European Journal of Scientific Research
ISSN 1450-216X Vol.45 No.4 (2010), pp.529-539

http://www.eurojournals.com/ejsr.htm

A Cross Layer Approach for Packet Scheduling at Downlink of WiMAX IEEE802.16e

D. S. Shu'aibu
Faculty of Electrical Engineering, Universiti Technologi Malaysia Skudai Johor Bahru, Malaysia
E-mail: dahiru.75@gmail.com
Tel: +6075536584

S. K. Syed-Yusof
Faculty of Electrical Engineering, Universiti Technologi Malaysia Skudai Johor Bahru, Malaysia
E-mail: kamilah@fke.utm.my
Tel: +6075535749

N. Fisal
Faculty of Electrical Engineering, Universiti Technologi Malaysia Skudai Johor Bahru, Malaysia
E-mail: sheila@fke.utm.my
Tel: +6075535750

Abstract

WiMAX is one of the broadband wireless access networks with promising technology for providing better coverage and quality of service (QoS) to subscribers in the network. It supports different scheduling services with different QoS requirements the challenge here is how to efficiently scheduling the packets of subscribers having different QoS queuing in different buffers? We proposed a new algorithm capable of scheduling of all service flows type considering the nature of the wireless link, delay and buffer size. Performance of the proposed algorithm is evaluated by computer simulations it showed that the algorithm developed has achieved high throughput and good frame utilization.

Keywords: Scheduling, buffer size, delay, latency, QoS, service flow and throughputs

1. Introduction

In the recent technology of wireless communication, different service flows having different QoS can be served by a single service provider. WiMAX IEEE802.16 is one of the recent technology, that support five different scheduling service flows namely; unsolicited grant service (UGS), real time polling service (rtPs), extended real time poling service (ErtPs), non-real time polling service (nrtPs) and the best effort (BE) each of this, has its own QoS attached to it [1,2]. The standard does not specify any standard algorithm to be used to provide QoS of these service flows.

In WiMAX, the minimum resource that can be allocated to subscriber station (SS) is a slot. The definition of a slot depends on the type of permutation used. Four types of permutations are used in WiMAX; partial usage subchannelization (PUSC), full usage subchannelization FUSC, band AMC and tile usage subchannelization (TUSC) [1]. Bandwidth request by SS is either by unicast polling, multicast group polling piggyback or the request through ranging channel [1]. When the base station (BS) responded, the bandwidth can be granted by grant per subscriber station (GPSS) or by grant per connection (GPC). In GPSS, the BS grants a bandwidth on per subscriber basis such that, all its connection is treated as a block. Hence, there is need for a local scheduler at SS to control the service order and respect it's QoS. In GPC mode, the bandwidth is granted per every connection packet. The request of this kind is by piggyback. Communication BS and SS is in either point to multipoint (PMP) or in mesh mode.

Many research papers were presented on how to provide QoS in IEEE802.16 network [3-9]. In [3] bandwidth allocation in the downlink of mobile WiMAX was presented, the bandwidth was allocated by partitioning the link bandwidth in to three, this only guarantee the QoS in term of bandwidth, the architecture does not explain how the packets should be scheduled of the service flows after been admitted, GPSS mode was used in granting the bandwidth. In [4] scheduling architecture for the MAC protocol in IEEE802.16 was addressed. Both bandwidth allocation and scheduling were considered, the bandwidth allocation is dynamics whereas the scheduling is based on priority, GPSS mode was used but the performance of the architecture was not evaluated the authors did not present any simulation to support the architecture.

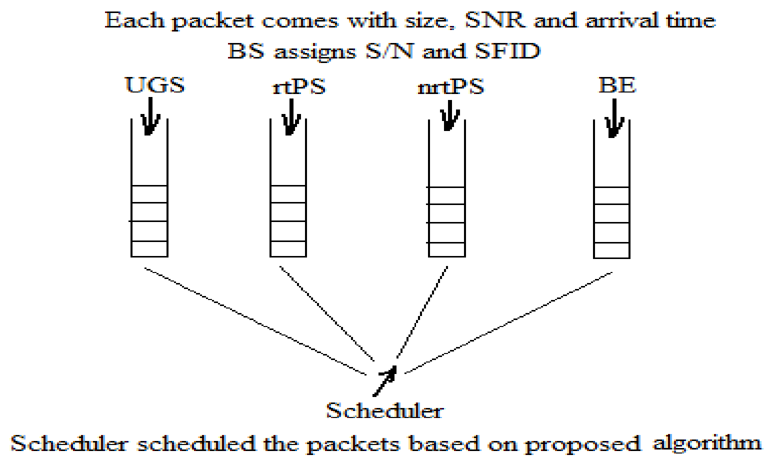
In [5] GPC mode for bandwidth allocation in the uplink scheduling architecture was used. Its centralized approach in point to multipoint mode also this architecture did not present any results to validate it. A simple bandwidth allocation with scheduling architecture was presented in [6], the packets were scheduled without considering any QoS parameters to improved the work in [6], [7] introduced packet delay prior to scheduling so as to reduce the packet loss of the algorithm in [6]. The proposed scheduling algorithm in [8], was based on queue information and packet delay for fixed data rate application (UGS).

In WiMAX, adaptive modulation and coding is employed, whereby the nature of the wireless link determines the type of modulation to be use [2]. However, none of the aforementioned algorithm has considered the nature of the wireless link. To explored the advantage of using wireless link in scheduling, the work in [8] was improved in [9] by estimated the SNR value which is used for choosing modulation before scheduling the packet. It was showed that, scheduling algorithm that considered wireless link (SNR) performed better then the algorithm without consideration of wireless link [9]. The algorithm developed in [9] is for real time users (rtPs), in TDMA mode for the uplink of IEEE802.16. In this paper, we present scheduling algorithms at the downlink of mobile WiMAX IEEE802.16e, in GPSS, TDMA/TDD mode, it considered both wireless link (SNR), delay and buffer size information for scheduling of four different service flows namely; UGS, rtPS, nrtPS and BE. The rest of the paper is organized as follows, Sections 2 deals with system description, Section 3 focuses on the propose scheduling algorithm, simulation results is presented in Section 4 and section 5 concludes the paper.

2. System Description

We consider IEEE802.16e in Point to Point mode, which uses centralized scheduling. Grant per subscriber station (GPSS) mode of bandwidth grant is used, details of bandwidth grant of the call admission control used in this paper can be found in [3]. We assumed that all the subscriber stations are uniformly distributed around the base station and transmit their packet during the uplink sub-frame. Each transmitted packet, arriving at the BS is given a serial number, service flow identifier (SFID), it comes with packet size, arrival time and signals to noise ratio (SNR). Figure 1 shows how the scheduler works.

Figure 1: Scheduler



It was shown in [10] that the optimal packet size in IEEE802.16e is limited to power of two values in the range of 16 to 1024 bytes. The packet size for rtPs, nrtPS and BE were limited to this range and each transmitted packet size is 2^N bytes where N ranges from 4 to 10. UGS has a fixed packet size.

The SNR received for each transmitted packet is a pseudo-random value drawn from a normal distribution with mean λ and standard deviation δ. The SNR value is estimated using equation1 given by [11] as explained in [3].

$$SNR = f\{p_t, A_n, B_n, p_l, \beta\} \tag{1}$$

Where p_t is the BS transmitted power, A_n is BS antenna gain, B_n is the receiver antenna gain p_l is the total path loss and β is the receiver sensitivity. The path loss used in this paper is Erceg model given in [1]. Based on the value of SNR in (1), the type of modulation can be chosen from table 1 given by [2].

Table 1: Receiver SNR in (dB)

S/N	Modulation	Coding rate	SNR (dB)
1	QPSK	½ ¾	5.0 8.0
2	16-QAM	½ ¾	10.5 14.0
3	64-QAM	½ 2/3 ¾	16.0 18.0 20.0

We used partial usage subchannel (PUSC) as the permutation of the frame, so each slot is one subchannel by two OFDMA symbols. The total data rate of a slot is given by equation

$$slot = f_d * N_s * C_r * \frac{\log(m)}{t_d} \tag{2}$$

Where f_d is the number of OFDMA symbol, N_s is the number of subcarriers per slot C_r is the coding rate m is modulated symbol and t_d is the frame duration in second. From equation 2, the total number of slots required by each packet can be evaluated. A packet with P bytes requires PS slots given in equation (3) as

$$PS = ceil(\frac{8 * P}{slot * t_d}) \tag{3}$$

By using equation (1), (2) and (3), each packet transmitted from subscriber station, can be scheduled to give the required QoS. The SNR value estimated at the physical layer is fed to the scheduler in the MAC layer which estimates the number of slots required by the packet and scheduled it.

3. Proposed Scheduling Algorithm

Four different buffers were used each for one service flow, as shown in Figure 1 each buffer has length *k*, each packet received in the uplink session, is stored in the buffer with its serial number, service flow identification, SNR, arrival time and the packet size. The scheduler visits each buffer during the downlink sub frame and scheduled the packet based on the proposed algorithm.

3.1. UGS Traffics

The UGS packets have fixed size, we use 256 bytes for each packet, since the packet may have different SNR value; they will require different number of slots for transmission. If UGS traffics has n Kbps, and the frame duration is t_d we have $1/t_d$ frames per second, for P byte packet, we have $J_s = floor(n*1000/8P)$ number of p byte packets to be transmitted per second. Each packet is scheduled at equal interval spaced by $1/J_s t_d$ the packet is scheduled first in first out when $y=0$ at frame number *(fn)*. Where y is given by $y = mod(fn, 1/J_s t_d)$.

3.2. nrtPs Traffics

None real time traffics have packets of different size, it required variable bit rate application the packet in nrtPs are scheduled based their delay and latency. The buffer is visited by the scheduler when the oldest packet in the buffer has reached j% of its latency. For a packet with the arrival time t_v and the latency t_l to be scheduled at time t_s, the condition $t_s - t_v = j*t_l$ should be satisfied and the packets are scheduled based on the earliest deadline first.

3.3. Best Effort Traffics (BE)

Best effort traffic does not have any QoS attached to them, but when transmitting packets of BE there should be minimum errors. In this buffer, packets are scheduled based on the information from the buffer size. Stack pointer is utilized very well in this service flows. The scheduler visits this buffer when the stack pointer of the buffer size reached 98.5% of the full capacity of the buffer size.

3.4. rtPs Traffics

Real time traffics are very much sensitive to delay with variable packet size. By default the algorithm always scheduled real time traffics by earliest deadline first manner when the scheduling conditions of the other traffics have not been met. The packets have variable size each comes with its own SNR.

The approximate algorithm is as follows;

```
For  fn=1: frame no         % scheduler algorithm
        While t <=  uplink duration
                All subscribers with packets transmit to BS with estimated SNR for each packet
                Assigns serial number, service flow identification and arrival time
                Stored  packets at the appropriate buffer of their corresponding service flows
          End while
         While t1 <= downlink duration
            Get the duration (m) of the oldest packet whose life time approach from the nrtPs buffer
            Get the position of the stack pointer (S) from BE buffer with buffer size d
            y=mod ( fn,  1/J_s t_d).
                If   y==0
```

```
                Schedule UGS
        Else if   m >= j*t_l
                Schedule nrtPs
        Else if   S=.985*d
                Schedule BE
        Else
                Schedule rtPs
      End if
    End while
End for
```

Scheduling Algorithm is how the slots are allocated to the packets on the frame

```
For  p = 1: no of packets                  %algorithm on each buffer
    For the received packets
    Evaluate the total slots required based on link condition
    If  frame = new   %(fresh frame no packet scheduled yet)
        Scheduled the packet
        Marked the used slots
        Next packet
    Else if frame = used % (already carrying some packets)
        If required slots <= Unmarked slots
            Schedule the packets
            Marked the used slots
            Next packet
        Else
            Wait for the next new frame
            If packet delay >= latency
                Drop the packets
                Next packets
            End if
        End if
End for
```

4. Simulation Results

The simulation parameters settings is shown in Table 2, during the uplink session all service flows with packet transmit, we assume that all the subscribers participating has been admitted as in [3]. Base station received all transmitted packets from subscriber stations, assigned packet serial number, packet service flow identification, arrival time and store the packet in appropriate buffer of the service flow. Each transmitted packet has its own estimated SNR value, details on the estimated SNR is given in [3].

During the downlink session, the scheduler in the BS scheduled the packets base on the proposed algorithm in section 3. Two parameters (packet size and SNR value) are used to allocate the required number of slots of each packet. If the required number of slots on the current frame is not enough to schedule the current packet, its delay to the next frame provided its latency will not be reached otherwise the packet is loss.

The buffers were used each for different service flow, each buffer can store 200 packet at a time, if the buffer is full and there is packet on the queue, that packet is loss because no memory to store it. Each packet scheduled, is removed from the buffer and the memory is declared empty so that it can be used to store the next packet. The uplink duration was set to 5.2ms and the downlink is 14.8ms, the simulation time was set to 20s and a total of 1000 frames were used in the scheduling. The simulation results were shown in the Figures below.

Table 2: Parameter settings

Parameter	Settings
FFT Size	512
Bandwidth	5MHz
DL/UL ratio	30:17
No. of Subchannels	15
No. of frames	1000
Frame duration	20ms
Slots/frame (downlink)	225
Permutation	PUSC
Buffer size	200 packets
UGS packets	256 byte
Packets for other services	16, 32, 64, 128, 256, 512 and 1024 byte
Modulation scheme	QPSK, 16-QAM, 64-QAM (different coding)
Simulation time	20s
Path loss model	Erceg
Transmission frequency	2.4GHz
Path loss exponents	4
Shadowing component	9
BS transmitting power	42 (dBm)
BS antenna gain	16 (dBi)
Receiver sensitivity	-109.1 (dBm)
Receiver antenna gain	-1.0 (dB)

Figure 2 shows the utilization of the frame, it can be seen that most of the slots are utilized each frame carries 225 slots. Figure 3 shows the total throughputs in kilobits as the simulation time varies. Figure 2 and Figure 3 are related since the no of slots and throughputs are related. Packet arriving and departure time is shown in Figure 4, it can be seen that rtPs and UGS traffics have the least arriving and departure as it should because they are delay sensitive. None real time (nrtPs) is the next priority after rtPs and UGS, the first packet of nrtPs was scheduled at about 780ms of the simulation time this is little bit more than 90% of the latency which is set at 800ms. The first best effort packet was scheduled after about 2 seconds or 200 frames of begin of the simulation as shown in Figure 4.

Figure 5 shows packets sizes and corresponding link condition of all the service flows. The varying nature of the packet sizes for the three service flows can be seen. The UGS traffics have fixed packet size but the link condition varies as shown in Figure 5 with green color. The effect of the link condition on the slots can be seen in Figure 6, the packet size is the same for UGS traffic (256 byte) but due to different link condition the slots required for each packet is different. The wireless link condition varies as the packet arrived for all service flows as shown in Figure 7. Total throughputs in Kilobits of each service flow as the simulation time varies can be seen in Figure 8 for the first 4 seconds, it can be seen that there is fair scheduling among the service flows.

Figure 2: No Slots Used on Each Frame

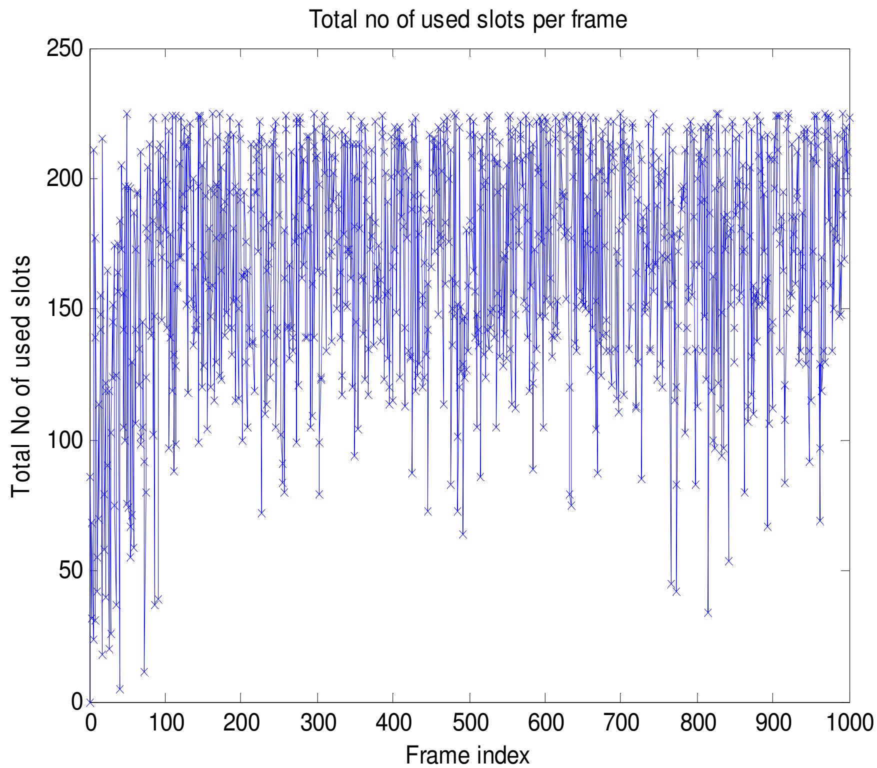


Figure 3: Total system throughputs per Simulation time

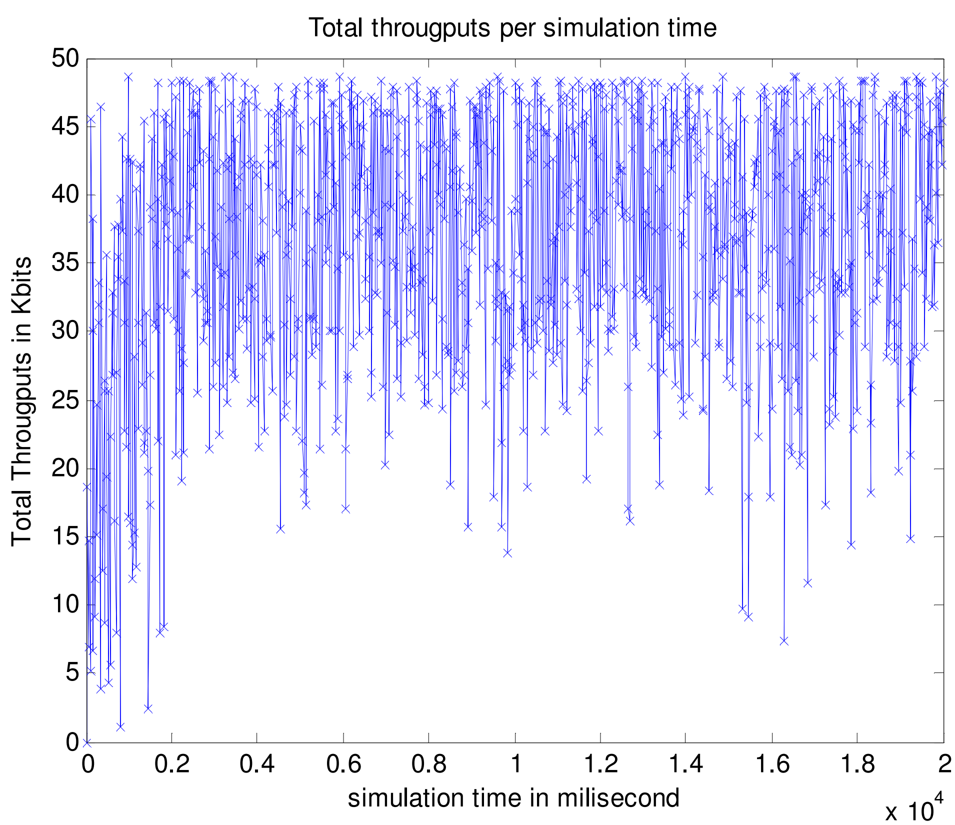


Figure 4: Arrival time and Departure time for all the service flows

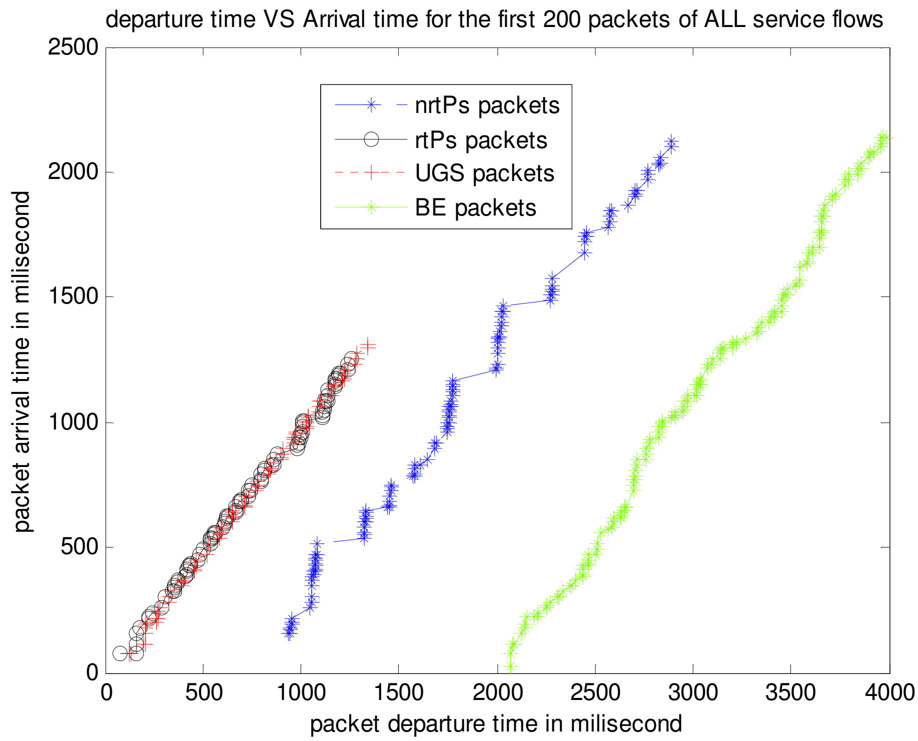


Figure 5: Packet size and Corresponding SNR value for all service flow

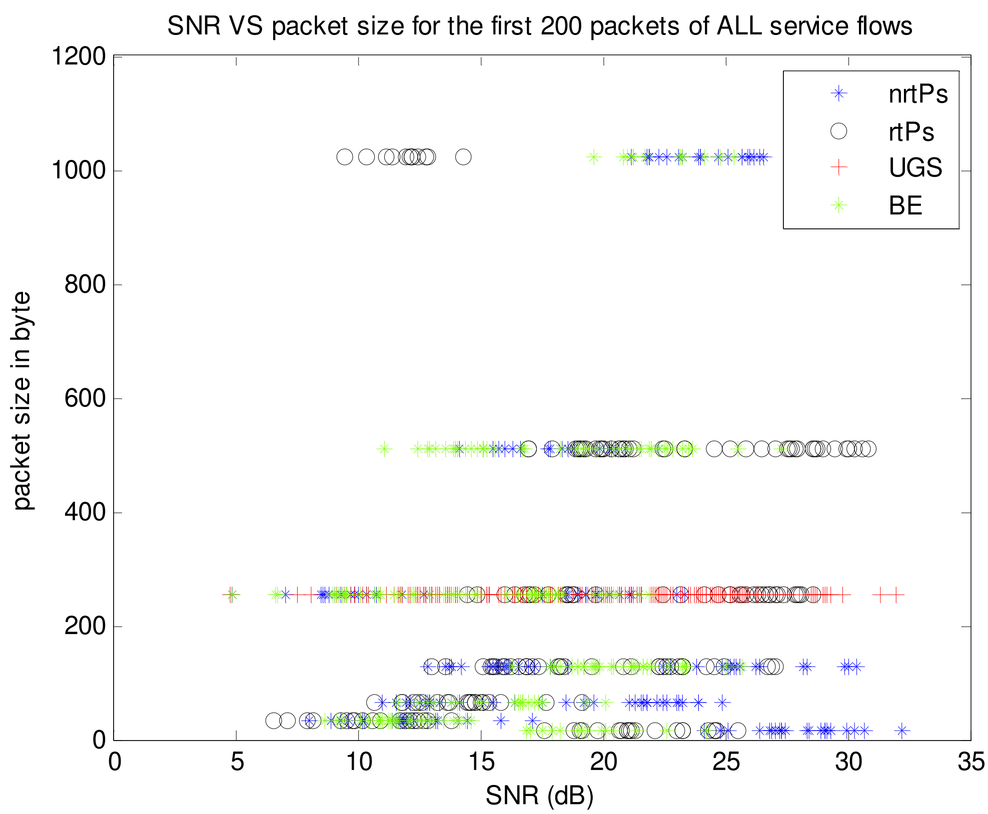


Figure 6: Total slots required based on link condition for UGS packets

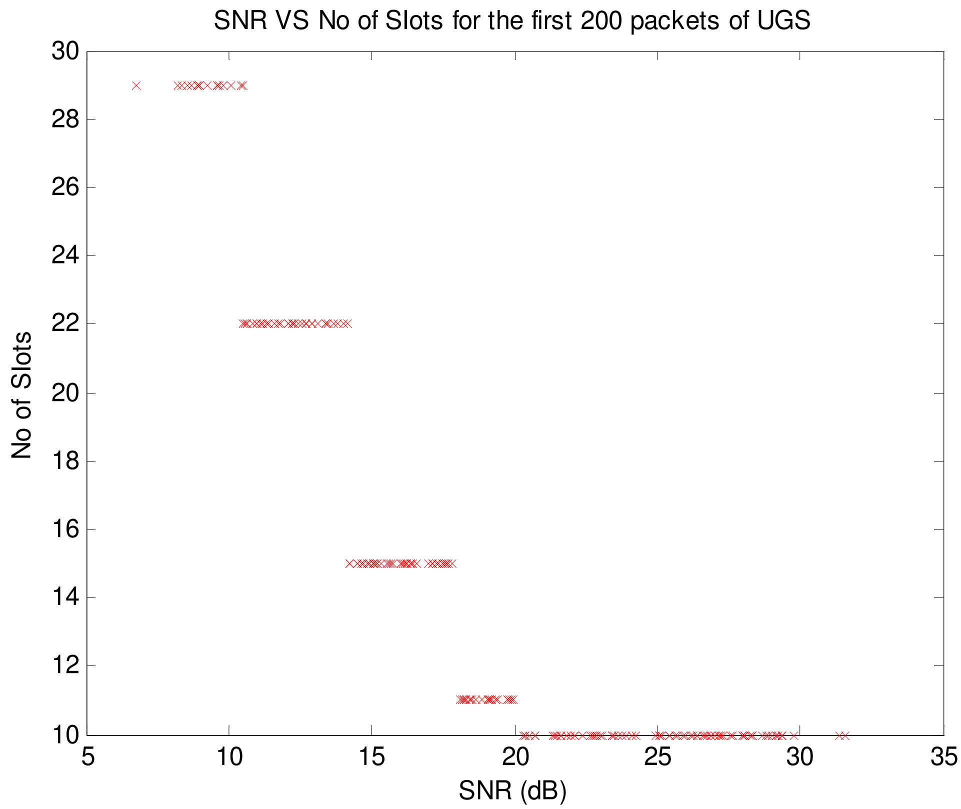


Figure 7: Link condition based on packet arrival time.

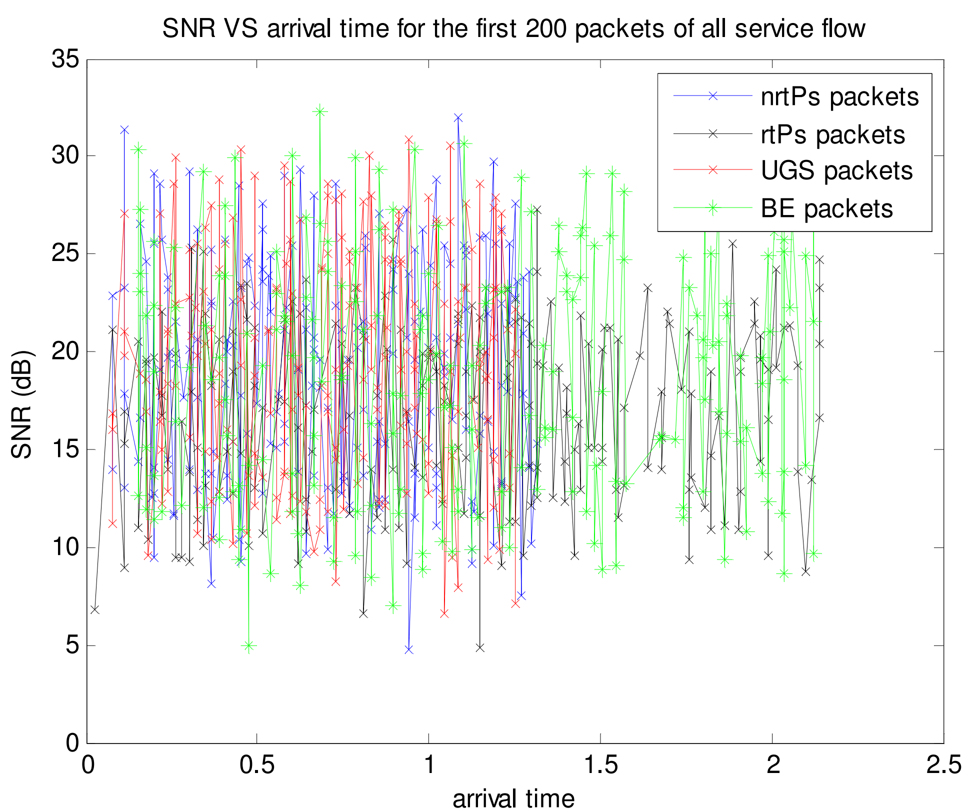
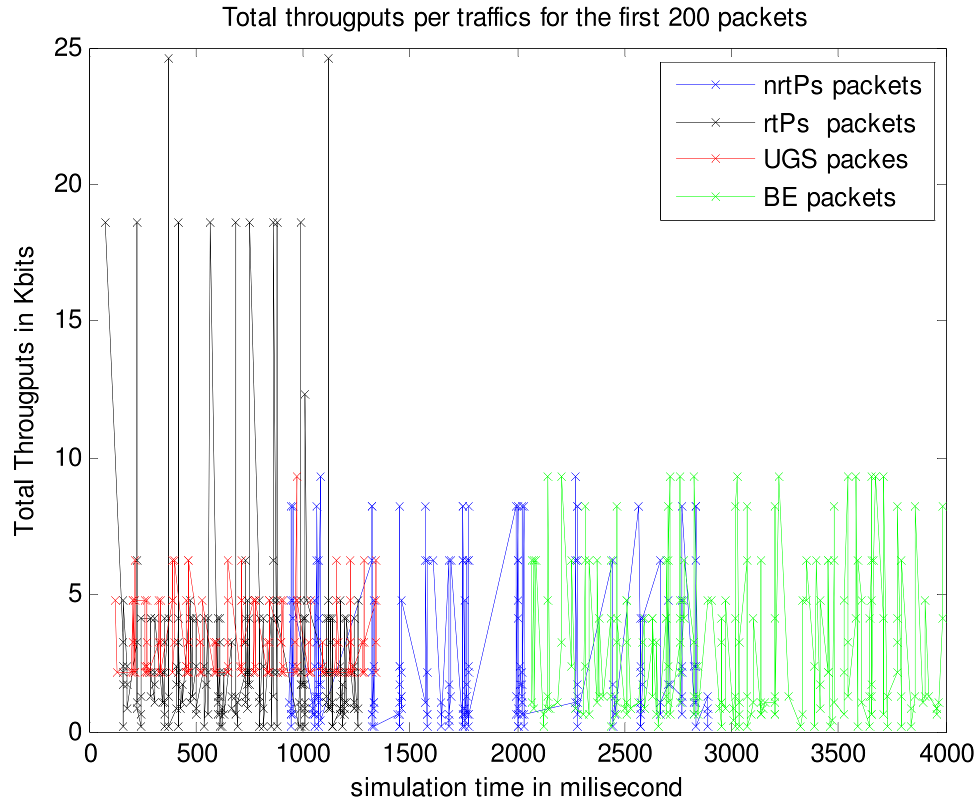


Figure 8: Total throughputs of all service flows



5. Conclusion

In this paper, we presented a cross layer approach to packet scheduling in Mobile WiMAX. We showed how the wireless link affects the throughputs and the frame utilization. Since WiMAX uses adaptive modulation and coding, the wireless link should be considered in the scheduling part, the algorithm can be suited to be implemented at the base station. The simulation showed a fair scheduling among the service flows despite the use of the wireless link condition in scheduling.

Acknowledgment

The authors would like to thank all those who contributed toward making this research successful. Also, we would like to thank to all the reviewers for their insightful comment. This work was sponsored by the Research Management Unit, Universiti Teknologi Malaysia.

References

[1] WiMAX Forum, Krishna Ramadas and Raj Jain, "WiMAX System Evaluation Methodology" version 2.1 July 2008

[2] IEEE Std. 802.16e, "IEEE Standard for local and metropolitan area networks, part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems, Amendment 2: Physical and Medium Access Control Layers for Combined Fixed and Mobile Operation in Licensed Band and Corrigendum 1", May 2005

[3] D. S. Shu'aibu, S. K. Syed-Yusof and N. Fisal "Partition-Base Bandwidth Managements for mobile WiMAX IEEE802.16e, " International Review on Computers and Software (IReCoS) Vol 5 no 4 July 2010

[4] G. Chu, D. Wang, and S. Mei, "A QoS Architecture for the MAC protocol of IEEE802.16 BWA System," IEEE International Conference on Communications, Circuits and Systems and West Sino Expositions, June 2002.

[5] M. Hawa and D. Petr, "Quality of Service Scheduling in Cable and Broadband Wireless Access Systems," Tenth International Workshop on Quality of Service, May 2002.

[6] JianfengChen, W. Jiao, and H. Wang, "A Service Flow Management Strategy for IEEE802.16 Broadband Wireless Access Systems in TDD Mode," IEEE, 2005.

[7] Sarat Chandra and Anirudha Sahoo "An Efficient Call Admission Control for IEEE802.16 Networks Proc of the 15th IEEE Workshop on Local and Metropolitan Area Networks, Princeton, New jersey, pp 188-193 June 2007.

[8] J.H. Jeon and J. T. Lim, "Dynamic bandwidth allocation for QoS in IEEE802.16 broadband wireless access network," IEICE Trans Communication, Vol.E91-B, no. 8 pp 2707-2710, August 2008.

[9] Hyun-Wook JO, Jae-Han, JEON and Jong-Tae LIM "Bandwidth allocation for QoS Using Adaptive Modulation and coding in IEEE802.16 networks," IEICE Trans Communication, Vol.E92-B, no. 12 pp 3919-3923, December 2009.

[10] Henrik Martikainen, Alexander Sayenko, Oilli Alanen and Vitality Tykhomyrov "Optimal Mac PDU Size in IEEE802.16", IT-NEWS pg 66-71 2008.

[11] Bernard Sklar 'Digital Communications Fundementals and Application second edition, Prentice-Hall International Inc Upper Saddle River, New Jersey USA 2001 pg 286-289.