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IEEE 802.16 E&M SYSTEMS BANDWIDTH RESOURCE ALLOCATION PERFORMANCE INVESTIGATION

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Abstract :– This paper concentrates on the downlink resource allocation strategy for the bandwidth in Mobile WiMAX system based on IEEE802.16E and IEEE802.16M standards. Most of the recent algorithms deal with giving higher priority to the applications that have higher delay tolerance known as real-time applications. While the applications using the non-real time doesn't have enough priority or interest. still, the real-time applications should have a higher priority than the non-real time applications to be served first. In this paper, a higher priority and QoS (quality of services) will be given to the real-time applications while the applications for the non-real time will be helped to give some fairness to decrease the dropping packets without starving. It will be done by merging two major technologies (minimum resource allocation and round robin manner. Also, a comparison will be done between the IEEE802.16E and IEEE802.16M systems to evaluate the system throughput, the resource allocation, the average end-to-end delay, and the data packet dropping to demonstrate the affecting algorithm of the entire system. Finally, the paper will demonstrate that the total system will be enhanced by reducing the nonefficient dropping packets. As a result, it will simulate and verify that there is lesser packet drop in the applications of real and non-real time. therefore, every single service and the total system throughput will be enhanced.

Keywords: *B.W Bandwidth, Resource scheduler allocation, (QoS) Quality of Service, OFDMA, WiMAX IEEE 802.16 E and M.*

1. Introduction

The mobile IEEE802.16E model system was made to enhance the weakness and the poor performance of the system IEEE 802.16 fixed broadband access. It's based on the WiMAX which stands for worldwide interoperability for mobile access technology. it delivers and supports more distance area and more system total average performance.

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As well as, the IEEE802.16E model improved many different types of features in the standards system of IEEE 802.16 such as the MIMO (Multiple Input Multiple-output) technology and the use of elastic subchannels,

where it allows the sender or the Base Station to support the data rates maximum downlink to reach 128Mbps using a BW bandwidth of just 20 MHz [1].

It also supports and enhances the different types of services used for the real-time multimedia applications, where these services need higher priority to take their turn in sending their packets through the transmission such as real-time video and audio, live broadcast, interactive gaming, and other data services that require less delay. Table 1 shows the different types of application requirements for the loss and delays sensitivity [2].

Table 1: Types of Application Requirements

Application	Losses	Delay Sensitive	Rate
File transfer	Not Acceptable	No	Elastic
Email	Not Acceptable	No	Elastic
Web	Not Acceptable	No	Elastic
Audio/Video real time	Acceptable	Yes, hundreds of ms	audio: 5kbps-1Mbps vídeo:10kbps-5Mbps
Audio/Video stored	Acceptable	Yes, some seconds	audio: 5kbps-1Mbps vídeo:10kbps-5Mbps
Interactive games	Acceptable	Yes, hundreds of ms	Starting at some kbps
Instant Messaging	Not possible	Partially	Elastic

The IEEE 802.16m is an extension model for the IEEE 802.16e model, which is also named as WiMAX 2. The main difference between them is that the first one uses one single carrier whereas the second uses several carriers to give a higher performance for the applications.

The enhanced WiMAX system uses some advanced technologies such as MIMO technology, OFDMA (Orthogonal Frequency Division Multiple Access), and AMC (Adaptive Modulation and coding), hence by using the previous technologies can affect and enhance the bandwidth usage. Due to the increasing size of users which forces the system to allocate enough bandwidth and resources for their demanding services [3], therefore, it will use the full link bandwidth for The high priority applications and makes other applications starve for more bandwidth resulting in a high increase in the packet dropping.

It can be shown from [3] and [4] that the approaches in both IEEE 802.16 E & M mainly focus on increasing the service priority and serve the data of RT applications first. While giving the non-real-time NRT applications some priority for its data services as much allowed as possible, in this paper a method will be used and demonstrated named as a Minimum Resource Allocation (RA) which it means allocating and reserving for the required bandwidth, every service flow as its minimum demand, but still one of the downsides of this technique is that the remaining unused bandwidth will be lost if it will not be reserved or used. Therefore, to solve this problem another technique will be used named round robin manner in which it re-allocates the unoccupied bandwidth to all applications SF Service Flow. Another goal of this paper is to simulate to decrease the dropping packets for the RT and NRT applications while guarantee QoS for different SF for RT applications, as it can be shown from the performance result of both model's E and M. as final comparison will be made between both systems regarding both throughput and dropping packets terms.

The paper will be sectioned as follows: ch 2 will explain the system methodology, formulation, and the analytical model description. ch 3, demonstrate the performance of both models along with the results. Finally, ch 4 will conclude the paper.

1. IEEE802.16 E & M System Model

The RA (resource allocation) in this paper will mainly deal with the utilization B.W for the usage of the spectrum efficiency level to support the requirement of a different number of users with different types of applications requirement services, which is used in both RT and NRT applications. Meaning each data service will allocate a specific slot from the available BW to be transmitted. A scenario was used to allocate these slots (basic unite for the E model) and DRUs (Distributed Resource Units) which are the basic unit of resource allocation for the M model from OFDMA frame to E and M users. Therefore, the RA will be depending on the available BW and resource number of which are available to be used and support the different network applications. The chosen scenario for this paper will explain the procedure of allocating slots as demonstrated in fig 1, where the data in the upper layers classify and divide the data into Service Flows SFs, and the MAC layer schedules them by their demanded SF. The WiMAX technology basically depends on OFDMA, which divides the BW into units for RA in both terms of time and frequency domains known as the data slots for the E Model, while the M model uses the DRUs distributed resource units. The data slots chosen for SF has the highest priority to be served and used depending on the input data values used for that transmission, also the BER bit error rate was chosen 10^{-3} as a static value for all types of SF. The chosen scenario also will help to serve the most higher priority SF for both UGS (Unsolicited Grant Service) and the (ERTPS Extended Real-time Polling Service, while to give the chance to the BE (Best Effort) service to be served. Therefore, the allocated bandwidth for all services flow must not be wasted, the chosen model will give both UGS and EERTPS services the highest priority for using the SFs types before other services [5], due to the delay sensitivity for these RT applications used. The slot allocator will use DRUs slots to allocate the other slots for each type of services. Finally, the OFDMA frame will send the data services to the PHY layer to be transmitted. The IEEE802.16E model uses five types of SFs as shown in table 1. While Fig 2 will show the methodology for allocating the Slots.

Table 2: Sfs types used by the IEEE802.16E model

Service Name	Abbreviation
Extended Real-time Polling Service	ERTPS
Unsolicited Grant Service	UGS
No real-time Polling Service	NRTPS
Real-time Polling Service	RTPS
Best Effort	BE

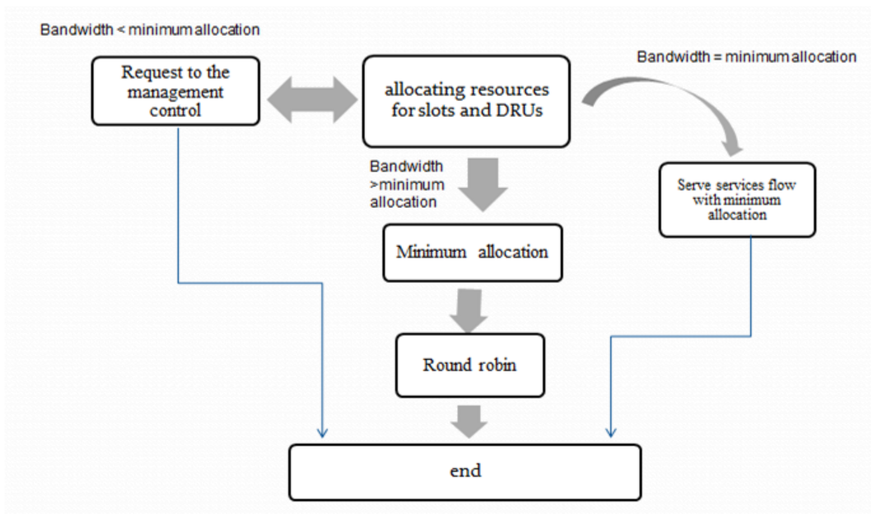


Fig 1: The Methodology for allocating the Slots

The slot allocator assigns the data slots to the frame of OFDMA, and also the allocator will allocate the requested DRUs, as seen in fig 2. Therefore for assigning and allocating the slots, two models the M & E modal's approaches were used.

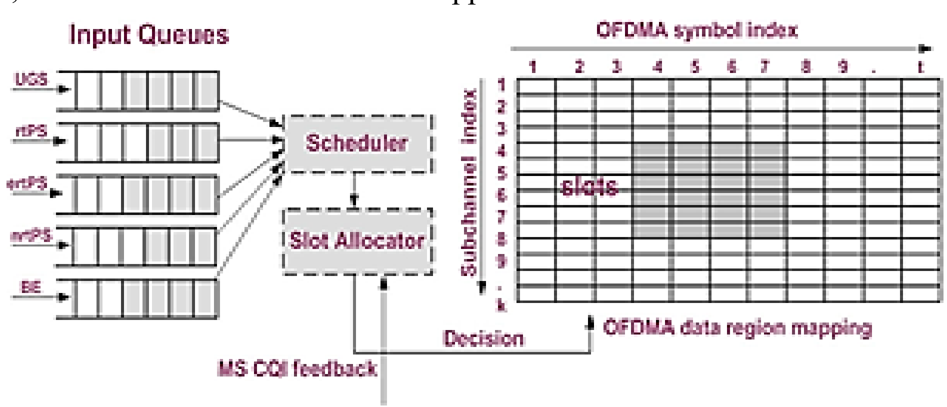
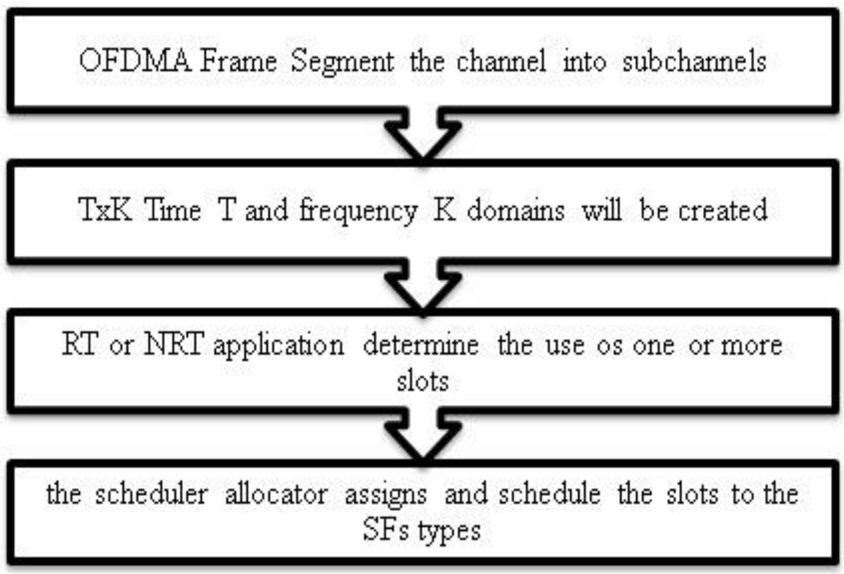


Fig 2: Scheduler & Allocator in the frame of OFDMA.

2.1 IEEE802.16 E & M Allocating Standard Methods

The frame of OFDMA used in the E-model segments the channel into subchannels in both time T and frequency K domains, therefore the TxK will give us the slot matrix that will form the frame of OFDMA, where each user uses one or more than one slot determined by the RT or NRT used applications, also the quality of the channel and the SFs requirement was determined by the BS before the transmission. Therefore, the services for each user will be ordered at separate queues. Finally, the scheduler allocator at the BS will assign and schedule the slots to the SFs types as the following flowchart diagram shows.



flowchart diagram : Allocating standard methods

In our scenario we will use 5 users to require the SFs, then each user will request different types of services as fig 4 shows, the KxT matrix units shows the number of users that attached to the sender BS, where each row is the user's SFs requirement, 5 rows ,5 users, and the numbers (21,000) in each first row determined the slots for the specific service, 2 for UGS, 1 for ERTPS, the 3 zeros is for (RTPS, NRTPS and BE respectively), and then it will repeat itself for 10 iterations to finish. the chosen scenario was simulated with the following parameters as shown in table 3.

```
data=[2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0];
```

Fig 3: User requirement for Service Flow.

Table 3: Scenario chosen Parameters

Parameters	Values
Physical Layer Tech.	OFDMA Tech.
No. of Users	5
Mode of Duplexing	TDD
Length of Frame	5 ms
Distance from each User	30 Km
Bandwidth of System	10 MHz
Spacing Frequency	1024
Length of Cyclic prefix	1/8
D/L permutation	PUSC
TTG + RTG	1.6 unit
Ratio of D/L:U/L	2:1 (29:18 OFDMA frame)
Preamble of D/L	one symbol per column
PDU MAC size	Variable - length
packing and ARQ	Disable
Fragmentation	Enable
Time per slot	625 µs
MAPs of D/L-U/L	four symbols per columns

To fill one frame for OFDMA with different subchannels the following equation will be used

$$the\ sub-channels\ per\ OFDMA = (Bandwithd)/(\text{frequency spacing} * (\text{sub}-\text{carrier no. per sub}-\text{channel}).) \quad (1)$$

Where the frequency spacing is the variance range between the allocation spacing in the frequency domain plan, the B.W will determine the bit-rate of all resources used in bps or its multiplications.
For path loss calculation, equation 2 was used.

$$\text{Path Loss} = 20\log_{10}\left(\frac{4\pi d}{\lambda}\right) \quad (2)$$

The path loss increases when the distance of the user increases from the BS transmitter, therefore the carried bits for each slot will be affected, as a result, the received power can be determined from the BS, resulting in determination of SNR by using the equations 3 and 4 respectively

$$\text{Power Received PR} = \text{Power Transmitted PT} - \text{PL} \quad (3)$$

$$\text{Signal to noise raion} = \text{Power Recived} - \text{Noise of the signal} \quad (4)$$

While the appropriate type of modulation for data transmission will depend on eq. 1, 2, and 5 with table 4.

$$\text{The Value of Modulation} = (\ln (5 * \text{Bit Error Rate}) - (1.5 * \text{Signal to noise ration})) / (\ln (5 * \text{Bit Error Rate})) \quad (5)$$

Table 4: Type modulation parameters

Types of Modulation	Condition
BFSK	If m bigger and smaller than 2
QPSK	m bigger than 4
16 QAM	m bigger than 16
64 AM	m bigger than 64

The process of the transmission for the SFs services in table 2 will choose the minimum number of packets to be sent, and the remaining unused data slots will be rounded again to be used using the round robin manner from UGS to BE [6]. Therefore, as a result, the OFDMA frame will be occupied partially by the types of PUSC subchannels, resulting in 48 sub-carriers for a total of 3 frames of the OFDMA that can be found in a single slot. After that, the UGS slots allocation will begin.

The 5 users will choose the services requirement of UGS and the number of slots needed, so each user will select the first data matrix column and then it will be multiplied by the bits per packet number and after that it will be divided by the number of bits in the slot, as shown in fig 4 the 1st alteration will show that all the users only used and demanded for 2 slots only. Likewise, for the Sfs ERTPS slots will be chosen in the same method, as seen in fig 6. Still for RTPS, 3 columns will be chosen and tested if the value resulted bigger than the number zero, the program will count the delay value, where the delay counter will add the transfer time of the data slots, and when it will reach the maximum delay number which is Tmax = 2 ms then the data slots will be assigned the frame of the OFDMA current frame so that the service of ERTPS will be used, finally the counter of the delay will be equal zero again, as seen in fig 5.

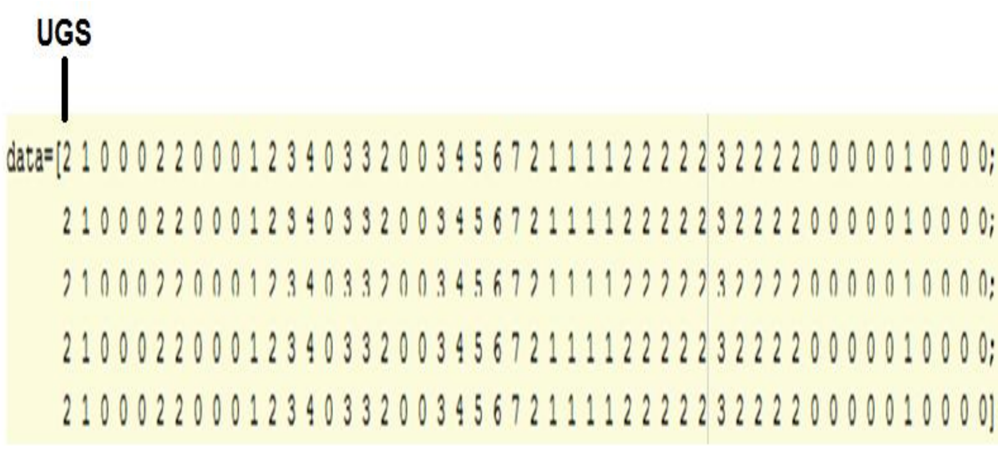


Fig 4: The data matrix of UGS' service.

ERTPS

```
data=[2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0];
```

Fig 5: the data matrix of ERTPS' service

RTPS

```
data=[2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0];
```

Fig 6: the data matrix of RTPS' service

Similarly the NRTPS SF tries to assign the slot and it will be the same by choosing the 4th row of the matrix data, and also the final value will be compared to the maximum valued needed to use the data slots if yes it will reuse the remaining slots needed or it will wait to reach the minimum value as shown in fig 7. Finally, the BE service slots will be also allocated, but still, if the buffer packets will reach 5, the packet will be transmitted and the BE counter will equal zero, or it will wait for the new packets to be received as shown in fig 8.

NRTPS

```
data=[2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0];
```

Figure 7: the data matrix of NRTPS' service

BE

```
data=[2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0;
      2 1 0 0 0 2 2 0 0 0 1 2 3 4 0 3 3 2 0 0 3 4 5 6 7 2 1 1 1 1 2 2 2 2 2 3 2 2 2 2 0 0 0 0 0 1 0 0 0 0];
```

Fig 8: the data matrix of BE' service

After all types of services slots was allocated as the minimum number required, in the end if remains more slots unused in the OFDMA frame, the scheduler allocator will start again to use the round-robin manner for all above SFs. Therefore, all empty slots will be filled fairly, leading to increasing in the throughput and the efficiency of the total system. During network congestion, the slots will be lower at numbers to be filled with the SF types, as a result the first two services namely UGS and ERTPS will have higher priority, due to their smaller delay tolerance, which causes the BE dropping its packets, therefore the throughput and the efficiency of the total system will be decreased also. Therefore, this paper will use some enhancement to overcome this problem. Where the scheduler allocator will send a message to the sender to inform him about the congestion in the network, therefore the MAC layer will force the controller to discover new slots or to replace the type of modulation used with larger slot number. To simulate this, let's say that the data inserted and the minimum number of slots are known for each SF, for assigning the number of slots. The unused slots number will be calculated by 6

$$\text{left slot number} = (\text{frame } slots \text{ total number } - \text{slots allocated}\ ) > 0 \qquad (6)$$

After applying the equation, and the slot number was sufficient, the remaining unused slots will be allocated using round-robin manner for the mentioned services types, resulting in a decrease in the system delay while increasing the system throughput. Finally to ensure the quality of the system we must calculate the packets drop number using the equation 7, and if the result is bigger than zero, it means there are some packet dropped otherwise not.

$$number\ of \text{ droppin packets } = (\text{minimum number of slots alocated } - \text{frame slots total number}\ ) > 0 \qquad (7)$$

After explaining the E model, the M model will be explained and simulated also, where the data used for the scenario of the M model is similar to the E model, the same data matrix, 5 number of users, five services types. 10 alterations were used, the distance is the same, and 4 different type of speed were simulated, as shown in the table 5. Also, all other parameters used are the same for the M model.

Table 5: E model user speed

Required bits Number	Meter per Second	Movement
192,000	192,000	Static
96,000	96,000	Slow
48,000	48,000	Fast
144,000	144,000	Ordinary

One of the biggest movement problems of users is the modulation type, which can cause the DRU to change its number of slots needed depending on the speed of the user, as a solution, each user will be classified depending on its speed mention in the table above.

The data will be distributed in the same method as the E model, where the 1st user will choose the 1st row for the requirement of the DRUs, and each 5-data matrix values for each row for all columns will be counted as 1 alteration and the SF allocation will be defined. To ensure the allocation of the UGS SF, the required number needed of DRUs after the transmission will be calculated by the bits block number of the FEC. While in the service of ERTPS it uses the same previous method. If the data size will exceed 4800 bit which is the size of the FEC including the CRC (Cyclic Redundancy Check), the segment will be divided to many blocks of FEC, which is distinctly encoded as seen in fig 9.

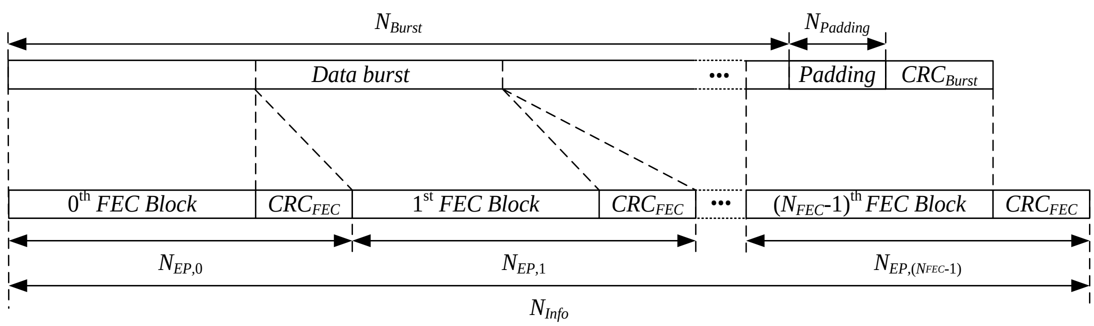


Fig 9: The encoding of FEC block segmentation & the FEC CRC block.

The division of segments for the FEC block will be used in the M model due to the utilizing if the FEC schemes to use lower power for the transmission over a longer distance with higher data rates, and same data of E model will be used to compare both systems. The DRU data transmission slots number will be allocated by the FEC schemes. Because the M model uses blocks of FEC, it will send the data matrix used to the block directly [7].

Same as E model, the RTPS maximum delay used is 2 ms, in case of the waiting queuing time is bigger than 2 ms the data service of RTPS will be inserted in the OFDMA slots frame to be transmitted. For NTRPS SFs the minimum packet number is 2, while BE is 5 packets per queue, more than that it should be served. Finally, after allocating all DRUs requirement, the round robin manner will start to guarantee fairness. This method will also be iterate for 10 times, and the packets drop will be calculated in equation 8, while the 1st user iteration dropping packet can be calculated by equation 9.

$$number\ of\ \text{dropping packets} = \text{minimum allocation of data} - (\text{data matrix} - \text{left data matrix}) \quad (8)$$

$$single\ \text{iteration use}r\ \text{dropping packet} = \text{summation of all (user dropping packets)} \quad (9)$$

In case of low B.W, the dropping packets will start, where it's value will be bigger than one, and when the B.W is high or enough the dropping packet will stop and its value will be 0. The throughput for a single alteration was calculated as follows

$$\text{Transmitted Data} = \text{total number of data matrix} - \text{left data matrix} \quad (10)$$

$$\text{Throughput of single alteration} = \text{Transmitted Data} + \text{round robin manner for data transmitted} \quad (11)$$

2. Scenario Results of IEEE802.16 E & M models

In most multimedia applications the delay tolerance for the RT applications are low, therefore, their priority for slot severing will be higher such as UGS' packets [5], while the NRT applications have lower delay tolerance, therefore, their priority is less, and the BE service will wait to reach five packets in its slots queue [5]. Likewise, the ERTPS will be served after the delay will reach 2 packets in the single queue. While other services such as NRTPS and BE will be equal to the required minimum throughput calculated. Again both simulation scenarios parameters simulated and used are the same.

For the E model as shown in fig 10 the services throughput was recorded, where the algorithm first served UGS service with two data slots, and the throughput started from 600 Kbps in the time of 1 ms.

While the ERTPS service started to be served when the delay value reached 2 ms. Similarly, after reaching the delay value of 3 ms, both RTPS and NRTPS transmitted their packets. Finally at value 5 ms the BE reaches the maximum packet number, and all slots will be filled with data services, the packet will be transmitted, otherwise, it will be dropped.

This happened due to that the transmitted packet was sent without resource allocation, while the BE service is used, therefore if there is any unpredictable congestion in the network or heavy traffic it will result in low utilization level, but still the packet multiplexing statistical allows to achieve higher utilization level, the BE service depends on network congestion and the load of the packets in the network, that's why the BE tries to transfer all the packets rapidly without received acknowledgments [8].

After assigning the slots for each service, and during the 6^{th} iteration, the highest priority services will be served while the others will wait for the round-robin manner to allocate their slots until it reached the 10^{th} iteration and no packets to be sent from other services just the UGS

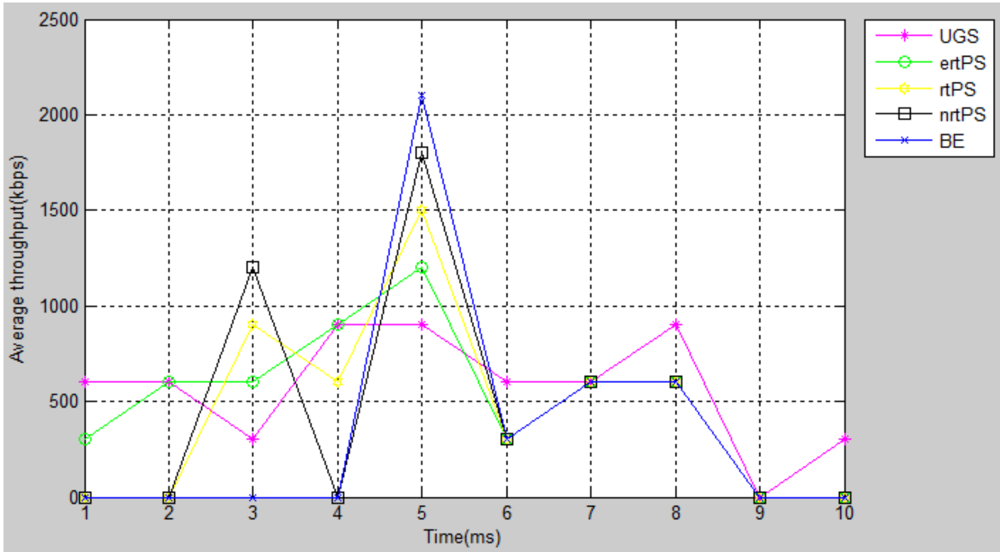


Fig 10: E model All Services throughput

In case there is enough B.W, no packets will be dropped, but if the B.W is not enough the scheduler allocator will decrease the dropping packets, by informing the sender to change the modulation type or to increase the slot size [9], as shown in fig 11, there are no dropping packets available.

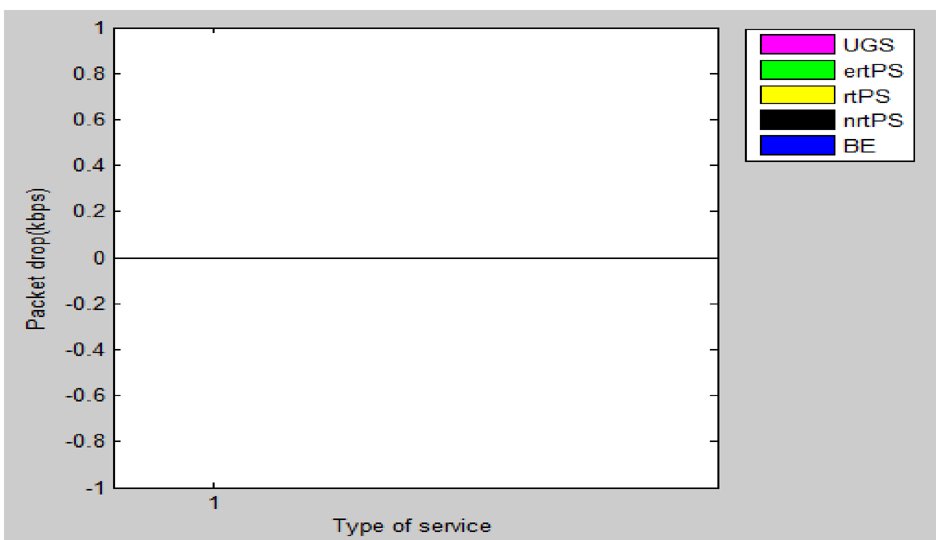


Fig 11: E model Packet drop for all services.

For the M model, which depends on transferring the required DRUs rather than using the data slots, the DRUs SF value was defined by using FEC blocks to decrease the error number in the channel. Again UGS will need two slots for the DRUs requirement to be allocated at the 1st iteration, starting from 1Mbps and then it will transfer its packets. As shown in fig 12 the other services require the DRUs allocating services instead of UGS. While BE and ERTPS reaching their maximum delay value of 2 ms, the NRTPS will reach its value for minimum requirement to transfer the data packets, which is also 2 ms, at 5 ms the frame of OFDMA will be filled with the data of SF without dropping any packets. Finally, at the 9th iteration, there are no transferred data in the system, resulting a zero throughput for the all types of services

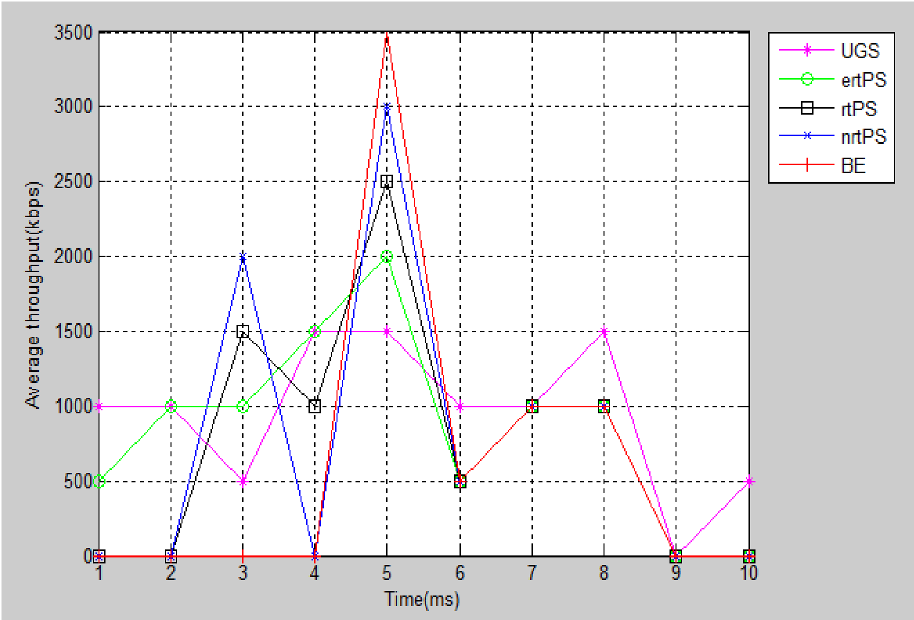


Fig 12: M model All Services throughput

Due to the transferring of all packet services in the UGS slots, there will be no packets dropped in the network as shown in fig 13. As a reminder, the two simulated approaches may not concern about NRT applications and their dropping packets.

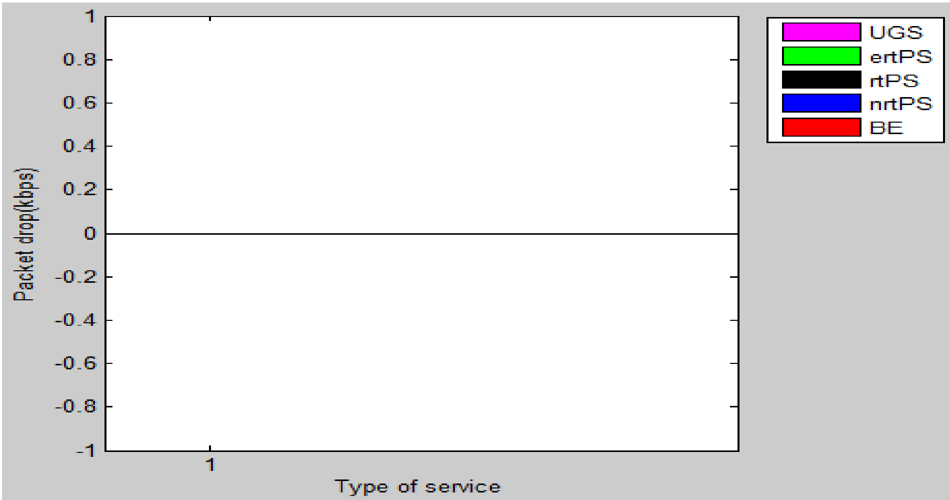


Fig 13: M model Packet drop for all services.

As both models compared, it can be seen that the E model shows better throughput, performance, and efficiency for using the network available B.W. resulting better scalability for controlling and managing the remaining B.W available without packets dropped. Still, some packets can be dropped if there is no enough B.W available. Therefore, only the RT application will be served and the NRT applications will starve and start dropping their packets. Subsequently, the M model shows higher throughput for all services types. As a result, there is no high difference between the two models, but it can be seen also that the M model can improve and fix the weakness of the E model by using the round-robin manner, that increases the fairness of the system and gives more chance for the NRT applications to send their data packets.

3. Conclusions

This paper simulates a proposed method of two systems IEEE802.16 E & M. The E model allocates and guarantee a fairness for both RT and NRT applications to send their data packets across the network. The proposed algorithm suggests that during the transmission only the services packets with higher priority that satisfies the QoS requirement should send their packets and if there are available slots in the frame a round robin manner technique will be used to serve other lower priority services, which gives the BE service fairness for transmission. Due to the use of BE service the system can sense if the slots are full or not, if yes it will send the frame if not the round robin manner will fill the slots with lower priority data services. To enhance the previous system and the transmission the M model used the DRUs to replace the slots to allow more bits to be inserted in the frame, also by using the FEC blocks, it will decrease the error of the channel. In conclusion, the proposed method is more than guaranteed to provide higher fairness and better scheduling services and increase the user B.W capability, and therefore decreasing the dropping packets to its minimum value.

As compared to other studies [10] shows that in the use of channel estimation improves the system performance, but the system suffers from severe performance degradation and high probability of error whenever channel estimation is not applied. where in [11] they proposed a distributed scheduling scheme, RTDS, for IEEE 802.16j networks. RTDS will allocate bandwidth dynamically for different types of connections to meet each connection’s QoS requirement resulting in less packet delay time and packet loss rate than other representative researches.

4. References

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