New Cross Layer, Resource Allocation scheme for Delay Sensitive Service in 802.16e and 802.16m

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Abstract

This paper provides an efficient channel aware resource allocation scheme for IEEE 802.16e and 802.16m WiMAX Mobile, real-time and delay sensitive service. Compared to a similar scheduling approach, our considered scheduler can guarantee and achieve lower delay with a good average throughput. In order to achieve this object, we introduce a scheduling scheme with three different segments in a decision making process. The first part, a time dependent function that considers the time when packets wait in queues and in a jitter area to prevent packet deadline. The second part, retrieved from proportional fairness algorithms, which in normal conditions gives a fair share to users. Channel SNR and service class weight are also involved in this part. The final section of scheduling relationship, channel condition, is defined more accurately by CQI, RSSI and CINR parameters. The simulation results in OPNET show that our proposed scheme has a very good delay and packet loss ratio accompanied by a high throughput. In another scenario, with different number of users and limit resources, we show relationship between admission control and scheduling.

Keywords: *IEEE 802.16e, IEEE 802.16m, Delay, Resource Allocation, WiMAX, Scheduling, OPNET.*

1. Introduction

The advent of high-speed media which mainly include voice, video, image, game and etc, has increased the necessity of bandwidth for transmission. In most of these media, users would like to have access not only in a specific place, also in a situation of mobility. In the present world, attempts have been made to set telecommunication infrastructures based on wired networks and instead, network toward user direction as is possible as use wireless transmission media. But, the largest obstacle for this method is channel or wireless transmission media and bandwidth limitation. Wireless channel varies and different factors lead to signal destruction; a very rare problem in wired networks, due to their isolation from the external environment. Lack of bandwidth is another problem too.

For this purpose, different institutions such as IMT, 3GPP, and IEEE have developed standards, and considered them in the field of research and development. IEEE standard competed with mobile broadband standard (3GPP and EGPP families) by presenting the mobile version of standard 802.16, i.e. 802.16e [1] and 802.16m [2], and is a powerful competitor for the 4th generation of the family in this field.

Eight key features can distinguish 802.16 standard family and WiMAX network from other wireless metropolitan networks which are: 1- Using OFDMA access technique, 2- Utilizing variable bandwidth (from 1.25 to 20 MHz), 3- Capability to use both TDD and FDD methods in connection, 4- Applying the technology of advanced antennas such as Beam forming and MIMO, 5- Modulation and adaptive coding per user, 6- Advanced coding techniques such as STC and Turbo coding, 7- High security, and 8- Different classes of service quality which are appropriate not only for a service like voice or data, but also for a combination of them.

The key point for establishing QoS, specifically for services such as voice and video with delay and error limitations, is the way of scheduling and allocating resources to users in order to remove these limitations and also maximize throughput to minimize consumption capacity and keep complexity of the algorithm at a desirable level and also consider different network dimensions [3]. IEEE 802.16 standard has not determined any mechanisms for resource allocation or admission control and has made manufacturer to undertake the responsibility.

IEEE 802.16 standard has defined the logical frame structure according to Fig. 1. In this structure based on ofmda, in which scheduling and mapping are more complicated than sc-fdm and ofdm, there are 3 important problems: 1- On the downlink side, the standard has bounded all the bursts to be rectangular [4], which has generated the problem of mapping data for the suitable slot. 2- The second point is the number of bursts in a frame; a rise in it would raise the header of UL-MAP and DL-MAP and consequently, the number of IEs would be increased [5]. 3- Each PDU has a 6-byte header. Due to error on the bytes, a rise in PDU size would increase

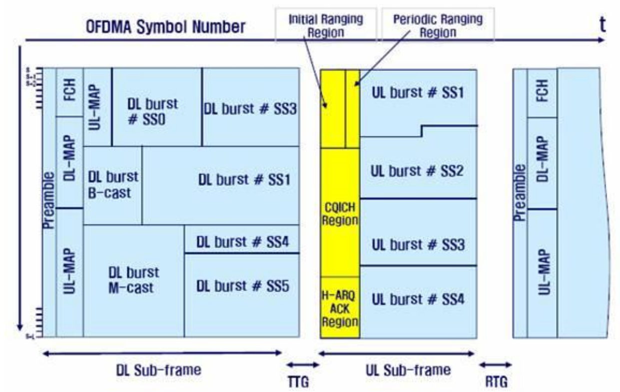

Fig. 1 The frame structure in 802.16 standard based on OFDMA

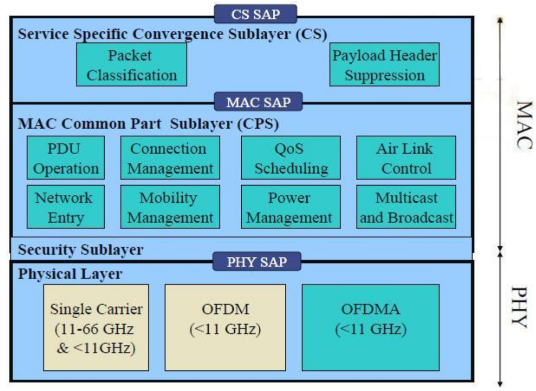

Fig. 2 Physical and Media Access layer in 802.16m standard

probability of PDU loss [6]. And vice versa, its great decrease causes the 6-byte header to lose its efficiency.
IEEE standard and 802.16 group, have only documented two physical and MAC layers. In this regard, the media access layer in WiMAX is divided into three main parts and is observable in Fig. 2. Scheduling and resource allocation are accomplished in the MAC-SAP sublayer. Admission control is another important process which estimates system and frame capacity for the admission of a new contact. Usually, admission rate exceeds the precise amount of capacity, since most of the connections do not use their maximum rate in transmission.
IEEE 802.16 standard has defined 5 classes for quality of service and, each has its own special parameters. 1- UGS: This class of quality provides constant bandwidth service with regular and periodic intervals. When a connection is established, there is no need to send any requests for bandwidth and mostly applied for realtime traffic with constant data rate. 2-ertPS: is applied for VoIP traffic along with silence removal, and no traffic is sent at the time of silence. It is similar to UGS because, in the active mode, the base station allocates maximum stable rate. Since no bandwidth is allocated during off period, base station should inspect to be aware of the silence timeout in the user's earphone. 3- rtPS: This class of service quality is related to realtime traffic with variable byte rate such as compact video traffic with MPEG format. 4- nrtPS: This class is applicable for asynchronous traffic with variable rate, as a result, no delay is assured but traffic rate is guaranteed. 5- BE: Most data traffic is in this class, where no throughput or delay is guaranteed.
The scheduler, in the base station, makes decisions for both the traffic in downlink transmitted to different users, and for the users in uplink to transmit information to network. For the downlink, the scheduler in the base station has complete information about length of queues and size of packets for decision-making. But on the uplink traffic, users require queues to send bandwidth request to the base station and accordingly make decisions on the allocation of slots numbers to users.

2. Delay-sensitive services and previous works

Scheduling and channel allocation are the main parts in the media access layer to ensure quality of service for different classes. Scheduler, in fact, acts as a resource distributor between users and connections. Allocated resources are the slots which are mapped into a number of sub-channels (each sub-channel includes a number of subcarriers) and time slots (OFDM symbol). In OFDMA system, the smallest logical unit is slot and its definition differs depending on type of sub-channelization. The scheduler should make decisions in both physical and logical areas. In the logical area, it should calculate the number of slots based on the class of QoS and, in the physical area, it should make decisions on the OFDM symbol (i.e. time) and subcarrier (i.e. frequency) that are more appropriate for sending.
Scheduling and channel allocation are performed in three distinctive processes. Two of these processes are accomplished in the base station, one for uplink and the other for downlink, and the third process is done in the user equipment (Fig. 3). In downlink, packets of the upper layer are put in different queues and it is ideal to have one queue per CID (this method is appropriate to confront blocking of HOL packets of the queue). Also, queues can be optimized and their number can be reduced. After making the queues based on quality of service parameters and other applied information like channel conditions, the downlink scheduler makes decisions to determine the queue and number of SDU which should be put in servicing mode; it means transmit their own packets in the forward frame.

2.1 Delay-sensitive services

There are two general methods to request bandwidth: incremental and cumulative. Each includes several mechanisms, which is either explicit or implicit. Bandwidth requesting methods are unsolicited polling, piggybacking, poll-me bit, bandwidth stealing, contention region, Code

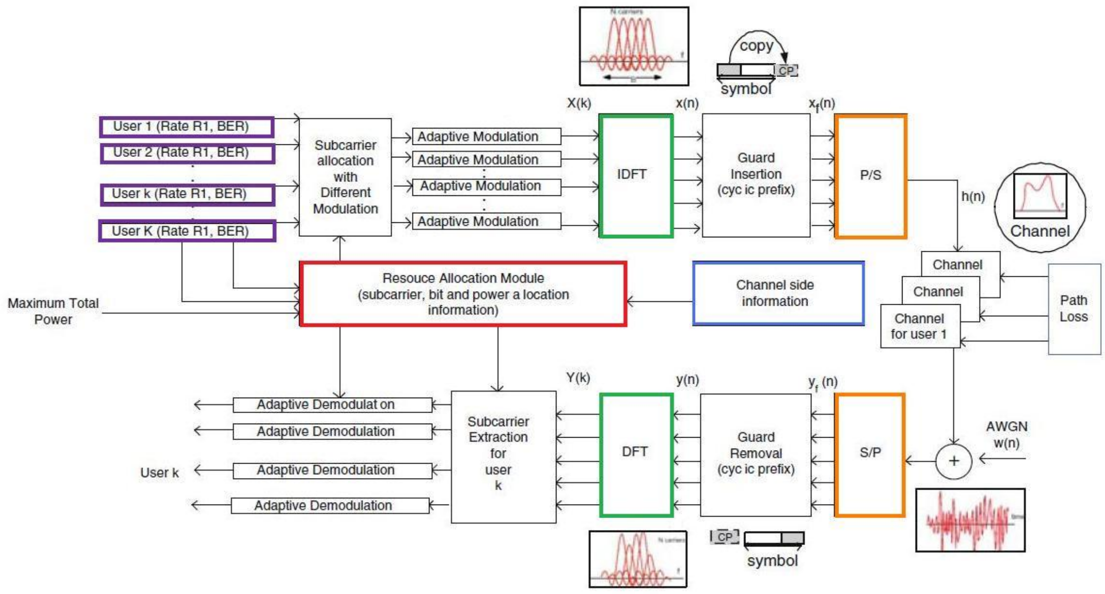

Fig. 3: Central resource allocation and scheduling scheme based of channel parameters

word over CQICH, CDMA code-based, unicast polling, multicast polling, broadcast polling, and group polling.

Choosing the optimal method for request in each class of QoS is selective and disputable. A balance should be established between resource optimization and QoS needs. For example, multicast polling method guarantees delay [7]; but, if there is no packet in the queues for sending, it will seriously waste the resources; instead, unicast and broadcast methods make the resources more efficient and cannot guarantee delay.

Among different service classes in WiMAX, UGS, rtPS, and ertPS services are very sensitive to delay and jitter so that they may lose packets in case of excessive delay. UGS service does not use polling methods for quality class and scheduler should be aware of all class needs to use resources efficiently. For example, there are 10 bursts, each needs 500 bytes and must be send in one frame after 5 frame periodically; if in each frame only 2500 bytes are allowed, all 10 bursts cannot be sent in one frame, and consequently before full sending, another 10 bursts is coming, which is problematic since the scheduler needs to guarantee delay and error in addition to frame optimization. This problem will be more serious when another UGS service intends to enter or exit dynamically. From the viewpoint of delay guarantee, polling method is the best method in each frame and guarantees delay; but, it has high header.

The important point about rtPS service is that it is extremely delay-sensitive. If the determined deadline for the packet is passed, it will be thrown away and some solutions should be considered for avoiding it. However it may be acceptable for a very low amount of delay. Finally there is ertPS service, this class of QoS for VOIP traffic has silence and active intervals. A famous sample is sound along with AMR coding; during activation (speaking), it sends 33 bytes per 20 ms and, during silence, 7 bytes are sent in general. The scheduler needs to be aware of silence periods so not to waste resources by bandwidth allocation at this time.

In ertPS and rtPS class, the fundamental issue is the observance of failure passing from the deadline of packets. Also, it is important for the base station to know whether there is a packet for sending or not. Polling mechanism should be smart enough so that, when there is a packet for sending, it could allocate bandwidth through the station and transmit data packet before the end of maximum possible delay.

2.2 Available Scheduling Algorithms

Most scheduling and channel allocation algorithms are related to the type focused in the base station [8], specifically scheduling for downlink services. In these schedulers, queue length and packet size are easily accessible. Polling mechanism is applied to guarantee QoS for uplink from the user side. Recent scheduling and resource allocation techniques for WIMAX can be classified into two major groups: channel-aware scheduling and channel-unaware scheduling. Obviously, the channel-unaware algorithms do not use channel information for decision-making, which is a scale for classifying scheduling algorithms.

Channel-unaware algorithms such as FIFO, RR, WFQ, DWRR, or EDF generally assume that channel is free of error and thus quality of service can be easily guaranteed; in contrast, in wireless channels, changes are very high due to noise, fading, attenuation, and interference and the scheduler needs to be aware of channel conditions.

The algorithms based on channel information are classified into 4 different groups according to their goal such as: 1-

fairness, 2- guarantee of QoS [9], 3- maximize throughput, and 4- optimize power. CINR, ratio of power of carriers to interference and noise, are the channel information used more in decision-making. CINR approximation is reported from the user to base station via CQI channel and is used in modulation and coding scheme. In all channel-aware algorithms, it is assumed that channel conditions are constant along the frame; i.e. in 5ms of the frame length which is conventional for mobile WIMAX, the channel condition is the same measured value; if the conventional assumption of constant channel conditions during 15cm around the antenna is considered, around 120 km/h is obtained as the allowable speed for mobile WIMAX. Generally in all methods, scheduler desires users with optimal channel conditions and places them at their high priority in order to reach multi-user diversity, confront fading, and prevent allocation waste due to bad channel conditions and loss of packets.

The most important class of algorithms which guarantee quality of service is Modified-LWD [10], the optimized and redefined LWD algorithm. In fact, every algorithm that presents a different, but related definition for LWD may be placed in Modified-LWD family and can guarantee QoS by guaranteeing minimum throughput and maintaining delay lower than a determined threshold. It is proved that throughput is optimum for LWD algorithm. Modified-LWD algorithm can also reach this throughput; however, there are regions in which decreased throughput occurs. This algorithm also uses the conditions of channel and queues so that the user with maximum amount in Eq.1 is allowed to access and transmit:

$$K_i \times D_j(t) \times R_j(t) \quad (1)$$

where i denotes class of QoS, j the queue of the jth user, Ki a constant which varies for different classes (the most difficult part is K_i determination), $D_j(t)$ may be delay of the forward packet in queue j or Head packet or jth queue length, and $R_j(t)$ is channel rate for user or *j*th queue, in which channel conditions are considered. These parameters are observed in Fig. 4.

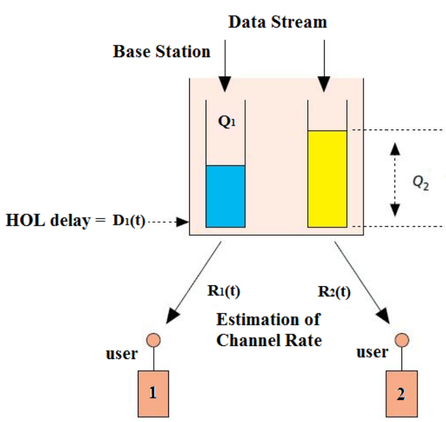

Fig. 4 Parameters used by M-LWD Algorithms.

Different definitions have been proposed for this relation; for example, in [11], the scheduler chooses the i^{th} user, in which Eq.2 is maximum for kth subcarrier.

$$(Ri/Ti) \times Hi \times G(i,k) \quad (2)$$

*R*i is average received rate, T_i, average throughput during a determined time window that will shift over time, H_i is the time, during which Head package has been waited in the queue, and $G(i,k)$ is ratio of second power of noise in receiver to variance of AWGN noise. Conditions of buffers are considered in delay of Head packet and channel conditions for k^{th} subcarrier in the channel gain.

Another algorithm is EPS [12] that has a function similar to Modified-LWD and is applied for a more efficient utility from radio resources and acceleration in emergency transmission (two important parameters in scheduling). This scheduler first measures priority degree based on emergency factor for all users, indicated by $U'i(t)$ and depends on queue delay and spent time. A user who can maximize Eq.3 will achieve priority in resource usage.

$$|dUi(t)/dt| \times R_i(t) \times [1/(dR_i(t)/dt)] \quad (3)$$

$Ri(t)$ is current calculated rate of channel for i^{th} user and $dRi(t)/dt$ is average rate in the last w slots which can be calculated by Eq.4.

$$dR_i(t)/dt = (1-1/w)(dR_i(t)/dt) + (1/w)R_i(t) \quad (4)$$

According to Jensens' model in [13], TUF (time utility function) might be considered in two forms for realtime traffic (Fig. 5), before deadline of packet error, it might be considered one (constant) and the slope could be given just within the jitter area (Fig. 5.U1) or sloping could start out of the jitter area and would take a number between 0 and 1 (Fig. 5.U2). Getting closer to the deadline would raise slope.

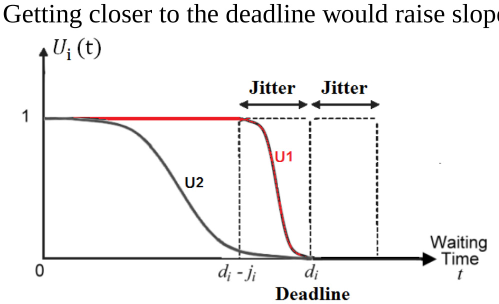

Fig. 5 Two different type of time utility function for realtime traffic

When waiting time or delay in Head packet enters its permitted jitter area necessity arises, because slope is sharply increased. $dUi(T)/dt$ is the derivative of time utility function and determines slope considered in Eq.3 as an absolute value of its size with no sign. When this scheduler gives a higher priority to the classes sensitive to time delay rather than to delay-sensitive classes, it will lose its efficiency. Some definitions are presented here to modify this behavior; for example, waiting time in head packet queue is considered only for the classes which require service quality and impact of queue size is applied in BE class.

A few number of scheduling and channel allocation plans have been proposed for maximizing throughput [14], [15]. In these methods, ratio of carrier to interference is used to opportunistically assign resources to users with good channel condition. Two CINR and RSSI parameters are used to show channel quality in WiMAX. Calculating CINR in WiMAX system might be implemented by the standard proposed method, also calculation method can be done by vendor designed scheme. This value is calculated on the user side and presented to the base station. Base station calculates signal to noise ratio based on the received value and then selects desirable modulation and coding scheme. RSSI can be used in power control problem [16]. In throughput maximizing algorithms, the value of CINR is used more frequently.

3. Proposed scheme

The goal of this method is to provide quality of service for delay sensitive classes such as ertPS and rtPS. It is worth noting that resources are allocated for UGS class once at first and there is no need for allocation during the session; so, quality of service guarantee is applied for polling-based services. Furthermore, we need to be aware of queue conditions to guarantee QoS provided by polling. In contrast, fairness in resource allocation is important for BE and nrtPS classes. Although fairness is not much important in the classes requiring QoS but it is considered here [17].

For polling-based classes, inter-class scheduling is proposed by developing EPS algorithm, which performs scheduling for rtPS and ertPS classes. For the term $1/(dR_i(t)/dt)$, EPS behavior is similar to the behavior of proportional fairness algorithm or PF, which deviates from fairness only at the moments when slope of utility function is increased and allocates resources to the packet with a close deadline and needs to provide service quality. Moreover, impact of buffer sizes has not been considered in this function. One reason to consider two classes for a scheduling algorithm is to reduce complexity and calculation volume of the system. Furthermore, if a separate algorithm is considered for each of them, again another inter-class scheduling algorithm would be needed to make decision between two classes. Indeed, inter-class prioritization method is mostly used. In other words, when each of the classes arranged their queues by an inter-class scheduler, they would send their own packets based on priority. Assigning a number for class priority, for example number 3 for one class and 2 for another, cannot meet the system need and is usually called a rough estimation. At the same time, presenting one scheduler for two classes brings up different condition in calculations, lack of appropriate observance of which leads to functional weakness. For this purpose, "soft quality of service parameters tracking" has been chosen for the presented solution. Eq.5 indicates a statement which is applied in this method and gives service times for a user who can maximize it.

$$\left|\frac{dDelay_i(t)}{dt}\right| \times \log\frac{RSSI}{|dQueue_i(t)/dt|} \times e^{\frac{DataRate_i(t)}{Utility_i(t)}} \times Normalized(CINR) \quad (5)$$

Where, *Queuei*(t) is the queue waiting function, $Delay_i(t)$ is queue waiting function at the moment of decision-making, $DataRate_i(t)$ is user rate in forward channel based on CQI channel information, $Utility_i(t)$ is average user rate in Wi window related to several last frames, and Normalized(CINR) is weighing coefficients of channel conditions. All of these parameters are discussed in two separate sections below.

A) $Delay_i(t)$ and $Queue_i(t)$ functions: Delay in queue is very important for rtPS and ertPS classes and, when reach to deadline, packets disappear. However, in these classes, size of packets is smaller than that of nrtPS class such as VoIP. So, in each function, the following formula is used to calculate its value at that moment. In Eq.6 and Eq.7, the amount of delay impact for ertPS and rtPS services is separately calculated.

$$Delay_i\_ertPS(t) = \begin{cases} t & w_t \le 0.8D \\ e^{-3t} & w_t > 0.8D \end{cases} \quad (6)$$

$$Delay_i\_rtPS(t) = \begin{cases} t & w_t \le 0.6D \\ e^{-2t} & w_t > 0.6D \end{cases} \quad (7)$$

In this formula, D is maximum allowed delay in queues and W_t is current waiting time in queue. As observed in Eq.6 and Eq.7, delay slope for ertPS exerted its changes faster than for rtPS.

In order to implement size function of queues, Eq.8 was used. This equation was equally considered for both services. Since there is no large differences in the size of packets, queue filling and buffer overflow should be prevented.

$$Queue_i(t) = \begin{cases} t & B_t \le 0.8B \\ e^t & B_t > 0.8B \end{cases} \quad (8)$$

In this equation, B_t is current size in queue and B determines general capacity of the queue.

B) $DataRate_i(t)$ and $Utility_i(t)$: *DataRatei*(t) is in fact data rate for forward sub-channel. The scheduler estimates and calculates rate in forward sub-channel using modulation and coding scheme. For example, if QAM64 modulation with 3/4 code rate is used, it can send 8 bytes in each subcarrier, which is 6 bytes considering its useful code rate; if each sub-channel is 48 subcarriers with two OFDM symbols, rate of sub-channel is 6×48×2; it means 576 bytes for i^{th} user. Modulation and coding scheme makes decision based on SNR, which is not sufficient and will be discussed later.

W_i parameter which was formerly known as *T*i in equitable sharing formula plays an important role in scheduling and is usually considered constant; but, here, it was considered in proportion to class type and average number of frames required for transmitting one packet from user's QoS class. Eq.9 indicates calculation manner of $Utility_i(t)$, in which parameter W_i is different for two classes according to table 1.

Table 1: W_i Parameter for 2 class of delay sensitive service

Class of QoS	*W*i
rtPS	Average Required frame for sending packet ×3
ertPS	Average Required frame for sending packet

$$Utility_i(t) = \begin{cases} \left(1-\frac{1}{W_i}\right)\mathrm{Utility}_i(t-1) + \frac{DataRate_i(t)}{W_i} & q_i(t) \neq 0 \\ Utility_i(t-1) & q_i(t) = 0 \end{cases} \tag{9}$$

In most of the existing algorithms, size of this window is constant and mostly considered 1000. During the simulations, change in the size of this window such as 10, 100, and 1000 is studied. Also, there is less relationship between window size and other parameters. Here, window size has gone out of the constant mode and is related to a parameter via making a relationship with the number of frames required for transmitting and weighting. Average user rate is related to the number of required frames for sending its packets, because if the number of frames is high, the rate required for more sending will be higher and, if the number of frames is lower, the rate needed for sending will be less. Numbers 1 and 3 are in fact the priorities that were assigned to classes during inter-class scheduling and its effect is considered here.

C) Finally, CINR and RSSI are considered. In standard 802.16 and OFDMA system, SNR is used for decision-making about the application of modulation and coding in each sub-channel (a capability which exists in OFDM in contrast to OFDMA). Although the standard has obligated CINR and RSSI calculation, it is not obviously used in modulation and coding scheme and it is assumed that, in case of more reduction of CINR (due to high interference), it will affect noise and SNR. SNR and SINR are obtained by averaging CNR and CINR on all subcarriers, respectively. Impact of CINR and RSSI, specifically CINR which shows interference, cannot be ignored. The interesting point in measurement is that, for example, when RSSI=-50dBm and CINR=12dBm, QAM16 modulation is used. Also, when RSSI=-80dBm and CINR=23dBm, such a modulation is again applied. Converting dBm into mW equation is the inverse of Eq.10 and Eq.11.

$$dBm = 10 \times \mathrm{Log}_{10}^{(\mathrm{mW})} \tag{10}$$

$$mW = 10^{dBm}/_{10} \tag{11}$$

Result of converting RSSI into mW is the value of about a few thousandth, while the converted value of CINR into mW is a number larger than 10. Indeed, applying high weight to RSSI is not correct, because it shows the power of received signal and signal power may be still high in the case of high interference. Also, if it is conventionally applied, there will be no change in the value, because it becomes a very smaller number than CINR. To consider CINR values (such as throughput maximizing algorithms) and RSSI, they were brought in normalized forms as one formula.

The obtained relation was achieved based on the point that different weighing for parameters and their values were needed. Using this relation, channel is considered more accurately in decision-making and it is not only SNR which applies its own weight by $DataRate_i(t)$ in modulation and coding scheme, but also more parameters of the channel were considered. Another point was the assumed difference between the two classes.

3. Simulation

The object of this project was to improve the performance of access layer to media in WIMAX and the major part of work was done in scheduling as a part of 802.16e standard that has not been defined. The goal was to consider the behavior of the proposed algorithm in a real WIMAX system in order to evaluate the impacts of other layers, ARQ and HARQ, mobility, handover, application possibility, acceptability of calculations, accessibility of desired parameters, behavior in the worst scenarios, effect of current header in frames and packets and so on, which can only be studied with having real layers of WIMAX and changing them. OPNET software [18] which was used for its simulation is applied by most of large companies in the world such as Ericson, Nokia, Alcatel-Lucent, AT&T, Vodafone, and …. It can be also mentioned that this software is one of the best in the field of network and wireless communications. In channel-aware scheduling algorithm, parameters of physical layer are needed in access layer to media, which include CINR that is calculated in physical layer and transmitted via CQICH channel. Therefore, channel-aware scheduling algorithms might be called Cross Layer.

3.1 Implemented scenario

In this scenario which was provided by the structure according to Fig. 6 and simulation parameters in Table 2, delay, throughput, and lost rate of packet per second were investigated and no additional traffic or mobility was made in order to accurately observe important figures for quality of service.

Table 2: Simulation parameters

Mobile Station	MSR (Mbps)	MRR (Mbps)	Traffic	Transmitted Frame	Transmitted interval	priority
ertPS	0.7	1	game	10 Kb	0.01 sec	١
rtPS	1.5	1.5	video	100 Kb	0.1 sec	٢
nrtPS	1	1.5	download	700 Kb	0.5 sec	٠
Access	Freq.	BW	All Subcarrier	Useful Subcarrier		subchannel
TDD-OFDMA	5 GHz	20 MHz	۲۰۴۸	1440 Downlink 1120 Uplink		PUSC
Pathloss	Shadowing		Multipath	BTS Transmitted Power		MS Power
Free Space	No		ITU Pedestrian B	0.5 w		0.5 w

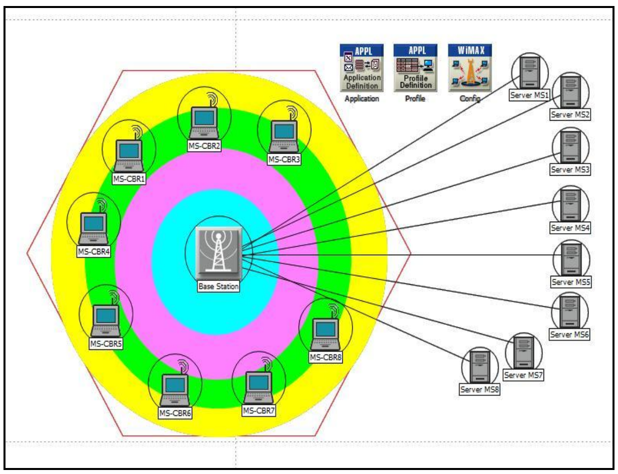

Fig. 6 First Scenario Diagram

In Fig. 7, average delay for different algorithms is visible. In this figure, two modes with sliding window, variable W_i, and without it were considered for the algorithm. No change occurred in the average delay; at the beginning, delay with window in variable size was less than delay in constant window, because in the latter case, average rate of the previous frames were considered zero at first; but, in the proposed method, based on the number of the required frames, it had higher weight and transmitted faster.

No study was seen about jitter on delay; however, the standard has determined a limitation for it and it is among the parameters which should be considered in quality of service. The higher delay in sending and, the more the waiting time of packets in the queue, would cause the higher jitter. In fact, jitter is related to the oscillations around the average delay and these oscillations along with delay determine the entry into jitter area and must not exceed a determined limit for a special class.

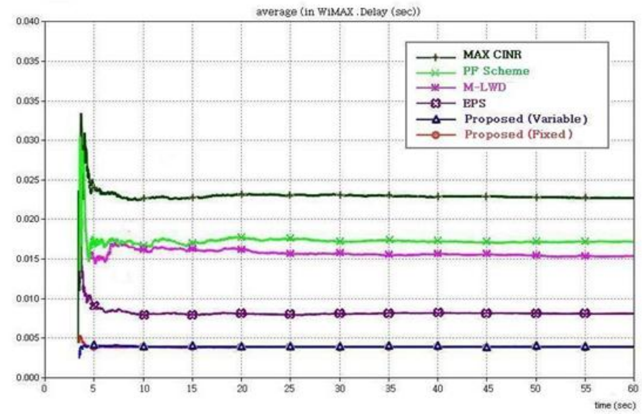

Fig. 7 Average Delay for different Algorithms

In all the studied conditions, no algorithm can improve throughout as much as MAX CINR, because users are in the best selection conditions and most of the packets are correctly taken to the destination. However, other considerations like delay or size of buffers do not exist and quality of service cannot be guaranteed. As can be observed in Fig. 8, throughput had a behavior like MAX-CINR in the proposed method for a large number of users and performed better than EPS, M-LWD, and PF algorithms. If the channel conditions are more precisely considered in decision-making and appropriate resources are allocated to users, the number of lost packets would be decreased and there is no need for retransmission. Consequently, the system throughput would grow. As can be observed, effect of RSSI was not considered in one of the figures, which compared with the mode its effect was considered showed less passing.

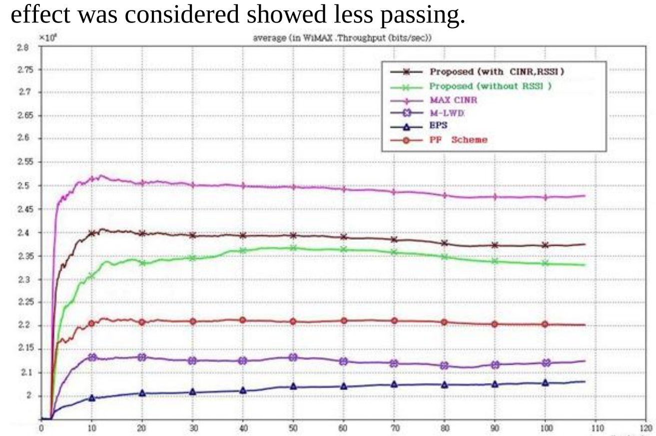

Fig. 8 Average throughput for different Algorithms

As Fig. 9 shows, EPS algorithm had a better condition than PF for having the time utility function. EPS had more changes, since it used a utility function in handling 3 different classes of service quality: ertPS, rtPS, and nrtPS. In contrast, in the proposed solution, this utility function had two criteria and in terms of delay considered less importance for nrtPS.

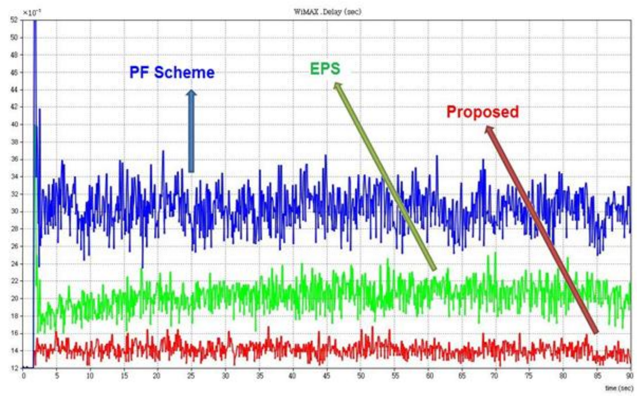

Fig. 9 Instant Delay and it's variation around average delay

In the proposed algorithm, effect of queue length was considered too; i.e. when the buffer size reached 80% of its maximum, it was necessary to prevent its saturation. For the simulation, buffer of the base station with the maximum size of 1 MB was to have an easy traffic and also network layer traffic or IP Flow was applied. It can be observed in Fig. 10 that, when $Queue_i$(t) function was exerted, the input and output behaviors of the buffer became more stable; however, lack of its application at the load of 9 MB/s stopped the saturation of the buffer of the base station. But, its slope was incremental and there was higher saturation possibility of the load.

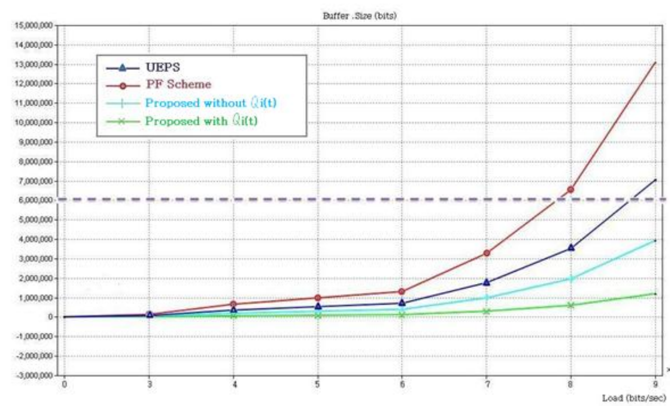

Fig. 10 one megabit buffer size in different load

4. Conclusions

Since the channel condition is changing in wireless environment, resource allocation and their scheduling without considering channel condition for wireless systems cause loss of resources. So in this article, a solution was proposed for channel-aware scheduling in order to improve quality of service, decrease delay in sending, and increase throughput. Furthermore, some attempts were made to reduce volume of complexity by combining scheduling for two classes of QoS based on polling and delay-sensitivity. In the proposed solution, a higher priority was assigned to the packets close to their deadlines and buffers which were being saturated. Moreover, an inappropriate channel decreases the chance of achieving resources. In the proposed method, the impact of some parameters was considered for the first time; therefore, there are enough space for changing and improving the formula. Whether the number of frames through which large packets might be sent by the scheduler or sending all the packets via more allocation in one frame can be investigated. Reduced delay was one of the major specifications of this technique; in spite of more improvement in passing than most of the algorithms but it was still less than other algorithms and some efforts should be made to achieve it. Another effective factor for improving the system function and optimal performance of affairs is cooperation with admission control unit in terms of the number of users and their type of service. While the effect of power management and silence or wake time was not considered.

Finally, the system complexity would extremely decrease if a solution can be proposed by which another polling class (nrtPS) may use this scheduler too. The major difference of this service is the lack of sensitivity to time, while the lack of buffer overflow in this class of service has the utmost importance.

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