

# A Latency and Modulation Aware Scheduling Solution for IEEE 802.16 Base Stations

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**Abstract**—The mobile worldwide interoperability for microwave access (WiMAX) system provides high data rate, wide bandwidth for wireless network. However, the link quality is unstable owing to air interference and long-distance transmission. When link quality change, mobile node can revise the modulation and coding scheme (MCS) dynamically. Thus, we propose a modulation-aware scheduling solution for WiMAX base station (BS). It contains three primary methods: an admission control scheme (ACS), a scheduling algorithm, and a bandwidth allocation scheme. In addition to MCS, we also take latency for real-time service into account.

The proposed admission control scheme calculates the admission condition based on different kinds of MCS and minimum reserved rate requirements. The modulation aware scheduling algorithm and bandwidth allocation scheme can allocate bandwidth more efficiently in the finite frame duration and guarantee the minimum quality of service (QoS) requirements to each admitted connection by translating the QoS requirements into time slots. Finally, our simulation results show that the proposed scheduling solution can improve the throughput of the entire network and guarantee the minimum QoS requirements to each admitted connection.

**Keywords:** modulation aware, 802.16, WiMAX, QoS, scheduling.

## 1. Introduction

The WiMAX standard based on the IEEE 802.16-2004 [1] and IEEE 802.16e [2] specifies a metropolitan area broadband wireless access network technology. The main advantages of WiMAX are the higher data rate and the longer transmission range. The standard defines two basic operational modes to achieve medium sharing:

point to multipoint (PMP) and Mesh mode. In the PMP mode, a subscriber station (SS) can communicate with other stations only through the BS and thus the BS is responsible for managing the downlink and uplink transmissions and allocating bandwidth for SSs. In the Mesh mode, a SS can communicate with other SSs or BS directly.

To meet the QoS requirements of multimedia applications, the WiMAX standard defines a QoS signaling framework and five different scheduling service types: Unsolicited Grant Service (UGS), real-time Polling Service (rtPS), non-real-time Polling Service (nrtPS), Best Effort (BE), and Extended real-time Polling Service (ErtPS). The UGS supports real-time data streams consisting of fixed-size data packets issued at periodic intervals, such as VoIP without silence suppression. The rtPS supports real-time data streams consisting of variable-sized data packets issued at periodic intervals, for example, Moving Pictures Experts Group (MPEG) video transmission. The nrtPS supports delay-tolerant data streams consisting of variable-sized data packets for which a minimum data rate is required, for example, FTP. The BE service supports data streams for which no minimum service guarantees are required and therefore may be handled on a best available basis. The ertPS is suitable for variable rate real-time applications that have data rate and delay requirements such as VoIP with silence suppression. Each scheduling service type is associated with a set of QoS parameters to specify its QoS requirements. Table 1 presents the characteristics of five different scheduling service types.

Although the service types and the associated QoS parameters have been well-defined in the IEEE 802.16 specifications, the resource management and scheduling mechanisms for 802.16e BS to guarantee QoS still remain an open issue.

**Table 1.** Characteristics of service types

Feature		UGS	rtPS	nrtPS	BE	ertPS
Request size		Fixed	Variable	Variable	Variable	Fixed but changeable
QoS parameters	Min. rate	N	Y	Y	N	Y
	Max. rate	Y	Y	Y	Y	Y
	Latency	Y	Y	N	N	Y
Application		VoIP without silence expression	Video, VoIP with silence expression	FTP, Web browsing	E-mail	Video, VoIP with silence expression

Several research works [3, 4, 5, 6, 7] were proposed using complex scheduling algorithm, such as weighted fair queue and worst-case weighted fair queue, and even a hierarchical schedulers. These scheduling algorithms are neither simple in implementation nor concerning the MCS. Besides, it is a difficult task to use a hierarchical scheduler when the frame duration is very brief in time. Thus, we propose a one level scheduling solution considering the QoS parameters defined in 802.16e standard, the QoS requirements of each connection (such as latency requirement, the maximum sustained traffic rate, and the minimum reserved traffic rate), and the adaptive MCS. The proposed scheduling solution contains admission control, scheduling and bandwidth allocation mechanisms. Furthermore we take latency parameter into account for real-time service flow to calculate the deadline of packets.

The proposed scheduling solution possesses several superior characteristics, including modulation and latency aware, implementation feasibility in BS, guarantee of minimum QoS for certain services (UGS, rtPS, nrtPS), and allocating bandwidth efficiently.

The simulation results indicate that the proposed scheduling algorithm can improve the throughput of the entire network and guarantee the minimum QoS requirements to each admitted connection.

## 2. The Proposed Scheduling solution

The scheduling solution we proposed concerns only the scheduler at the BS that uses TDD duplex in PMP mode. It contains three primary schemes: admission control policy, scheduling algorithm, and bandwidth allocation scheme.

### 2.1 Admission control scheme

To guarantee the minimum QoS requirement for different kinds of service connections, the admission condition of the modulation aware ACS is calculated based on the minimum reserved rate for each admitted data connection. In the TDD node, the transmission unit in WiMAX system is

symbol whose capacity depends on the MCS. Therefore, when a new frame starts, the required data size/bandwidth request size in downlink/uplink must be translated first to amount of symbol as shown in expression (1):

$$number\_of\_symbols = \frac{data\_size}{bits\_per\_symbol} \quad (1)$$

An OFDM symbol contains 192 data sub-carriers and 8 pilot sub-carriers in WiMAX PHY and the MCS decides the number of bits carried in a sub-carrier. Hence the number of bits per symbol can be calculated as

$$bits\_per\_symbol = 192 \times efficiency \times coding\_rate - 8 \quad (2)$$

Where the efficiency and the associated bits\_per\_symbol for different kinds of MCSs is presented in Table 2.

**Table 2.** The efficiency and the associated bits\_per\_symbol for different MCS

MCS	Efficiency	Bits_per_symbol
OFDM_BPSK_1_2	1 bps/Hz	88
OFDM_QPSK_1_2	2 bps/Hz	184
OFDM_QPSK_3_4	2 bps/Hz	280
OFDM_16QAM_1_2	4 bps/Hz	376
OFDM_16QAM_3_4	4 bps/Hz	578
OFDM_64QAM_2_3	6 bps/Hz	760
OFDM_64QAM_3_4	6 bps/Hz	856

The average required symbols per frame for the  $i^{\text{th}}$  connection  $T(i)$  based on minimum reserved rate and MCS can be calculated as

$$T(i) = \frac{BW\_min(i)}{bits\_per\_symbol(i) \times FPS} \quad (3)$$

Where the  $BW\_min(i)$  is the minimum reserved rate of the  $i^{\text{th}}$  connection, and the  $bits\_per\_symbol(i)$  stands for the bits per symbol in the  $i^{\text{th}}$  connection.  $FPS$  means the frames per second, and it can be derived as

$$FPS = \frac{1}{frame\_duration} \quad (4)$$

A connection which wants to access the WiMAX network will be allowed when the sum of  $T(i)$  is smaller than the available number of symbols of a WiMAX frame. The admission

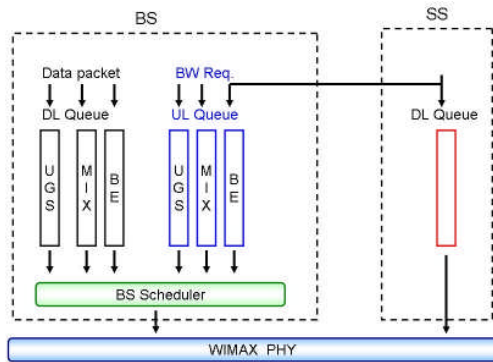
condition of proposed ACS is presented in expression (5).

$$\text{if } \sum_{n=1}^i T(n) < \text{available symbol in a frame} \quad (5)$$

The modulation-aware ACS can provide a precise scheme to calculate the number of SS which can be admitted. So, bandwidth allocation in scheduling will be more efficient.

## 2.2 Scheduling algorithm

Figure 1 presents the proposed scheduling architecture in WiMAX system. The proposed scheduler uses two directional queues, i.e., downlink queue which deals with incoming data packets from each downlink connection, and uplink virtual queue which deals with bandwidth requests corresponding to the uplink resource demands of SSs. Three kinds of queues were defined in the two directions, referred as UGS, MIX, and BE queue. The UGS queue serves the bandwidth requests and data packets for UGS connections. The MIX queue holds the request and data packets sent by rtPS and nrtPS connections. The BE queue stores the requests and packets of the BE connections. In the uplink, we also need to store periodic grants and unicast request opportunities for rtPS and nrtPS in the UGS queue.



**Figure 1.** Proposed scheduling architecture

The priority of downlink queue is higher than the uplink queue. Therefore, the scheduling order for all kinds of queues is as follows: first, UGS DL, UGS UL, DL rtPS packets which will expire in the next frame, UL rtPS bandwidth requests which will expire in the next frame, mix DL, mix UL, BE DL, and BE UL. The proposed scheduling order can adjust the DL/UL sub-frame ratio dynamically.

The deadline to each packet/request can be calculated through the arriving time of the packets/requests at the SS/BS queue. In the uplink, the BS can not get this information, and the worst

case is considered. This corresponds to the arrival at the queue immediately after the connection sent the last bandwidth request [7]. Thus, in the uplink case, the deadline of an rtPS request is equal to the arriving time of the last request sent by the connection plus its maximum latency. In the downlink case, the deadline of a data packet is equal to arrival time of the data packet plus its maximum latency for the connection. For each packet/request  $i$  to rtPS connection in the MIX queue,  $1 \leq \text{check\_deadline}[i] < 2$  means packet/request  $i$  will expire in the next frame. The  $\text{check\_deadline}[i]$  can be derived as

$$\text{check\_deadline}[i] = \frac{\text{deadline}[i] - \text{now}}{\text{frame\_duration}} \quad (6)$$

Where  $\text{now}$  means the current time, and  $\text{deadline}[i]$  presents the deadline for packet/request  $i$ .

To guarantee the minimum bandwidth requirement for rtPS and nrtPS connections in the MIX queue. The BS scheduler sorts the packets/requests in the MIX queue based on the granted bandwidth to each rtPS/nrtPS connection. After dealing with the UGS and MIX queues, the remaining bandwidth will be assigned to the BE queue. Figure 2 represents a simple flow chart for proposed scheduling algorithm.

## 2.3 Bandwidth allocation scheme

The scheduling algorithm decides the order of different traffics, and the bandwidth allocation scheme calculates the bandwidth allocated for each connection.

### 2.3.1 UGS queue

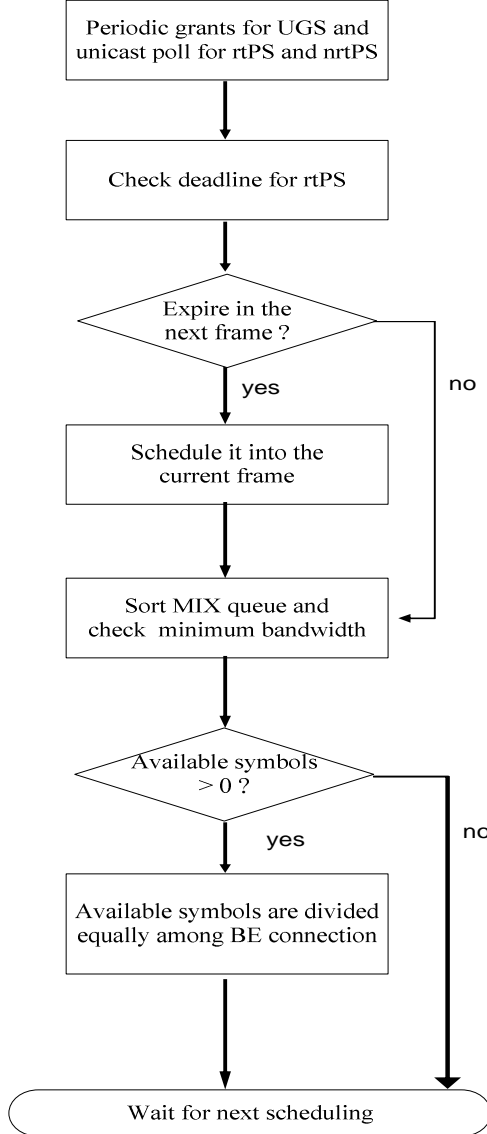
The bandwidth allocation to the  $i^{\text{th}}$  connection for UGS downlink is presented as expression (7).

$$\text{allocated\_symbol}(i) = \frac{\min\{Q_{\text{size}}(i), \text{BW\_min}(i) - \text{granted\_bit}(i)\}}{\text{bits\_per\_symbol}(i)} \quad (7)$$

Where  $Q_{\text{size}}(i)$  stands for the queue size in unit of bit to the  $i^{\text{th}}$  connection,  $\text{granted\_bit}(i)$  stands for the number of the allocated bits to the  $i^{\text{th}}$  connection.

The bandwidth allocation to the  $i^{\text{th}}$  connection for UGS uplink is derived as expression (8).

$$\text{allocated\_symbol}(i) = \left( \frac{\text{grant\_interval}}{\text{frame\_duration}} \right) \times T(i) \quad (8)$$



**Figure 2.** Proposed scheduling algorithm

### 2.3.2 rtPS queue

The bandwidth allocation to the  $i^{\text{th}}$  connection for this class downlink and uplink are presented as expression (9) and (10) respectively.

$$\text{if } (granted\_bit(i) < BW\_min(i))$$

$$allocated\_symbol(i) = \frac{\min\{\sum packet\_size(i), BW\_min(i) - granted\_bit(i)\}}{bits\_per\_symbol(i)} \quad (9)$$

$$\text{if } (granted\_bit(i) < BW\_min(i))$$

$$allocated\_symbol(i) = \frac{\min\{\sum BW\_req(i), BW\_min(i) - granted\_bit(i)\}}{bits\_per\_symbol(i)} \quad (10)$$

### 2.3.3 MIX queue

The bandwidth allocation to the  $i^{\text{th}}$  connection for the MIX queue downlink and uplink are presented as expression (11) and (12) respectively.

$$\text{if } (granted\_bit(i) < BW\_min(i))$$

$$allocated\_symbol(i) = \frac{\min\{\sum Qsize(i), BW\_min(i) - granted\_bit(i)\}}{bits\_per\_symbol(i)} \quad (11)$$

$$\text{if } (granted\_bit(i) < BW\_min(i))$$

$$allocated\_symbol(i) = \frac{\min\{\sum BW\_req(i), BW\_min(i) - granted\_bit(i)\}}{bits\_per\_symbol(i)} \quad (12)$$

### 2.3.4 BE queue

The bandwidth allocation to the  $i^{\text{th}}$  connection for BE downlink is presented as expression (13).

$$allocated\_symbol(i) = \min\left\{\frac{available\_symbols}{n}, \frac{Qsize(i)}{bits\_per\_symbol(i)}\right\} \quad (13)$$

The bandwidth allocation to the  $i^{\text{th}}$  connection for BE uplink is derived as expression (14).

$$allocated\_symbol(i) = \min\left\{\frac{available\_symbols}{n}, \frac{BW\_req(i)}{bits\_per\_symbol(i)}\right\} \quad (14)$$

Where  $n$  is the number of BE connections in which the allocated bandwidth is less than the maximum sustained rate.

## 3. Simulation and Numerical Results

We use NS-2 (version 2.31) simulation tool [8] with WiMAX NIST module [9] which is originally designed and developed by the National Institute of Standards and Technology (NIST) in U.S.A to simulate the proposed scheme.

Each simulation scenario includes a wired node, a BS, and several SSs with different service flows. The wired node is connected to the BS with a link, whose bandwidth and delay are 1000 Mbps and 1ms, respectively. The bandwidth of this link is larger than the WiMAX network. Figure 3 depicts the network structure for the simulation. In each simulation scenario, the general parameters of the 802.16 network are the same, and they are presented in Table 3.

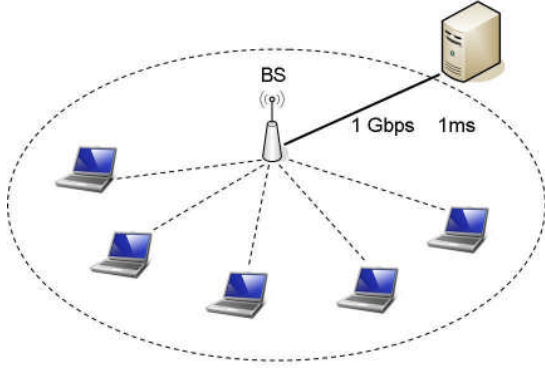


Figure 3. The network topology for simulation.

Table 3. Parameters of simulation scenario

Parameter	Value
PHY	OFDM
Duplexing mode	TDD
Frame duration	5ms
Frames per second	200
Available symbols per frame	70
DL/UL ratio	dynamic
Simulation start time (second)	10
Simulation end time (second)	30
Polling interval (rtPS) (ms)	20
Polling interval (nrtPS) (ms)	100

The UGS connection is simulated with the constant-rate data sent over the UDP protocol. According to the G.711 audio codec [10], we set the packet size to 160 bytes and the granted interval to 20ms. The rtPS connection is simulated with variable-rate data sent over the UDP protocol. We set the rtPS packet size to 1378 bytes, the maximum sustained rate to 410880 bps, the minimum reserved rate to 205440bps, and maximum latency to 50ms. The nrtPS and BE traffic are generated by FTP application. For nrtPS connections, we set the packet size to 1000 bytes and the maximum sustained rate to 200K bps, the minimum reserved rate for 150K bps. For BE connections, we set the packet size to 200 bytes, the maximum sustained rate to 120K bps, and the minimum reserved rate to 0 bps. Table 4 presents the traffic type and QoS requirements.

### 3.1 Simulation scenario 1

In this scenario, we analyze the proposed ACS, and general ACS which assumes all SS using the same MCS. We use several rtPS service

connections to test the throughput of ACS. Each SS has one rtPS connections to the BS. There are 20 64QAM\_3\_4 SSs, 15 16QAM\_1\_2 SSs, and 10 BPSK\_1\_2 SSs to randomly access the WiMAX network in our simulation scenario. The General\_BPSK\_1\_2 ACS means that the general ACS assume all SS using the BPSK\_1\_2 MCS. The meaning of the General\_16QAM\_1\_2 is similar to the General\_BPSK\_1\_2. The simulation result is shown in Table 5.

Table 4. Traffic type and QoS requirements

Service class	Traffic type	Packet size (byte)	Bandwidth (bps)	
			Max	Min
UGS	UDP/CBR	160	80000	64000
rtPS	UDP/VBR	1378	410880	205440
nrtPS	TCP/FTP	1000	200000	150000
BE	TCP/FTP	200	120000	0

Table 5. Simulation result of admission control

ACS	Admitted stations	Stations		
		64QAM_3_4	16QAM_1_2	BPSK_1_2
General_BPSK_1_2	5	3	1	1
Proposed	19	10	6	3
General_16QAM_1_2	25	12	8	5

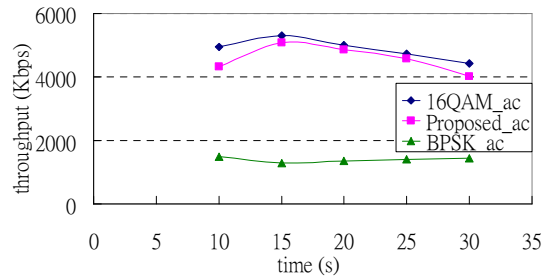


Figure 4. Total throughput of admission control

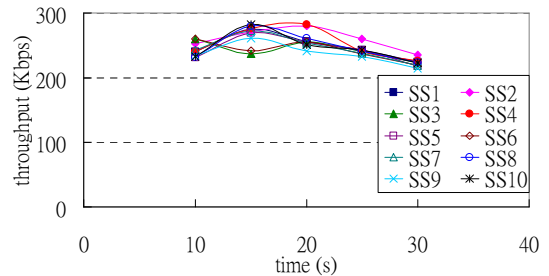
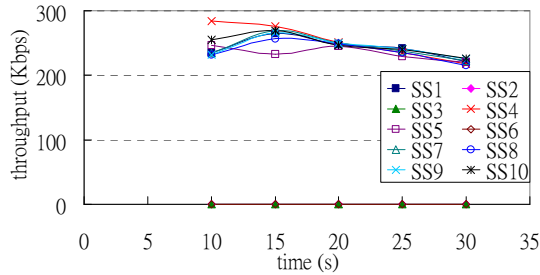
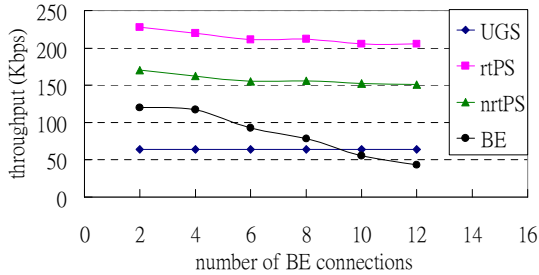


Figure 5. Throughput of each SS for the proposed scheme



**Figure 6.** Throughput of each SS for the general\_16QAM\_1\_2 scheme



**Figure 7.** Average throughput of four kinds of service connections

In Figure 4, we can observe the total throughput of general\_BPSK\_1\_2 ACS is about 1400 Kbps. The total throughput of the proposed ACS can achieve at least 4000 Kbps. The throughput of the general\_16QAM\_1\_2 ACS is higher than the proposed scheme slightly. However, the proposed scheme can guarantee the minimum reserved rate requirement for each admitted SS, but the general\_16QAM\_1\_2 scheme can not (see Figure 5 and 6).

### 3.2 Simulation scenario 2

The goal of scenario 2 is to investigate whether the increase of the BE service traffic load influences or not the QoS level of service which has higher priority. For this purpose, the simulated scenario includes one BS and 32 SSs. There are 4 UGS connections, 8 rtPS connections, 8 nrtPS connections, and the number of BE connections varies from 2 to 12. The UGS, rtPS, and nrtPS connections use 64QAM\_3\_4 MCS. The BE connections use 16QAM\_1\_2 MCS.

As we can observe in Figure 7, the average throughput of the rtPS and nrtPS connections decreased slightly as the BE connection increased. The average throughput of the UGS is always 64 Kbps. The proposed scheduling solution can guarantee the minimum reserved rate (64, 205.44, 150 Kbps) requirement to UGS, rtPS, nrtPS connections even though we increase the BE traffic load. Furthermore, we also guarantee the

maximum sustained rate (120 Kbps) requirement to the BE connections. Since the BE connection has no minimum reserved rate requirement, the average throughput decreased as the traffic load of whole system increased.

## 4. Conclusions

In this work, we take latency and MCS into account in our scheduling solution. The proposed scheduling solution contains three primary components: an admission control scheme, a scheduling algorithm, and a bandwidth allocation scheme. Some properties of our proposed scheduling solution are as follows: modulation and latency aware, being implemented in BS easily, guaranteed minimum QoS requirements for certain specific services (UGS, rtPS, nrtPS), and allocating bandwidth efficiently.

The simulation results show that the proposed admission control scheme can admit the suitable number of SSs to help the scheduler guarantee the minimum reserved rate to each admitted connection.

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