. PHY layer also handles the interaction between BS and SS involves several protocols addressed in between layers of WiMAX architecture. Although WiMAX physical layer supports TDD and FDD duplexing techniques, only TDD been used in mobile WiMAX as in IEEE 802.16e standard. The frequency supported for RF bands is 10-66 GHz for Light of Sight (LOS) and below 11 GHz for none Light of Sight (NLOS). WiMAX PHY layer supports different multi-user systems such as Wireless MAN-OFDM, Wireless MAN-OFDMA and Wireless MAN-SC (Single Carrier).

Downlink (DL) is referring to the channel access from BS to SS, and Uplink (UL) is from SS to BS. Two duplexing techniques; Time Division Duplex (TDD) and Frequency Division Duplex (FDD) are used to establish communication between DL and UL. In TDD mode, DL and UL shared the same frequency channel when transmitting signal. For FDD mode, DL and UL are in separate two frequency channels. However, TDD is most preferably used in WiMAX due to efficiency in managing radio resources. Fig. 1 shows the frame format for TDD downlink and uplink frame in a mobile WiMAX network.

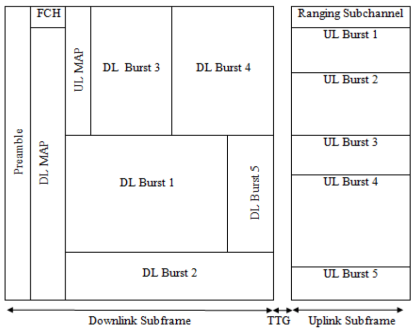

Fig. 1. IEEE 802.16 frame structure

In general, BS starts transmitting broadcast channel to all SSs by sending downlink subframe followed by a short gap of Transmit Transition Gap (TTG). TTG is used to avoid collision between BS and SS. The SS then sends UL subframe to the BS. In additional, preamble in Frame Control Header (FCH) is used for synchronization and channel estimation. Besides that, DL-MAP and UL-MAP are types of control messages in MAC PDU which responsible to inform SS about the allocation of timeslots for data transmission. The message structure that consists of DL-MAP, UL-MAP, DCD and UDC is as depicted in Fig. 2. Meanwhile, Downlink Channel Descriptor (DCD) and Uplink Channel Descriptor (UCD) provide channel and burst profile. When a SS received the first burst, it checks the Connection Identifier (CID) in DL-MAP to know its receiving timeslot. The SS does check UL-MAP which defines the timeslot for uplink channel access [1]. Upon the BS received the uplink frame, the frame will be given Service Flow Identifier (SFID), information of Signal-to-Noise Ratio (SNR), arrival time and size of the packet.

DL-MAP	UL-MAP	DCD	UCD

Fig. 2. Message structure of DL-MAP, UL-MAP, DCD and UDC in a MAC PDU

B. MAC Layer

The IEEE 802.16 MAC layer is divided into three sublayers; Convergence Sublayer (CS), MAC Common Part Sublayer (CPS) and Security Sublayer. The CS provides transformation and mapping for external packet that received through CS Service Access Point (SAP) into MAC (Service Data Unit) SDU. In CS, the basic QoS is carried out by mapping and classifying Protocol Data Unit (PDU) into CID [2]. The three types of specification which are Asynchronous Transfer Mode (ATM) CS, packet CS and generic packet CS, as specified in [2]. In transmitter, the CS is responsible in converting network layer packets into the MAC SDUs, and from MAC SDUs to network layer at the receiver. Furthermore, the main function of CS is to receive PDU from higher layer and classifies the PDU into appropriate connections before process and delivery. Classification is a process where MAC SDU is mapped to particular transport connection of MAC service flow and CID. Once the classification done, MAC CPS sends the PDUs to MAC CPS functions for QoS, fragmentation, packing and etc.

For second sublayer, MAC CPS is the core function in MAC layer, which includes bandwidth allocation, QoS management, connection establishment (SS initialization and registration), service flow management and connection maintenance. Another importance feature of MAC CPS is the Service Flow (SF) or MAC Transport service. IEEE 802.16 MAC standard [2] defines two types of connections; management and data transport connections. The structure of MAC management message is illustrated in Fig. 3.

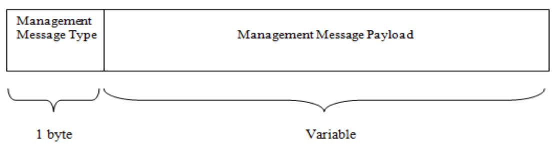

Fig. 3. Structure of MAC Management Message

Examples of some MAC management messages are DL-MAP, UL-MAP, DCD, UCD, RNG-REQ and RNG-RSP. These entire messages only carry control information and transmitted on management connections by multicasting and broadcasting. MAC management message is usually being divided into three messages, 1) basic, 2) primary and 3) secondary management message.

The communications between BS and SS take place in transport connection. After the initialization of SS with a 16-bit CID, service flows will be associated with a transport connection. At the SS initialization, there are two pairs of management connections, basic connections and primary connections between BS and SS [2]. Basic connections are used to exchange short and time-urgent message and primary connections are for longer delay control message. Secondary management is optional in WIMAX which generated by BS and SS to transfer delay tolerant message such as DHCP, TCP and SNMP.

WiMAX guarantees QoS at MAC level for application such as video streaming, video conferencing, voice over IP (VOIP) and other Internet services. In WiMAX, real-time services and non real-time services are differentiated based on the type of request. The main task of QoS is to ensure transmission ordering and scheduling mechanism between nodes in the air interface [3]. The performance level is measured in terms of throughput, packet loss, delay and jitter. The QoS requirements are also varied; they are depending on the application and service type. Furthermore, QoS requirement maintenance is very challenging due to uncertainty in wireless channel condition. There are five kinds of service class mechanisms suggested in [2] to support different types of applications, which are UGS, rtPS, nrtPS, ertPS and BE. UGS is designed to meet the minimum requirement of Constant Bit Rate (CBR) services for real-time application such as VOIP and T1/E1 emulation. This type of service needs a guarantee on bandwidth/throughput, latency and jitter [4]. Hence, UGS requires fixed bandwidth allocation or fix-sized packets at periodic intervals to service flow and no bandwidth request is needed [5]. rtPS is designed for variable size packets on periodic basis, such as VoIP with silence suppression or video streaming (MPEG video). BS provides unicast polling opportunities for SS to request bandwidth for rtPS connetions. Similar to rtPS, nrtPS supports non-real-time service flow that requires variable size data. It uses contention-based polling in uplink to request bandwidth on regular basis. For ertPS, BS offers same amount of bandwidth to SS unless explicitly requested by SS. Finally, BE is designed for non-real-time service where it has low requirement of speed and delay jitter.

In the third sublayer, security sublayer protects the network from been intruded and unauthorized access. The security sublayer provides authentication, security-key exchange, encryption and integrity control to WiMAX system. Encrypting connections between the SS and the BS is made with a data encryption protocol applied for both ways. An encapsulation protocol is used for encrypting data packets across the BWA. The rules for applying those algorithms to an MAC PDU payload are also given. With some additions security such as new encryption algorithms, mutual authentication between the SS and the BS, support for a handover and a new integrity control algorithm.

C. Bandwidth Request and Grant

Bandwidth request and grant are important especially in UL [6]. In DL, there is no bandwidth request and grant process involved and BS will schedule the MAC PDUs based on their local QoS requirements only. While in UL, it needs the involvement of SSs to request bandwidth from the BS. There are several ways of implementation in bandwidth request and grant for UL.

In DL, a BS has complete information about the SS on the status queues and due to this reason, it makes the decision making process at the scheduler much simpler. For UL scheduling, the BS does not know the status of UL queue which is resided at the SS. Hence, the BS sends unicast request to get the bandwidth request from the SS. Once the bandwidth request reply has been acknowledged by the BS, it translates the QoS requirement made by the SS to determine number of needed slots to be allocated. Once the BS has made its decision, the scheduling method will be announced in UL-MAP and DL-MAP at the beginning of a frame. Information in UL-MAP and DL-MAP indicates the slot that has to give to every SS for UL and DL.

There are two ways, where BS grants the bandwidth to a SS; 1) grant per connection (GPC) and 2) grant per subscriber station (GPSS) [7]. In the earlier method, the bandwidth is granted explicitly for a particular connection, while in the latter method, bandwidth is granted to SS as a whole bundle and it is not to an individual connection. The SS requires additional scheduling algorithm to manage the bandwidth between different types of service flow itself in the latter method. The most effective and efficient way is the GPSS approach due to its lower overhead compared to GPC which generate higher overhead when a lot of connection for a SS [6].

D. Adaptive Modulation Coding (AMC)

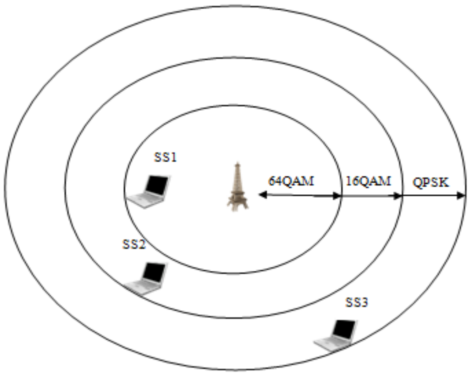

Fig. 4. MCS range in PMP environment

Adaptive Modulation Coding (AMC) is an alternative link adaptation used to adapt variability of radio channels. Therefore, data networks take this advantage to improve the overall network throughput. AMC is used in compensation-based or opportunistic approach by the BS scheduler to determine the channel condition during transmission. Compensation-based is a method where any missed transmission experience by any single flow will be retransmitted again in a latter time. For opportunistic approach, it takes into account of the gaining advantage in multi-user diversity at any given time. The AMC

Channel Quality Indicators (CQI) for all SSs are collected for analysis. Fig. 4 illustrates the range of Modulation and Coding Scheme (MCS) in PMP environment. From Fig. 4, SS1 will receive greater amount of bandwidth with higher code rates. MCS starts to decrease when the SSs are far from the BS. This is a typical way an AMC system works. The only challenge in AMC is to dynamically choose suitable MCS that meets target FEC block error rate.

III. Scheduling Mechnisms in IEEE 802.16

In wireless environment, it is difficult to maintain the variability and changes as compared to wired network. This problem is related to the QoS guaranteed in wireless network. Therefore, scheduling is needed to have the efficiency and fairness in meeting the QoS WiMAX requirement, as discusses in [8]. As scheduling mechanism is the key factor in distributing radio resources flow, the performance of scheduling method is highly depending on the BS equipment vendors. This gives room to everyone in developing new idea and delivers better services.

Scheduling can be either UL or DL. In this research, the focus is UL scheduling algorithm at the SSs. The scheduling is always been categorized into traditional scheduling algorithm and hybrid scheduling algorithm. In traditional scheduling, only one algorithm will be used to serve for all service classes. While for hybrid algorithms, a combination of two or more algorithm are implemented to meet the QoS requirements.

The needs for scheduling are vital towards solving optimization problem in relation to application QoS constraints. In addition, scheduling also helps to avoid traffic backlog and deadline. The simplest scheduling is Round Robin (RR). It is an alternative to First Come First Serve (FCFS) queuing. It equally serves slots to all queues, servicing a single packet from each, until each queue with packets has been serviced once. Once the process finished, the cycle repeats again until all the packets from queue have been transmitted [9].

Weighted Round Robin (WRR) is another alternative solution. Similar to RR, it serves all the queues rotationally. Unlike RR, WRR assigns a weight for each queue. Each queue will be given a weight and the number of weight is depends on number of packets transmitted from a queue. By having this weight differentiation in queue, prioritization will take place among the SSs [10]. In [11], the authors executed WRR at the beginning of each frame in BS. Allocation of bandwidth has been determined among SSs based on their weights. The weights depends on the QoS requirement priority. Higher weight assigned to SSs in rtPS classes compared to weight assigned to SSs in nrtPS and BE classes. In [12], the WRR mechanism being representing in pseudo-code as below:

```
for each connection c
  c.normalized_weight = c.weight/c.mean_packet_size
  min = findSmallestNormalizedWeight
for each connection c
  c.packets_to_be_served = c.normalized_weight / min
  //main loop
loop
for each non-empty connection c
  min(c.packets_to_be_served, c.packets_waiting).times
do
  servePacket c.getPacket
```

Weighted Fair Queue (WFQ) allows different scheduling priorities to statically multiplex data flows [13]. It also automatically sorts the traffic priority among individual traffic streams without requiring an access list. If N data flows currently are active with weight W_1, $W_2,\ldots\ldots,W_N$ data flow number ith will achieve an average data rate and calculated based from the (1).

$$R = \frac{w_i}{\sum w} \quad (1)$$

Random Early Detect (RED) is designed to achieve real-time QoS mechanism. RED is an active queue management which randomly drops packets when the average queue size exceeds the minimum threshold [14]. However, it is not practical to drop the packets frequently especially when the average queue size getting bigger. In order to minimize this, it needs to have a constant tuning on the RED parameters but it is hard to do. Although RED shows better performance than its predecessor Tail Drop (TD), its performance is highly sensitive to parameter settings.

In [15], the research measures the performance between TD and RED algorithm in high speed packet switched networks. The objective of the research is to detect higher rate of congestion by averaging the queue size. Results from the simulations shows that TD is having delay of two times more than RED. The average queue size formula can be calculated as below.

$$avg = \begin{Bmatrix} (1 - w_q)avg + w_q q, & if\ q > 0 \\ (1 - w_q)^m avg, & otherwise \end{Bmatrix} \quad (2)$$

where q is the current queue size, w_q is the weight given to the current queue size, m is the number estimated by idle time of the router. The dropping probability in RED is formulated as:

$$P_x = \begin{Bmatrix} 0, & \text{if avg} = T_{thr}\ (\text{min}) \\ P_{max}, & \text{if avg} = T_{thr}\ (\text{max}) \end{Bmatrix} \quad (3)$$

where P_x is the temporal probability which is varies from 0 to $P_{max.}$

Furthermore, the average queue size can be mapped into the corresponding probability, P_x (avg) as follows:

$$P_x(avg) = \frac{P_{max} \quad (avg - T_{thr}(\text{min}))}{(T_{thr}(\text{max}) - T_{thr}(\text{min}))} \quad (4)$$

The dropping probability can be calculated as below:

$$P_y = \frac{P_x}{(1 - c\,P_b)} \quad (7)$$

For expected number of packets which is discarded, it is calculated as in (8).

$$N\,(d) = P_a\,\widehat{N}_2\,(d) + N_3\,(d) \quad (8)$$

where N_2 (d) is the expected number of packets of the situation in which average queue size is lying between [T_{thr} (min), T_{thr} (max)]. N_3 (d) is the total number of packets discarded when average queue size is larger than maximum threshold T_{thr} (max) value.

Early Deadline First (EDF) was originally being used in wired network. The algorithm has been implemented in real-time transmission such as video or voice which has delay requirement, for example rtPS and UGS services class. In [16], the EDF algorithm for uplink happens at SS where the algorithm determines bandwidth to SS with earliest deadline and assign deadline to each packet. The scheduler services the packet in earliest deadline first with minimum deadline among all connections been selected, and packet will be discarded if the packet deadline is missed [17]. The formula for EDF can be calculated as below.

$$\text{Deadline} = \text{arrivalTime} + \text{latency} \tag{9}$$

In Deficit Round Robin (DRR), each connection has deficit counter which is initially zero. First packet which is called Head of the Line (HOL) from the queue is served only if the packet length is <= (Quantum size + deficit counter) value. Otherwise, the quantum size is added to the deficit counter. Quantum size is referring to the number of bytes for each queue can transmit in a cycle. Moreover, the principal of DRR is to avoid the difficulty of having to know the mean packet size in WRR or RR scheme. DRR works by not knowing the mean packet size in advance. So it will maintain the deficit counter and also the fairness of flows [18].

Strict Priority (SP) is one of traditional scheduling algorithms where the scheduler selection of traffic which has highest priority queue first until it is empty. Then, it moves to next highest queue and continuously this process until all queues are served. The SP algorithm is evaluated in [19].

In Self-Clock Fair (SC) scheduling, the process is similar to WFQ in terms of serving priority of queue. Due to the finish time calculating in WFQ was rather complicated, SC offers lower computational complexity by using a virtual time function which defined to be the virtual finish time of the packet currently being serviced, as explained in [20]. Advantage of using this scheduling is the time taken to compute the service time is very short since the information is extracted from the packet itself.

In [21], a virtual time is computed as:

$$F_k^i = \frac{L_k^i}{r_k} \max\left(F_k^{i-1}, \text{v}\left(a_k^i\right)\right) \tag{10}$$

Researchers investigate the use of uplink hybrid scheduling algorithm towards satisfying the QoS service class in opportunistic scheduling environment [22]. Proposed algorithm is the combination of SP and Earliest Due Date (EDD) for SS. Existing related work has also be done by the researcher by classifying into two group which is channel-unaware schedulers and channel-aware schedulers. The main focus of channel-unaware scheduler is the priority and EDD scheduling. In priority scheduler, it does not perform well for BE and nrtPS service class. While for EDD scheduler, the algorithm does guarantee throughput for UGS. Hence the main objective of the research is to improve the BE and nrtPS service class by using the P+E scheduling algorithm. It also increases the throughput and reduces the delay while QoS for UGS, rtPS, ertPS are maintained at same time. The structure of P+E scheduler consists of two layers, inner layer for EDD scheduler and outer layer for priority scheduler. Inner layer EDD will schedules rtPS, nrtPS and BE application dequeue the packets and put in the EDD output queue. In outer layer priority, the function is more to involvement of packet deadline whenever scheduler the packets. It will drop the packet if the deadline time exceed limit. As in Fig. 5, it first schedules UGS, then e-rtPS and finally EDD queue.

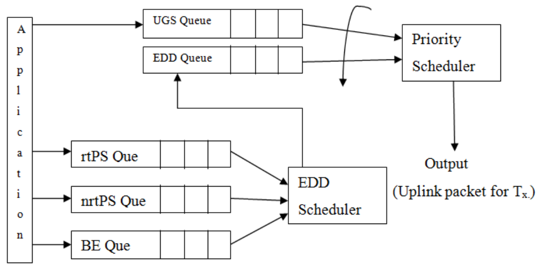

Fig. 5. Priority + EDD scheduler

Another part of the proposed hybrid scheduling is Deficit Fair Priority Queue (DFPQ) scheduling algorithm that works better in nrtPS and BE due to suitability in variable packet size, guarantee minimum bandwidth. Meanwhile, it eliminates starvation of lower priority service classes. The operation of DFPQ algorithm takes place after the allocation bandwidth of UGS and rtPS queues. In DFPQ, Quantum (Q) is allocated to each queue. Quantum of an *i*th queue (Q[i]) represents the maximum number of bits can be served in first round. Next, the scheduler visits nonempty queue after servicing Q[i] bits. If there are more packets in the *i*th queue, the remaining bits will be stored in queue state variable, which is called Deficit Counter (DC[i]) and scheduler continue to serve next nonempty queue.

IV. Simulation Model and Parameters

The simulation framework was referenced from [23] and [24], it consists of a BS and a number of SS that varies from 5 to 30 SSs in a PMP mode. The BS is directly connected to the SSs in LOS and the SSs are located surrounding the BS in a circular mode with a 1 KM distance as shown in Fig. 6. There are three CBR traffic generated by each SS; rtPS, nrtPS and BE traffic. Two scenarios have been created in this study. The first scenario has all the SSs in the region of 64QAM. Meanwhile, for the second scenario, 20% and 80% of the SSs are located in the region of 16QAM and 64QAM respectively. The performance metrics are throughput, delay and jitter as in [25] and [26].

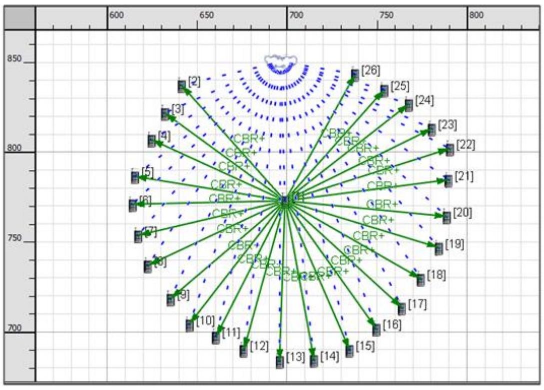

Fig. 6. Simulation topology

The WiMAX simulation parameters used for the simulations are shown in Table I, taken from [24] and for the traffic parameters are shown in Table II, derived from [27]. Property for simulation parameters is fixed throughout the simulation study whereas for traffic parameters are varies depending on the experiment environment.

TABLE I: SIMULATION PARAMETERS

Parameters	Value
Simulation time	60 Sec
Channel Bandwidth	20 MHz
FFT size	2048
Antenna model	Omni directional
BS/SS antenna gain	0 dBi
Transmission Power	20.0 dBm

TABLE II: TRAFFIC PARAMETERS

Service Class	Incoming traffic		
	Bytes	nterval (ms)	Data Rate
rtPS	400	3.2	1Mbps
nrtPS	200	6	240Kbps
BE	120	5	192Kbps

The simulator used in this research is Qualnet 5.1 simulator. Qualnet is a simulation software, design specifically for modeling large wired and wireless networks. The simulation predicts the behavior and performance of networks to improve their design, operation and management. Qualnet's kernel provides scalability and portability to run on hundreds and thousands of nodes on a variety of platform laptops and desktops. Users will interact with the kernel by using the Qualnet API to develop their protocol models.

Fig. 7 shows the modified BS downlink scheduler to be implemented in the simulator. The scheduler was taken from [28]. Due to each QoS service class has different scheduling algorithm, it was proposed to use an existing traditional scheduling algorithm where different service classes will be assign to four scheduling algorithms which are SP, RR, WFQ and SC.

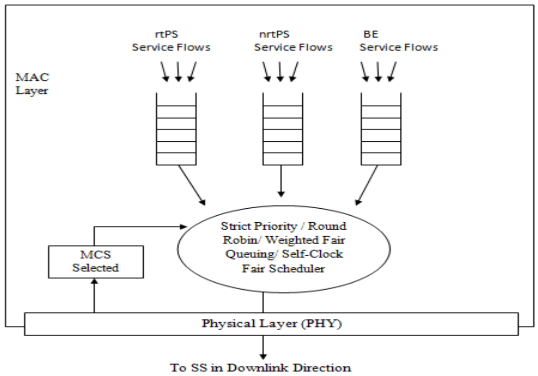

Fig. 7. BS downlink scheduler

V. SIMULATION RESULTS AND ANALYSIS

Fig. 8 illustrates the results for throughput of all the traffic classes with different schedulers; WFQ, RR, SP and SC respectively. It is observed that the throughput increased in accordance with the number of SS. However, when the number of SS increases to 25 and 30, the increment of the throughput has slowed down. This phenomenon is due to the maximum bandwidth has achieved. Furthermore, the performance for all schedulers is about the same with only small differences are noticed. As the number of SS approaches 25, the SP is slightly better than WFQ, RR and SC by 0.01, 0.37% and 0.37% respectively. When the maximum number of SS approaches 30, it is found that the RR is the best among all the schedulers. It surpasses WFQ by 0.29%, SP by 1.85% and SC by 0.56%. From the results, it is known that SP and RR are the best schedulers to perform at congested level.

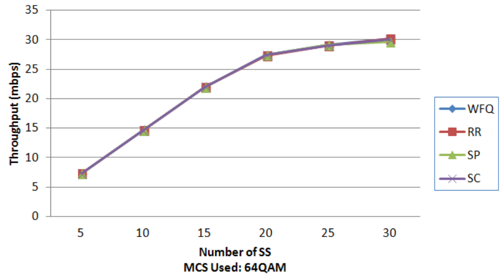

Fig. 8. Total throughput (64QAM)

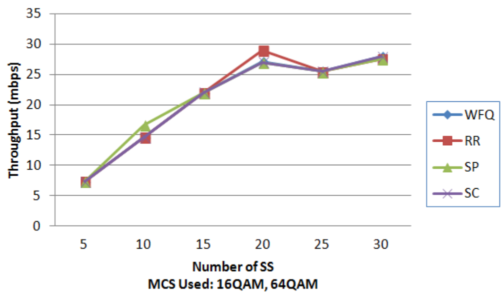

Fig. 9. Total throughput (16QAM&64QAM)

The total throughput for all the schedulers in the combination of 16QAM & 64QAM MCS is presented in Fig. 9. All schedulers share the same performance when the number of SS is 5. The small number of SS results in low traffic and it is still manageable by all schedulers regardless the MCS. However, the changes started to be noticed when more SSs attached to the BS. The SP scheduler achieved the highest throughput among the other schedulers by 11.9% at 10 SSs. In the between of 15 SSs and 25 SSs, the throughput is linearly increased, but there is a decline at 20 SSs before the throughput increases again at 30 SSs. This context indicates that the maximum throughput has reached at 20 SSs and minor fluctuation as observed even the number of SS increases. RR and SP scheduler are the top two highest leaving WFQ and SC in the throughput performance. The result is caused by the conserving factor implied by SP and RR. Overall, the schedulers in 64QAM having better signal compared to schedulers in the combination of 16QAM & 64QAM.

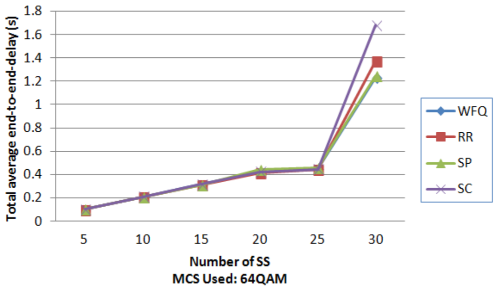

Fig. 10 Total end-to-end delay (64QAM)

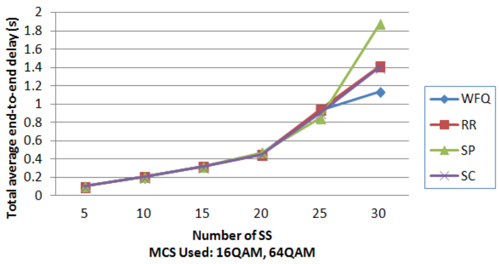

Fig. 11. Total average end-to-end delay (16QAM&64QAM)

Another important performance metric is the total average end-to-end delay, which is showed in Fig. 10. The delay is measured for rtPS traffic only because the latency is one of the concerns for real-time traffic but not for non-real-time traffic. From the observation, end-to-end delay increases as number of SS increases, this is due to the insufficiency of bandwidth to cater a large number of SS. Highest increment among schedulers can be seen when the number of SS is 30. The inadequate of delay control mechanism in SC results in highest delay leaving the WFQ, RR, and SP schedulers as many as 26.7%, 18.5% and 26.2% respectively at 30 numbers of SS. RR has shown its inability to handle large traffic with increased amount number of SS. Hence, it shows poor delay performance with only 18.5% lower than SC at 30 SSs. In RR, there is no priority or weight assign to the packet, where it equally serves slots to all queues and servicing a single packet from each queue. Hence, the RR is bad performed. In these experiments, WFQ and SP schedulers both are having good delay performance with 26.7% and 26.2% lower than the SC at 30 SS. The SP performs better because of its nature behavior where the highest priority is assigned to rtPS connections.

The total average end-to-end delay for WFQ, RR, SP and SC in the combination of 16QAM and 64QAM is depicted in Fig. 11. From Fig. 11, the delay increases 55% higher than Fig. 10 in average, when the number of SS is 25, whereby 5 SSs are in 16QAM. The difference is more significant when the number of SS approaches 30, whereby 6 of them are located in 16QAM. In terms of the scheduler performance, the difference between all the schedulers is about the same when the number of SS is below than 20.

However, as the number of SS approaches 25, the SP is at best performance, out passes the RR which is the lowest performance, by 9.7%. The combination of 16QAM & 64QAM MCS shows slowing down in the performance of the RR scheduler. The WFQ scheduler is the second lowest performance compared to the SP by 9.6%. Meanwhile, SC scheduler is the second best performance compared to the SP, it is about 6.6% lower. As the result, SC performs better than WFQ. Nevertheless, it is not as expected when the number of SS reaches 30, all the schedulers' performance change drastically. The main reasons behind this are the schedulers are unable to adapt too many SS and the effect of bad wireless channel quality.

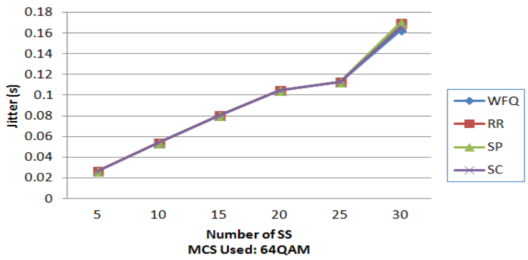

Fig. 12. Jitter (64QAM)

Fig. 12 compares the jitter which is also known as inter departure time among packets. The highest increment in jitter occurs when the number of SS approaches 30. Comparison shows that RR is 5.8% lower than the SP scheduler which has the highest value among all. No precedence or weight considerations for the RR scheduler are the causes that produce such result. As for the WFQ, the percentage difference between RR scheduler is 4.1% lower and the result justifies the WFQ policy which avoids resource competition in different priority of traffic has taken place. Higher buffer utilization in WFQ has proven its best performance in jitter. Besides that, SC scheduler is lower than the RR scheduler by 1.8%. SP is known by categorizing packets according to precedence value of UGS>rtPS>nrtPS> BE and in this research the allocation for rtPS traffic is higher than nrtPS and BE which might take long time to process real-time traffic, thus result in higher jitter. As the conclusion, the jitter results among all schedulers are about the same with only small different are noticed. Significant amount found for all the schedulers when the SS approaches 30, which is caused by the large number of SS.

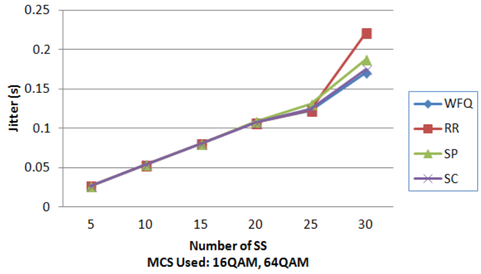

Fig. 13. Jitter (16QAM&64QAM)

As Fig. 13 illustrated, the schedulers' performances are about the same except for SP, which has significant difference. It can be seen that the jitter value increased in all schedulers in Fig. 13. The WFQ has increased to 3.2%, RR increases 10.8%, SP 7.1% and SC 4.3%. The result is expected as more than one MCS are used. It is found that RR is the most affected in jitter by increment of 27.3% compared to Fig. 12. This result is expected because the cause of different MCS that are used. The SP scheduler is in second lowest performance with an increment of 5.6% as compared to Fig. 12. The precedence value of priority has given the SP to show its improvement in performance although more than one MCSs are used. For SC and WFQ, both share the same increment percentage of 5.8% compared to Fig. 12. It also been observed that WFQ leads by having the best performance even when the network starts to be congested for more than 30 number of SSs.

VI. Conclusions

Results from the simulation and findings show that throughput increases as the number of SS increases. WFQ performs best in 64QAM and 16QAM & 64QAM with respect to the average delay. Assignation of weight in packets gives contribution in the performance of WFQ. SP performs second best in 64QAM but poor in 16QAM & 64QAM, in terms of delay. Its inability to adapt low channel quality wireless channel is the main cause for the bad performance. However, SP has the highest throughput among all schedulers. As for RR, it is among the best scheduler in throughput. But, for jitter and delay, its performance is very poor. This is known that the RR does not give any priority or precedence value for the packets. RR scheduler that does not consider the QoS classes fails to assure QoS for different service classes. It also becomes incompetency if the packet size in variable length which will result unfairness. On the other hand, SC was the second best performance in jitter 64QAM but it has extremely high delay in 64QAM.

Furthermore, the impact on lower MCS wireless condition also contributes to the poor performance among all the schedulers in the second simulation scenario. Hence, the schedulers need to be amended in order to suite the uncertainty condition in wireless environment.

Hence, mixed version scheduling algorithms with taking into account of wireless channel condition will be studied and proposed in the near future. A similar research work will also be extended to Long Term Evolution (LTE) to evaluate the network performance.

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