Bandwidth Control Protocol in WiMAX Network

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Abstract—In WiMAX network, service flows with different priorities are defined. The priority of real time traffic is higher than that of non real time traffic. As bandwidth is limited, the resource must be allocated in advance to guarantee the delay time of real time traffic in WiMAX network. For this, the proposed algorithm must respond quickly to the sudden shortage of resource.

In this paper, we will propose a Bandwidth Control Protocol (BCP). This algorithm evolves from the traditional bandwidth allocation protocol. When the bandwidth assigned to the real time traffic in advance is insufficient, it should be increased in following frames. With this, the bandwidth allocated to the non real time traffic must be reduced accordingly. In this way, the assigned bandwidth can be adapted to the real load of real time traffic. Also, the belonging bandwidth for non real time traffic can be borrowed by real time traffic if the load of real time traffic is fluctuated. The borrowed bandwidth will not be returned unless the allocated resource to the real time traffic is enough. With this mechanism, the delay time of real time traffic will be reduced effectively. Also, the bandwidth will be allocated more correctly.

Keyword: WiMAX, BCP (Bandwidth Control Protocol), Real time traffic, Delay time, bandwidth allocation

I. INTRODUCTION

Recently, the development of Broadband Wireless Access (BWA) Technology provides an easy, timesaving, and low-cost direction for the deployment of the next generation (4G) wireless communication. The demands of multimedia service in daily life are enhanced greatly. Video on demand, video streaming, and VOIP have become the most prevailing real time application for a lot of people. To lower the latency in the transmission of multimedia message, Quality of Service (QoS) is studied frequently in the WiMAX network. The novel BWA technology for Worldwide Interoperability for Microwave Access (WiMAX) is based on the IEEE 802.16 standard [1-3]. Several types of services can be integrated with this technology and each can have a good guarantee of QoS. Except this, WiMAX also has many important features like IP connectivity, larger geographic coverage, wider bandwidth, hgher mobility and secure communication. Protocols of PHY layer, MAC layer and QoS framework are specified in IEEE 802.16 standard. However, admission control and scheduling scheme are not defined in this standard. Therefore, the system developers have to design these mechanisms to fulfill the requirement of their subscribers.

The service classes in IEEE 802.16 are classified as UGS, ertPS, rtPS, nrtPS, and BE respectively [1-3] according to thier priorities. They are classified to satisfy their different types of QoS requirements. For this, the service class is specified with its QoS parameters. In the standard, the scheduling algorithm is not specified clearl. So, many research results have been published in this field for WiMAX network [7-15]. Throughout all the works, the author proposed his own scheduling algorithm to meet the QoS requirement and improve the system performance.

This paper focuses on the implementation of QoS control algorithm in WiMAX network. In multi-media application, a great volume of audio and video messages are sent with variable rate and they have stringent confinement on delay. These real-time data streams must have less latency and higher priorities in WiMAX network. In this paper, we will propose a scheduling algorithm named **Bandwidth Control Protocol (BCP)** in WiMAX network which evolves from traditional **Weighted Round Robin** (**WRR**) algorithm. The bandwidth assigned to the real time traffic is changed dynamically according to its queue length at the beginning of each frame. This is similar to the power control of mobile network in which the sending power is decided by the S/N ratio of the received signal to noise ratio. This is why we call our scheme as bandwidth control protocol. The object is to respond to the current traffic and assign proper bandwidth for queues of real time traffic and obtain the QoS requirement for these services. Simulation study of our algorithm is based on a popular simulator named NS-2 [4-6]. In the simulation, the results of proposed algorithm will be compared with that of traditional algorithm (i.e. **WRR**).

The remainder of the paper is organized as follow. In the Section II, we will present some background for the QoS problems in WiMAX network. The bandwidth control protocol will be described deeply in Section III. In Section IV, the simulation result and performance analysis will be presented and discussed. Finally, some remark about conclusion and future work are shown in Section V.

II. SOME BACKGROUND ABOUT WIMAX NETWORK

A. *TDD Mode for WiMAX SYSTEM*

The IEEE 802.16 standard supports both Point to Multipoint (PMP) and Mesh topologies. With PMP topology, subscriber stations (i.e., SSs) do not communicate with each other directly. The link between base station (i.e., BS) and SS are full duplex (i.e., downlink and uplink channels are separated). BS is the also the manager of the network and

978-0-7695-4190-7/10 $26.00 © 2010 IEEE
DOI 10.1109/ISPA.2010.102

responsible for the integration of the network resource. So, BS can control important functions such as connection set up, time slots allocation, and bandwidth assignment. With Mesh topology, SS can communicate with each other directly or relay packets for other SSs. In this paper, we will focus our study on the PMP mode. The IEEE 802.16 system supports both Frequency Division Duplexing (FDD) and Time Division Duplexing (TDD) mode. In Figure 1, the time slots allocation for a frame in TDD mode is shown. With this mode, the uplink and downlink links use the same frequency band and different time. The ratio for number of time slots in downlink and uplink can be adjusted dynamically in this mode. Those service classes like UGS, ertPS and rtPS have with higher priority than the non-real time traffic like nrtPS and BE. So far, most IEEE 802.16 networks are operated in TDD mode and with a PMP topology.

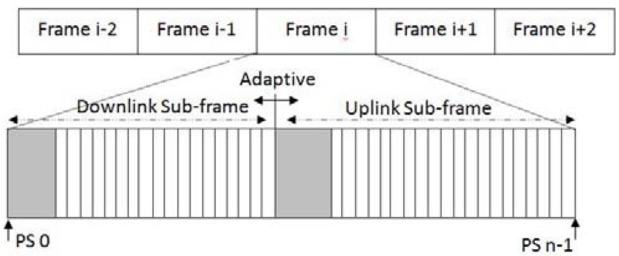

Fig.1. the allocation method for TDD Frame

B. The Framework of Scheduling for WiMAX

As shown in Figure2, the IEEE 802.16 standard uses request-granted mechanism to supply connection-oriented service. When an SS joins a BS in the WiMAX network and asks for a new connection, the BS will assign this connection a unique connection ID (CID) for the uplink or downlink connection. This approach can be explained by the signaling process between the BS and SS in Figure 2. The connection can be created, changed, and deleted by the issuing Dynamic Service Addition (DSA), Dynamic Service Change (DSC), and Dynamic Service Deletion (DSD) frames. Each process uses a two or three ways handshake approach and can be initiated by the BS or SS. In the standard of IEEE 802.16, the method for assigning the bandwidth of each service flow is not specified. And, there are a lot of researches on this part of the admission control [7-10].

As we can see in Figure 2, when a service flow want to generate or change its service, it will send DSA or DSC frame to the BS. From the application layer, numbers of service traffic (ex: T1/E1, VOIP, Video, FTP, HTTP, E-mail, etc.) can be passed to the MAC layer. In the MAC layer, the packets can be classified into different types according to their QoS specification and assigned to suitable queues in the BS. In the queue, these packets are scheduled with designed algorithm efficiently. The result of scheduling will be used to generate the UL_MAP and DL_MAP message.

The SS sends or receives the frames according to the direction shown in the UL_MAP/DL_MAP message. These messages will appear in the fixed time slots of the cyclic super frames. So, the MAP Generating Module will decide the partition of bandwidth according the BW_REQ messages from the SS.

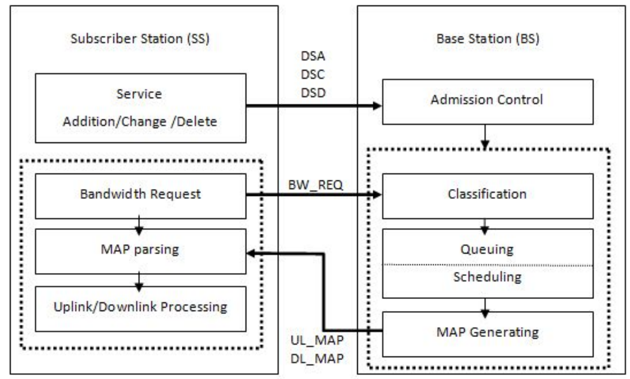

Fig.2. Scheduling Framework for BS and SS

C. Related Scheduling Algorithms for WiMAX

In recent years, the QoS control in the WiMAX system is a very popular issue. Although the related QoS parameters have been defined in IEEE 802.16 d/e, the scheduling algorithms have been remained opened. For this, several important researches have been done and presented [1-3].

In the first research [11], Jianfeng Chen, Wenhua Jiao,and Hongxi Wang design **Deficit Fair Priority Queue (DFPQ)** algorithm in the first stage of scheduling. This algorithm comes from DWRR [16]. In DFPQ, the bandwidth quota for each service queue is assigned efficiently to achieve fairness. The authors use hybrid algorithm (EDF+WFQ+RR) from [10] in the second stage. In this stage, the service class of UGS is assigned with fixed bandwidth and does not participate into the contention of bandwidth among queues. Early deadline first (EDF) algorithm is used for the rtPS service class, Weight Fair Queuing (WFQ) is used for the class of nrtPS, and Round Robin (RR) is used for BE class in this stage. Therefore, the starvation problem encountered in [10] can be solved with this two stages scheduling algorithm.

There are five types of service classes supported in IEEE802.16e. In the standards, the corresponding QoS parameters are also provided. However, the standard is only defined up to layer 2. In the second research [12], the authors propose a cross layers QoS framework for IEEE802.16 network. In this framework, the IP scheduling (layer3) is integrated with IEEE802.16 scheduling (layer2). There are two important functions are implemented here. For the function of fragment control, all fragments of the same packet in layer 3 will be treated as a group in layer 2. In this way, useless transmission and packet delay can be reduced effectively. For the function of remapping, the service types of layer 3 are mapped to the service type of layer 2. With this, the QoS mapping of rtPS and nrtPS traffics can be changed quickly when the network is jammed.

In the third research [13], **Maximum Signal-to-interference Ratio (mSIR)** scheduler was introduced. This scheduler assigns the bandwidth to SS according to Signal-to-Interference Ratio (SIR) value of its link. If the SIR value of the link is too small, most of the transmission will be failed. In this case, the scheduler will assign less bandwidth to it. If the SS moves to a different position and the SIR value of the link is improved, the scheduler will assign more bandwidth to it to compensate its foregoing loss.

In the fourth research [14], a hierarchical structure scheduler is proposed. At the first stage, the rtPS packets are scheduled with Earliest Deadline First (EDF) in their queues. The nrtPS packets are scheduled with Weighted Round Robin in their queues. And, the Best Effort packets are scheduled with Round Robin. At the second stage, all packets are scheduled according to their priority and the rtPS packets are classified as urgent packets and non-urgent packets according to their deadline of arriving. Then, Later Deadline Preemption (LDP) algorithm is applied and non-urgent rtPS packets could be preempted by other lower priority packets..

In the fifth research [15], a QoS-supported uplink packet scheduler and call admission control (CAC) mechanisms are proposed. The uplink traffics are regulated by the BS with the algorithm of token bucket. Since the output rate of all uplink links are regulated by token bucket, the call admission control can be applied easily.

III. BANDWIDTH CONTROL PROTOCOL

A. *Connection Admission Control Mechanism for WiMAX*

The call admission Control (CAC) mechanism determines whether a new request for connection can be granted or not according to the remaining bandwidth. The admission control method is not defined in the standard and is left as an opened choice for the manufacturers [14]. With this mechanism, the BS will assign a Minimum reserved rate ($\gamma_{\min}$) or maximum sustained rate ($\gamma_{\max}$) to each link. In our paper, a maximum sustained rate $B_{service-flow}$ will be given to each link.

In IEEE 802.16e, 5 types of service classes are defined. For UGS class, a fixed bandwidth is allocated in advance. For the rest classes, the bandwidth are allocated with the priority order of ertPS, rtPS, nrtPS and BE. The procedure of CAC will be described as follows:

1. Let's assume B_{total} is the total assignable bandwidth in one frame and U is an upper bound of bandwidth for all the UGS classes. A new connection request of UGS will be accepted only when the total assigned bandwidth to all services of this class will be smaller than or equal to U.
2. Let B_C be the assigned bandwidth to all classes except UGS until now. Assume a new connection request of ertPS class is arisen and ask for a maximum sustained rate of B_{ertPS} . It will not be accepted unless $B_{ertPS} + B_C \le B_{total} - U$.
3. Assume a new connection request of rtPS class is arisen and ask for a maximum sustained rate of B_{rtPS} . It will not be accepted unless $B_{rtPS} + B_C \le B_{total} - U$.
4. Assume a new connection request of nrtPS class is arisen and ask for a maximum sustained rate of B_{nrtPS} . It will not be accepted unless $B_{nrtPS} + B_C \le B_{total} - U$.
5. Assume a new connection request of BE class is arisen. It will be accepted. However, it cannot be guaranteed that enough bandwidth will be given to it.

B. *The Allocation of Basic Bandwidth for WiMAX*

With TDD mode, the bandwidth is allocated for each frame. The allocated bandwidth for each link must satisfy the CAC mechanism. Assume Q_i is the amount of data waiting for transmission, B_i is the maximum sustained rate, and R_i is allocated bandwidth in the i^{th} service flow queue at the beginning of a frame. Then,

$$\begin{cases} Q_i \le B_i\,, & R_i = Q_i \ldots(1) \\ Q_i > B_i\,, & R_i = B_i \ldots(2) \end{cases}$$

Let B_{slot} be the bandwidth of each slot (Let's assume the unit of all bandwidths considered here is byte. The value of B_{slot} will be decided by the modulation method used in this slot. However, we will assume this value is fixed here to simplify the discussion) Then, the assignable bandwidth will be $B_{Frame} = N \times B_{slot}$ at the beginning of one frame, where N is the number of slots in the uplink part of one frame. After each scheduling, the assignable bandwidth can be obtained with

$$B_{Frame} = B_{Frame} - R_i \ldots(3)$$

The basic frame of classifier is shown in Figure3. The CAC in the BS will determine whether the request of this queue is approved or not. After that, the frames are classified into each queue based on its CID and service type. Then, the scheduler will send the frames in each queue until the left assigned bandwidth is not enough for the next frame.

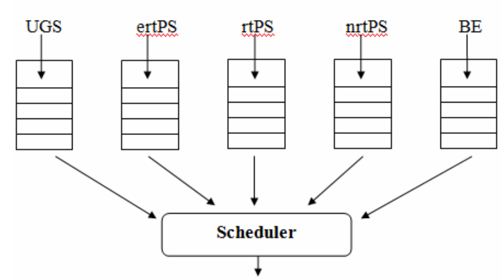

Fig.3. The basic frame classifier for IEEE 802.16e

Since some queues may not exhaust their maximum sustained rates and the queue of BE may not finish up the left resource, the scheduler can continue allocating the left bandwidth to those nonempty queues.

With equation (3), we can find the assignable bandwidth after the scheduling of the i^{th} queue. If all the queues have been scheduled and $B_{Frame} \neq 0$, the residue bandwidth B_{Free} can be defined as the left bandwidth B_{Frame}. At this time, we can use bandwidth control protocol to distribute B_{Free} to all queues.

Because UGS class has been assigned with fixed and needed bandwidth at this time, these queues will not contend for B_{Free}. As most of the foregoing researches, the scheduler in this paper will use weighted round robin (WRR) to distribute B_{Free} to the other four classes. As the power control method, the initial sending power is based on coarse open power control and may not be adequate. Further close loop power control is needed for most mobile systems. For bandwidth assignment, the initial estimation of maximum sustained rate may not be precise and some sort of close loop adjustment may be needed. Therefore, the maximum sustained rate of each class can be changed according the result of the first round of scheduling. In our paper, the maximum sustained rates of rtPS and BE classes will be changed to demonstrate the effect of this protocol. However, the protocol can be extended to include the other classes if it is necessary.

C. *Bandwidth Control (BC) Protocol*

In the first round of scheduling, the bandwidth is assigned according the priority of service class. The QoS of higher priority classes will be satisfied if the maximum sustained rates are estimated correctly. However, the future may not always be completely predictable. The sending power may need to be changed as MS moves. The basic requirement of bandwidth for each class may also need to be changed as time passes by. Since the situation may change, the QoS requirement of real time traffic may not be satisfied from time to time and many frames may be discarded because of time delay. At this time, some sort of auto adjustment for maximum sustained rate may be needed. Without this mechanism, the maximum sustained rate may have to be over-estimated to satisfy possible short-term load fluctuation. Since the whole bandwidth is fixed, the increasing of bandwidth assignment for some class implies the decreasing of the other class. This is an important factor for CAC to operate correctly. When the short-term effect disappears, the "borrowed" bandwidth must be returned. For this reason, the bandwidth control protocol is also called as adaptive bandwidth-borrowing algorithm (ABB) by the authors in the beginning. Although ABB may also be a proper name to understand the contents of the protocol, the increasing bandwidth may never be returned if the initial estimation of maximum sustained rate is not correct.

After the first round of bandwidth assignment, the frames of rtPS class will be assigned some slots to send. If the queue of this class is not empty at this time, their frames may be delayed and discarded without the second round assignment (WRR assignment). This means the maximum sustained rate is not enough. If this situation is happened for three times continuously, the assigned maximum sustained rate is not enough with very high possibility. Therefore, the maximum sustained rate will be increased before the start of next frame in BP protocol. Since three continuous times of insufficiency may not respond quickly enough, we also study the case the response is done for each time of insufficiency. To make the CAC function correctly, the maximum sustained rate of BE class must be decreased by the same amount in the protocol.

Two types of Bandwidth Control protocols are proposed here. They are described as follows:

Method 1: BC -General

Method 2: BC-Reactive

Some parameters used in BC protocol are listed in Table1:

Table1. Parameters Setting

Parameter	Description
Δ	The bandwidth getting from BE class when the maximum sustained rate of rtPS class is not enough
α	The bandwidth returning to BE class when the maximum sustained rate of rtPS class is enough
N3H	The number of continuous times that the maximum sustained is not enough
N3L	The number of continuous times that the maximum sustained is enough
$B_{rtPS-MAX}$	The upper bound for the maximum sustained rate of rtPS class

Method 1: BP-General

Let Q_{rtPS} and B_{rtPS} be the queue length and the maximum sustained rate of rtPS class.

- When $Q_{rtPS} \le B_{rtPS}$, the queue will be empted after the first round of scheduling. Then, N3L will be added by one. Otherwise, N3L will be returned to zero. This represents the case that the maximum sustained rate is enough for rtPS class.
- When $Q_{rtPS} > B_{rtPS}$, the queue will not be empted after the first round of scheduling. Then, N3H will be added by one. Otherwise, N3H will be returned to zero. This represents the case that the maximum sustained rate is enough for rtPS class.
- If N3L=3, $B_{rtPS} = B_{rtPS} - \alpha$. If B_{rtPS} is smaller than its initial value, let B_{rtPS} be its initial value. Also, the maximum sustained rate of BE class is increased by the same amount. Finally, let N3L=0. This situation means there are three continuous times the maximum sustained of rtPS class is enough.
- If N3H=3, $B_{rtPS} = B_{rtPS} + \Delta$. If B_{rtPS} is larger than $B_{rtPS-MAX}$, let $B_{rtPS} = B_{rtPS-MAX}$. Also, the maximum sustained rate of BE class is decreased by

the same amount. Finally, let N3H=0. This situation means there are three continuous times the maximum sustained of rtPS class is not enough.

Method 2: BP-Reactive

- If $Q_{rtPS} \leq B_{rtPS}$, $B_{rtPS} = B_{rtPS} - \alpha$. If B_{rtPS} is smaller than its initial value, let B_{rtPS} be its initial value. Also, the maximum sustained rate of BE class is increased by the same amount. This situation means the maximum sustained of rtPS class is enough.
- If $Q_{rtPS} > B_{rtPS}$, $B_{rtPS} = B_{rtPS} + \Delta$. If B_{rtPS} is larger than $B_{rtPS-MAX}$, let $B_{rtPS} = B_{rtPS-MAX}$. Also, the maximum sustained rate of BE class is decreased by the same amount. This situation means the maximum sustained of rtPS class is not enough.

IV. Simulation Results And Performance Analysis

A. System Framework and Simulation Settings

Based on PMP system in TDD mode for an IEEE802.16 network, we will use Network Simulator (NS2) [5-6] to simulate the proposed scheduling algorithm. To focus on the performance of the protocol, it will be implemented only on the downlink. The length of the frame is set to 5msec and the bandwidth of the system is set to 20MHz as defined in the standard.

Although the type of modulation type of SS can be adjusted dynamically according to signal to noise ratio (SNR) of each time slot, QPSK and coding rate of 3/4 will be used in our simulation. This will let the simulation focus on the proposed protocol. Important parameters used in simulation are summarized in Table 2.

Simulation Parameter	Value or Mode
Channel Bandwidth	20MHz
Frame duration	5ms
System	TDD mode
Total DL Bandwidth	14.34Mbps (QPSK 3/4)
Simulation Time	40 Sec

Table2. The parameters for the simulation

The topology of the simulated network is shown in Figure4. Five classes of packets are transmitted from BS to each MS. As we can see from Table3, the arrival rate for packets of UGS class is 320Kbps (One 200byte packet for each 5msec). The arrival rate of other classes is increased from 128Kbps to 1280Kbps in 12 rounds of simulation and the arrival process is random.

Packet Type	Traffic Type	Bit rate	Packet Size
UGS	UDP/CBR	320Kbps	200 Byte
ertPS	UDP/VBR	128 to 1280 Kbps	200 to 1000 Byte
rtPS	UDP/VBR	128 to 1280 Kbps	200 to 1000 Byte
nrtPS	UDP/VBR	128 to 1280 Kbps	200 to 1000 Byte
BE	UDP/VBR	128 to 1280 Kbps	200 to 1000 Byte

Table3. Data Traffic Setting

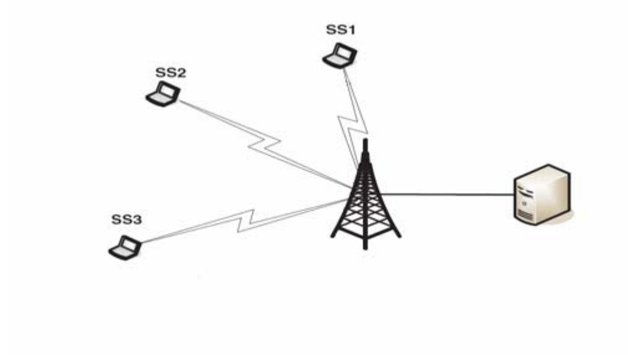

Fig.4. Network Topology

B. Simulation result and analysis

As shown in Fig.5, the BC-General and BC-Reactive have better performance than the original protocol for rtPS class when the load of the network is high. However, the performance for the throughput of BE class will be worse. This is because the bandwidth of this class has been borrowed by the rtPS class to handle the problem caused by high load. This is a desired result because the packets of real-time can't wait and those of BE class can wait.

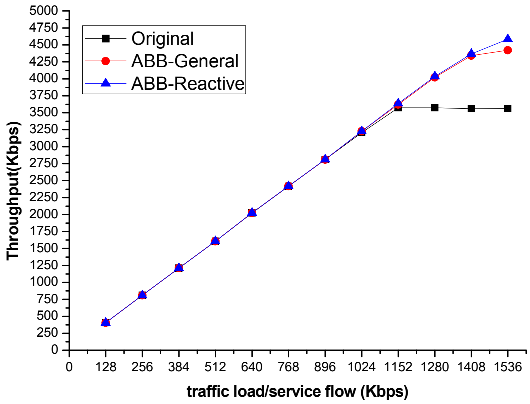

Fig.5. Throughput for packets of rtPS class

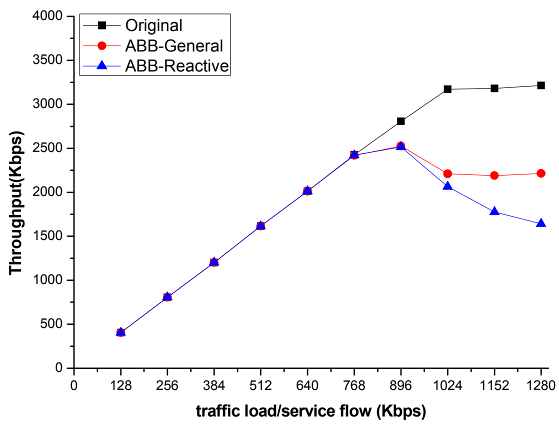

Fig.6. Throughput of BE packets

In Figure5, the result of original protocol shows the throughput of rtPS class reaches its upper bound that when load is 1152Kbps. At this load, the throughput is saturated and the throughput is equal to its maximum sustained rate. With BC-General and BC-Reactive, the throughput is still increasing when the load is above 1152Kbps. This is because the maximum sustained is adaptive adjusting when the load is heavy for these two protocols. .

It can also be seen from Figure5 and Figure6 that the increased throughput of the rtPS class is almost equal to the decreased throughput of BE class when the proposed protocols and original protocol are compared above 1152Kbps. This is because these two classes will almost use the maximum sustained rate to send their frames above this load and the increased part of this rate for rtPS class is “borrowed” from BE class.

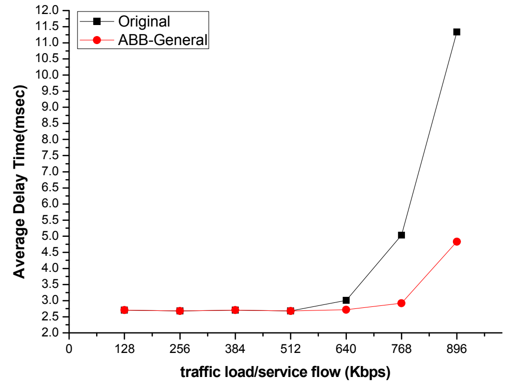

Fig.7. Average Delay Time of rtPS packets

In, Figure 7, the delay performance of rtPS class for original protocol and BC-General are shown. The result shows that the average delay time of rtPS class is improved at high load. When the load is over 896Kbps, the delay time is around 12msec. This delay time is about 4msec with the proposed protocol. As the curve shown, the improvement may be more when the load is higher. This improvement is will become very important as the path of the packets may contain several WiMAX links.

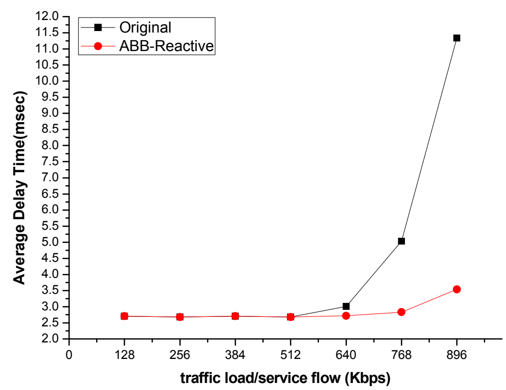

Fig.8. Average Delay Time of rtPS packets

In Figure8, the average delays for rtPS class with original and BC-reactive are compared.

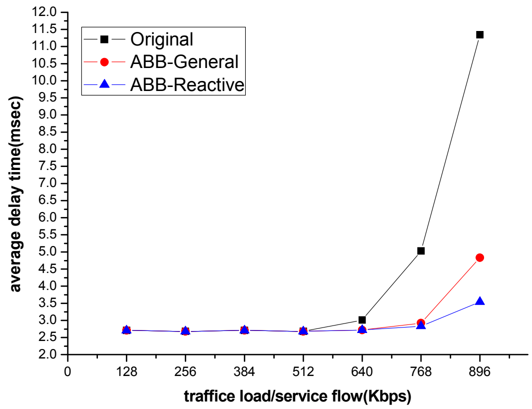

Fig.9. Average Delay Time of rtPS packets

In Figure 9, the result of Figure 7 and Figure 8 are combined. As the result shows, the performance of BC-reactive is better than that of BC-General at 896Kbps.This is because BC-reactive protocol can respond to the congestion condition faster.

If the delay of a packet is over a threshold, it will be discarded in the case of real-time traffic. Therefore, the ratio of frames with some delay is important. If this ratio is large for high delay packets, the QoS for rtPS class will be poor.

In Figure 10, the ratio of received rtPS frames at the load of 640Kbps is shown for each delay time. As we can see, the number of packets with delay above7 msec is about 5% of that of the total transferred packets. Comparing with the original protocol, BC-General and BC-reactive have even smaller ratio with these delays

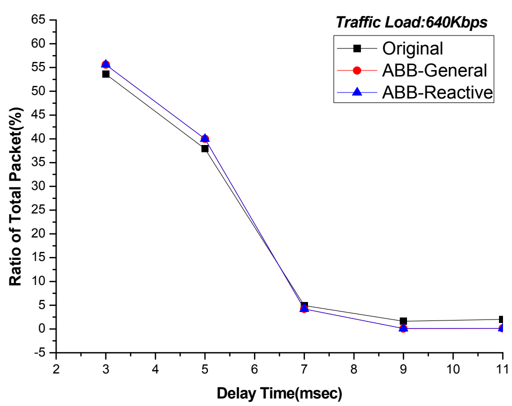

Fig.10. Delay Time-Ratio for rtPS packets at the load of 640kps

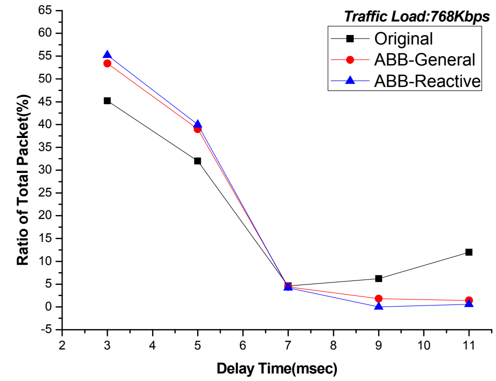

Fig.11. Delay Time-Ratio for rtPS packets at the load of 768kps

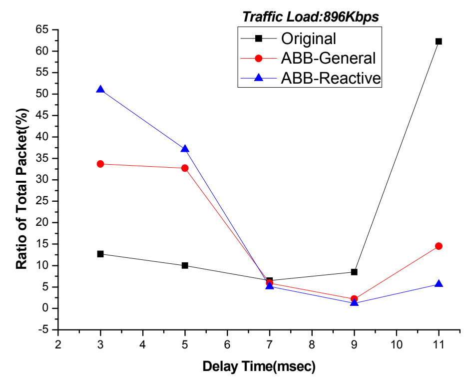

Fig.12. Delay Time-Ratio for rtPS packets at the load of 896kps

In Figure 11 and 12, the ratio of received rtPS frames at the load of 768 and 896Kbps is shown for each delay time. Comparing with the result in Figure 10, we can see the ratio of received frames with high delay is improved more in Figure 11 and Figure 12 when we compare the results of proposed protocol and original protocol. This is because the effect proposed protocol can be more profound at high load.

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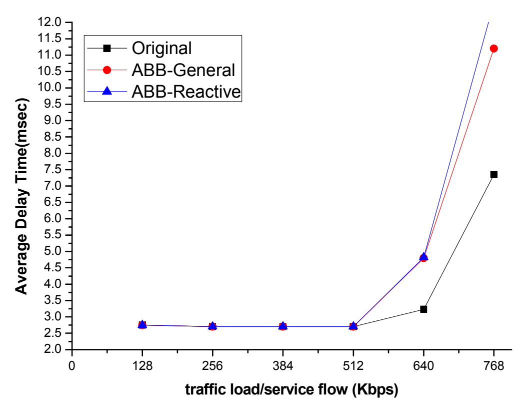

Fige.13. Average Delay Time of BE packets

In Figure13, the delay versus load as Figure9 is shown for BE class. As we can see from the result, the delay is larger at high load for the proposed protocol. This is because the maximum sustained rate of BE class has been "borrowed" by the rtPS class with proposed protocol and there are less time slots to send BE packets at high load. Therefore, the delay of BE class will be larger at high load for proposed protocol..

V. CONCLUSION

The maximum sustained rate is used to provide the QoS of each class. If it is not enough, the QoS cannot be guaranteed. If it is too much, the resource may be wasted and the service of the network will be limited. Also, the load of each class may vary with time and the maximum sustained rate was fixed before. With the proposed protocols in this paper, the maximum sustained rate can adapt to the variant of load and stay at a right point. As (S/N) is used as a standard of power control in mobile system, the queue length after scheduling is used as a standard of bandwidth control in our protocols. As the simulation shown, the proposed protocols have a good result at high load. In wide band mobile system, all users can use their maximum power to send signal if the number of users in a cell is small. Similarly, the adapting is not necessary at light load in our system. Thus, our proposed protocol can not have a better performance at light load. As we can see BC-reactive has a better result at higher load while BC-General has a better at lighter load. A combination of both protocols may be a good idea to try in the future.

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