Fair Sharing Using Real-time Polling Service to Adaptive VBR Stream Transmission in a 802.16 Wireless Networks

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Abstract: The Worldwide Interoperability for Microwave Access system's (WIMAX, IEEE 802.16) broadband networks is an attractive solution for high-bandwidth wireless access network. In the context of a wireless broadband access network, the term open real-time access implies the ability of multiple service BS (base station) to share the deployed access network infrastructure to make available to the SSs (subscribe Station). This paper contains brief variable rate control strategies for real-time multimedia VBR (Variable Bit Rate) service frame over IEEE 802.16 broadband networks. A data rate control mechanism is derived for the case where the downlink channel provides real time polling bandwidth allocation service while keeping the traffic rate parameter constant. The paper shows that common queue scheduling algorithms have some bandwidth allocation fairness problems for real time polling under the WIMAX system (IEEE 802.16) MAC layer. In other words, the use of a VBR real-time polling service by a WIMAX system results in additional access latency jitter and bandwidth allocation disorder in the transmitted variable data rate of a multimedia stream, during the base station's regular time interval polling of Subscribe Station (SSs) for the contention bandwidth request period. However, this scheduling algorithm solves SSs contented bandwidth resource problems, depending on the real-time polling service (rtPS) quality of service function under ranging response non-contention polling period, as the adopted fairness bandwidth allocation max-min algorithm is the only time constraint condition for transmitting real-time multimedia video or VOIP in the 802.16 wireless broadband environment. In addition, we utilized an NS2 simulation to analyze VBR stream capacity, and show that the extended-rtPS by fairness Max-Min algorithm can support more multimedia stream users than other rtPS scheduler algorithms.

Key–Words: WIMAX, extend-rtPS, rtPS, MAC, fairness Max-Min algorithm, VBR, base station, subscribe Station

1 Introduction

Wireless networks widely use multi-media transmission due to the wide use of radio frequency technology. This is what allows WiMAX to achieve its maximum range. The biggest effect factor difference isnt speed; its transmission wireless cover distance. WiMAX outdistances WiFi by miles. In the present day, the WiFis (ieee 802.11) range is about 100 feet (30 m). WiMAX will blanket a radius of 30 miles (50 km) with wireless broadband access. The increased range is due to the frequencies used OFDMA technology and the power of the transmitter. Of course, at that distance, terrain, weather and large buildings will act to reduce the maximum range in some circumstances, but the potential is there to cover huge tracts of land. WIMAX system is generally accepted that one of the several factors inhibiting the use of the real-time multimedia via wireless broadband is the absence of good congestion control mechanisms and persistent phenomenon of jitter. A major challenge to support real-time multimedia is bandwidth allocation under wireless broadband - this scarce resource must be allocated appropriately among participating VBR nodes to ensure an expected quality of service. Once BS (base stations) generate a large payload, the mobile nodes will not procure enough bandwidth when requesting the BS for real-time and variable data rate service.From the perspective of fairness,existing bandwidth allocation techniques can use max-min fairness scheduling for real-time multimedia queue transmission under wireless broadband. Initially only a few traffic parameters (i.e., sustainable maximum data rate and Minimum Reserved Rate) had been intended to be used in the early Wimax network.

Previous work [1], showed that significant number of VBR users to share the same bandwidth with different quality of service (QOS) profiles. However, quality levels were not satisfactory for one or more users with a joint MAC-PHY solution for controlling multi-user access to the wireless medium in a VBR WLAN. In particular, multiple access management aspects were faced when trying to propose an efficient strategy for resource allocation. Nevertheless, there are some WiMAX NS-2 modules implemented by Network and Distributed Systems Laboratory (NDSL) [2] and Chen [3]. These modules implement the physical (PHY) and Medium Access Control (MAC) layers of a WiMAX system. The NDSL module implements the Orthogonal Frequency Division Multiple (OFDM) PHY and queue scheduling for quality of service (QOS) under MAC layer. The MAC layer of these WiMAX modules contains the management messages. The NDSL WiMAX module supports, fragmentation, reassembly of frames and Call Admission Control (CAC). Camarda [8]proposed an original algorithm based on concepts of multi-class network queuing for statistical aggregate bandwidth estimation of smoothed VBR VOIP stream, and respecting a specified loss probability (the QoS parameter taken into account). Belghith [9] introduced a scheduling algorithm able to differentiate between the service QoS classes and to take into account the QoS reserve. If there are not enough available signals, the BS allocates all the remaining signals, whereas it removes this request. The BS frequently updates the bandwidth to be allocated in the succeeding frames. The unit change of the time-utility function value indicates urgency of scheduling of packets as time passes, i.e. the timeliness timing constraint often means that tasks should be completed by a certain given time instant with the urgency of scheduling being expressed as a function in delay.

Recently, the IEEE 802.16e provided a new service discipline aimed at different QoS for subscriber stations (SSs) that contend for resources by requesting BS bandwidth. The IEEE 802.16e standard supports five different scheduling QoS in the uplink: Unsolicited grant service (UGS), Real-time polling service (rtPS),extended-Real-time polling service (ertPS), Non-real-time polling service (nrtPS), and Best Effort (BE). The first three types of services support VBR stream in real-time demand. VBR streaming over wireless WMAN rate, can be a VBR stream or a constant bit rate (CBR) stream that follows in accordance with a bursty traffic model: the situation chosen. With UGS service, BSs generate fixed size data packets (such as CBR) on a periodic basis, at the same time, unsolicited grants allow SSs to transmit their packet data units (PDUs)without requesting bandwidth for each frame. Since the bandwidth is allocated without request contention, the UGS provides hard guarantees in terms of both bandwidth and access delay. However, the UGS class allocates grant size, like, VoIP without silence suppression (audio streaming without silence suppression). rtPS supports service flows that generate variable data packets on a periodic basis, like, MPEG video, and VoIP with silence suppression. Bandwidth requirements are an indeterminate data rate for UGS grant interval period of peer node connection setup time. As a result, peak stream bit rate-based using CBR allocation would exposed to reduce resource network utilization, whereas the average bit rate CBR allocation can result in unacceptable packet delay and jitter. rtPS service has been introduced to accommodate such flows, but rtPS service can adjusted variable bit rate situation for transmit video stream or data stream. For this rtPS service, BSs provide periodic transmission opportunities by means of a basic polling mechanism. The SSs can exploit these opportunities to ask for bandwidth grants, so that the bandwidth request can be ensured to arrive at the BSs within a given guaranteed interval. The ertPS class is designed for real-time traffic with variable data rate (such as VBR service with silence suppression) over the IEEE 802.16e Wireless Broadband network. BSs continuously offer the same amount of bandwidth to SSs unless otherwise explicitly requested by the SSs. The ertPS is a scheduling mechanism that builds on the efficiency of both UGS and rtPS. BS provides unsolicited unicast grants as in UGS, while it records every SSs latency in bandwidth request. However, ertPS does not allocate bandwidth on the basis of a fixed packet size as in UGS. nrtPS and BE classes are service for non-real-time traffic. The remainder of the article is organized as follows: In Section 2, we describe one VBR stream source system model and design a Poisson Markov process for a VBR system model. In Section 3 we analyze and compare difference between three rtPS scheduling algorithm for VBR stream using alike VBR technology. In Section 4, an algorithm is proposed that can be adopted for rtPS and improve performance for VBR transmission of indeterminate data rate on VBR stream over IEEE 802.16e BWA. Finally, in Section 5 some conclusions are given.

2 VBR Stream over WLAN via IEEE 802.16e System Model

UGS class multimedia traffic on 802.16e WMAN is quite bursty in nature. Simple models such as the Poisson processes are unable to capture the important characteristics of these sources. To model bursty traffic sources different approaches are available: many of them using Markov modulated processes (MMP).

These are doubly stochastic processes in which each state of N states of an embedded Markov chain originates another stochastic process. If this originated is a Poisson process, the MMP is called a Markov modulated Poisson process, One simple case that is very suitable to model video sources is the On-Off model (see Fig. 1). According to the On-Off model, the signal stream of a single source is modeled through an alternating sequence of burst and silence periods. Duration times of burst and silent periods are exponentially distributed with mean $T_{on} = 1/\lambda$ and $T_{off} = 1/\mu$ respectively.Therefore, the expectation of the period duration is denoted as $E[t_{on}]$ for ON period and $E[t_{off}]$ for OFF period.

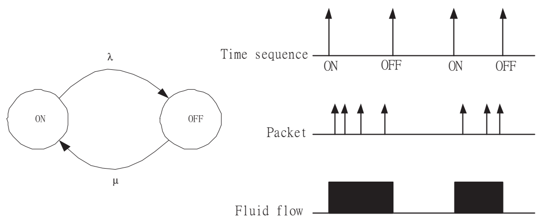

Figure 1: VBR ON-OFF fluid source coverage and packet source .

Assume that the cell arrival probability in ON state is determined by a Poisson random variable X with parameter λ, where $E[X] = \lambda$. Let ρ represent the utilization of the time-slot for a VBR source (the percentage of a slot being used). Then, we have $\rho = \frac{\lambda E[T_{on}]}{E[T_{on}]+E[T_{off}]}$. Given a cycle with length N and a VBR source with claimed peak rate c_i and mean rate μ_i ,we have $c_i = \lambda$N, and $\mu_i = \rho N = \frac{\lambda N E[t_{on}]}{E[T_{on}]+E[T_{off}]}$. During the burst period packets with fixed length are generated with constant inter arrival time Δt.

Let N be the independent identical number of VBR ON/OFF sources is modeled by Markov process with (N+1)stats [11]. In this model, the aggregate bit rate of the N sources is quantified into a number of discrete levels, as shown in Fig. 2. State of the process is determined by number of active VBR sources $(k = 0, 1, 2 \ldots, N)$. Corresponding burst arrival intensities are $(0, A, 2A, \ldots, AN)$.The denote separate states and quantified aggregate bit rate constitutes a Markov chain.Probability density of passing between near-by states are $(N-k)\lambda -$ passing from state k to $k+1$ and $k\mu -$ passing from k to $k-1$.

Thus the probability of k active sources follows a

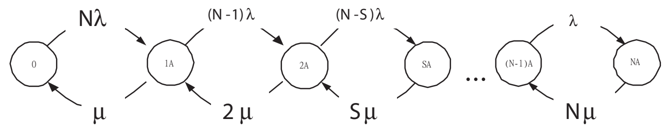

Figure 2: State transition diagram in the VBR model.

binomial distribution equation (6):

$$P_N(N) = \begin{pmatrix} N \\ k \end{pmatrix} \left(\frac{E[T_{ON}]}{E[T_{ON}]+E[T_{OFF}]}\right)^k \left(\frac{E[T_{OFF}]}{E[T_{ON}]+E[T_{OFF}]}\right)^{N-k}, k = 0, 1, 2, \ldots, N \tag{1}$$

Where p is probability of single VBR source is in the state ON. Mean intensity of composed flow is: $D = ND_n = NAp$.

The superposition of N independent VBR sources is modeled by Markov process with (N N+1) states and follows a binomial distribution equation (5):

$$\bar{N} = \sum_{k=1}^{N} n \cdot P_N(N)$$

$$= \sum_{k=1}^{N} n \cdot \begin{pmatrix} N \\ k \end{pmatrix} (\frac{E[T_{ON}]}{E[T_{ON}]+E[T_{OFF}]})^k \left(\frac{E[T_{OFF}]}{E[T_{ON}]+E[T_{OFF}]}\right)^{N-k} \tag{2}$$

In rtPS service SSs can request VBR stream at a different rate every time, such that its service polling and updated bandwidth requests are based on the traffic load change. rtPS service is more flexible, using CBR or VBR traffic transmission for VBR data streams. If the rtPS employs VBR traffic mode, it must consider the SSs request contention resource resulting in packet delay and packet loss rate.Fig. 3 shows the bandwidth allocation for QoS architecture present under 802.16. The uplink packet scheduling (UPS) module controls all the packet transmissions in the uplink (UL). As the protocol is connection-oriented, the application should establish a connection between the BS and the associated service flow (UGS, rtPS, nrtPS or BE). The BS identifies connections by assigning a unique Connection ID (CID) to each one. The 802.16 standard defines the signaling process for the establishment of a connection (Connection- Request and Connection-Response) between SS and BS, but it does not specify the rules for admission control. The call admission control mechanism distributes SSs requests to BS by employing an algorithm determining whether SSs can allocate bandwidth resource or not. For fairness sharing bandwidth allocation , our motivation in this work is to ensure fairness in terms of throughput in the shared access channel, among SSs and BS simultaneously. Accomplishing both of the above objectives is challenging, because the SSs and BS are connected to opposite ends of the wireless access channel network, which itself may be quite long about 20-50km and hence is subjected to some significant propagation delay. The real-time polling service is adopt Call Admission Control Dynamic Adjust Bandwidth allocation for different distanced SSs request requirement. In multiservice WiMAX networks, call admission control (CAC) plays a critical role. In this paper, we address CAC problem from the perspectives of both WiMAX service BS and SSs. Specifically, we formulate CAC as an optimization problem, in which the demands of service providers and subscribers are taken into account. To solve the optimization problem, we develop a utility and fairness constrained revenue algorithm. For the rtPS service, the maximum amount of time that each SSs can transmit in a MAC frame is given by

where:

$$M_{\text{rtPS,MAX}} = \left\lfloor \frac{E[T_{ON}] - E[T_{overhead}] - N \cdot R_p \cdot t_{rtPS}}{T_p \cdot N} \right\rfloor . \quad (3)$$

R_p : VBR active state data rate bits/sec.
$T_{overhead}$:VBR data sent extra time for packet stream and bandwidth request, including: initial ranging, contention period time.

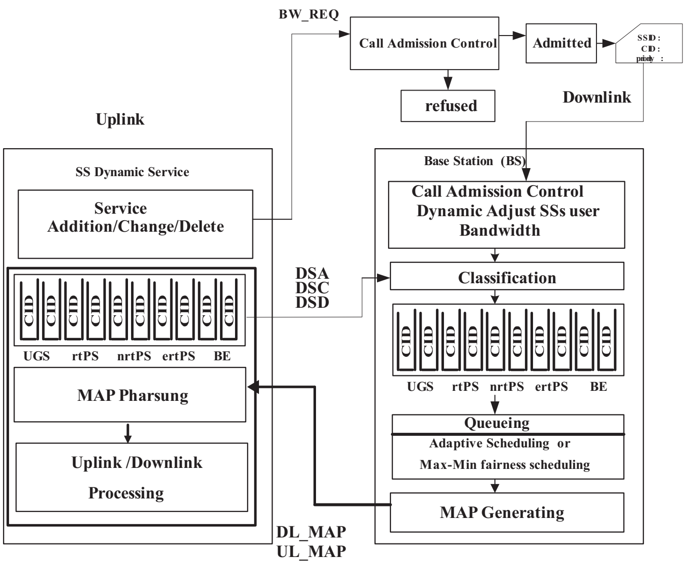

Figure 3: QOS PROCESS structure for 802.16.

ertPS service is considered to improve problem UGS packet loss and rtPS packet latency time. However, the paper proposes a VBR traffic strategy described by the following two-state Markov chain. For mean the $R_{ertPS}(maximun, minmun)$.Let X(t) is random variable associated to the state of the chain, the probability the system is shown that: $P_{ij}(t) = P_r(X(E[T_{on}] + E[T_{off}]) = j|X(E[T_{off}]) = i)$.

3 proposed algorithm real-time for VBR streams

In the preceding section 2, the ertPS class adopted the variable data rate. Moreover, the VBR traffic was subdivided into a real time (RT) class and non-real time (NRT) class. VBR (RT) is used for connections in which there is no fixed timing relationship between samples, but which still need a guaranteed QoS. This paper adapts to the above Markov chain function which must consider VBR packet transmission for real-time environment under 802.16 Broad bandwidth (see Fig. 3). Moreover the QoS adapts to transmit variable bit rate (VBR) packet under 802.16 Broadband considers the following parameters:

1. network loss: IP datagram lost due to network congestion (router buffer overflow)
2. delay loss: IP datagram arrives too late for active state at receiver The packet serviceflow delays include processing, queueing in network; end-system (sender, receiver) delays K etc.
3. generally the typical voice packet maximum tolerable delay is 400 ms.
4. loss tolerance: depending on voice encoding, losses concealed, packet loss rates between 1.

For an N SSs users queue M/M/1 type single server system with a unity service rate, the optimal system total delay is characterized by the constraint caused by the individual arrival rates of each input traffic stream. We are concerned with priority schedule such as (WFQ, RR, max-min fairness...etc) statement that balance traffic characterization on the MAC layer. If we have VBR streams, with a bit-rate variable between a minimum and a maximum it will be quite inconvenient to assign to a stream more bandwidth than truly needed. The proposed algorithm has shown that variable hit rate VBR traffic can be represented as a two-state Markov chain. In one state the transmission rate is less than the average bit rate, in the other state it is more. Since after a complete cycle, in the Markov chain the system returns to the initial state, it is possible to introduce the probability to deliver all the packets accumulated during the cycle. However, with the use of sequential round-robin allocation levels, this problem is not likely to be an issue, because most of the time the space allocated for real-time traffic is much less than the space booked. Moreover, BS allocated bandwidth resource to SSs for real-time polling data, can be fairly allocated bandwidth between minimum and maximum to adjust real-time polling traffic for each SS.

3.1 Max-min fairness

The notion of max-min fairness is well known in political science [5]. It was introduced as a design objective for communication networks by Bertsekas and Gallager [4]. The principle of max-min fairness is to allocate network resources in such a way that the bit rate of a flow cannot be increased without decreasing the bit rate of a flow having a smaller bit rate. The paper employed real-time polling service to transmit VBR packet with service flow that is variable for an active VBR state. Max-min fairness is uniquely defined by the following $water-filling$ procedure:

1. start from a bit rate equal to zero for all flows;
2. increase the bit rate of all flows at the same speed until the bit rate of some flows is constrained by the capacity set; freeze the bit rate of these flows;
3. apply step 2 repeatedly to non-frozen flows until the bit rate of all flows is constrained by the capacity set.

Note in particular that max-min fairness is Pareto efficient.If the boundary of the capacity set does not contain any segment parallel to one of the $class-i$ axes, $i = 1, \ldots, N$., the $water-filling$ procedure stops after the first instance of step 2 and the allocation is simply given by the intersection of the line of direction x with the boundary of the capacity set. $\mathbf{Theorem 1}$ Max-min fairness is balanced if and only if for some $L \geq 1$, the network bandwidth reduces to a set of $\mathcal{C}$ independent links in the sense that there exists a partition $C_1, \ldots, C_L$ of the set of classes $\{1, \ldots, N\}$ and some positive constants $c_1, \ldots, c_L$ such that:

$$\mathcal{C} = \{\varphi : \sum_{i \in I_1} \varphi_i \leq c_1, \ldots, \sum_{i \in I_L} \varphi_i \leq c_L\} \quad (4)$$

3.2 Min-max fairness and proportional fairness

The analysis of fairness issues in networks has its origin in the framework of wired networks [10][12][13]. Although we are free to define specialized notions of fairness for particular networks of interest, two fundamental fairness principles are established. These principles give rise to the majority of related fairness notions, which are applicable to different network types (wired/wireless), different network topologies (cellular/adhoc networks), and different QoS parameters (e.g., the end-to-end delay in multihop ad hoc networks or data rate in cellular networks). The first fairness principle is referred to in this work as min-max fairness; it involves making the worst QoS parameter (of a route, link, etc.) as good as possible. In wired networks, the min-max fairness equilibrium of QoS parameters is the one at which no QoS parameter q_i can be improved without the degradation of any QoS parameter q_j, $j \neq i$, which is already inferior to q_i. The same definition translates usually to the case of wireless multihop ad hoc networks, where the QoS parameters are associated with routes (end-to-end QoS)[6][14]. The available bandwidth for a Real-Time Polling Service, μ_n , may change in an unpredictable manner since the transmission rate of the VBR of video stream traffic is time varying. The bandwidth available to the controlled sources, on the other hand, is subject to further uncertainty due to the variations in the uncontrolled traffic. Our concern is to build a system model mechanism to control VBR video stream for a real-time polling service under WIMAX. The frame transmit process must effectively adapt to the variable data rate when users change to request for service a real-time polling service. For this reason, We have employee Max-min fair allocation (MMFA) to define the ideal fair distribution of a shared scarce resource. Given the resource capacity $\mathcal{C}$ and n SSs, under MMFA a SS i is guaranteed to obtain at least $C_i = C/n$ of the resource. If some SSs require less than what they are entitled to, then other SSs can receive more than C/n. Under weighted MMFA (WMMFA), a SS i is guaranteed to obtain at least

$$C_i = \frac{w_i}{\sum_{j=1}^{n} W_j}\mathcal{C}$$

of the resource. However, MMFA does not specify how to achieve this in a $dynamicsystem$ where the demands for a resource vary over time, thus guaranteeing (weighted) min-max fair resource allocation when averaged over a long run.The following Fig. 4. shows the max-min fair allocation procedure.

We assume that real-time traffic is generated according to the Poisson session model. We further suppose that flows have a constant peak rate drawn from a set $\mathcal{C} = (\mathcal{C}_1; \ldots; \mathcal{C}_N)$ with $c_1 > c_2 > \cdots > c_N$. Flows with peak rate c_i constitute WIAX QOS class i. The number of WIMAX QOS class i flows in progress

Src	Demands	Allocation #1 [bps]	Balances after 1st round	Allocation #2 [bps]	Balances after 2st round	Allocation #3 (Final) [bps]	Final balances

Figure 4: Weighted max-min procedure allocation

at time t is denoted by $X_i(t)$. This analysis is based on a quasi-stationary timescale separation. We first suppose that X_i is fixed and evaluate the distribution of the number of active flows (on state real-time VBR of video stream) for each state $x = (x_1; \dots :; x_N)$. We then introduce assumptions that allow us to estimate the stationary distribution of $\mathcal{X}$ in order to derive the required unconditional distribution. It should be noted that the max-min fairness allocation Algorithm 1 could be used for proper bandwidth allocation by the BS decision rule under wireless 802.16 broadband.

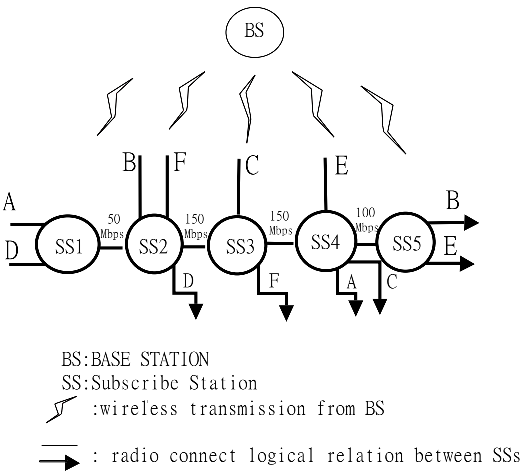

Figure 5: Max-min algorithm allocation rule

In step 4 of iteration k the service rate of each SS requesting service $p \in P^k$ receives a minimum rate increment(calculated in step 3). If service p is not included in P^k (i.e. service p already uses a saturated link from iteration k-1 from the BS bandwidth allocation), then its rate remains the same. Having calculated the total flow F_α (step 5 of iteration k), we then determine the new sets A_{k+1}, P_{k+1} at steps 6 and 7, respectively, by removing all of the saturated SSs with no successful links from a request to the BS, the services that use them, and the capacity associated with the QOS services. If we apply the aforementioned algorithm to the WIMAX network of Fig. 5 , then after four iterations we will obtain the following results. The service calls A and D will be allocated a rate of 5.55 Mbps; service calls B and E, a rate of 11.1 Mbps; service call C, a rate of 33.3 Mbps; and service call F, a rate of 50 Mbps. The notion of fairness can be explained by observing that services A and D, which

Algorithm 1 Calculate weighted max-min bandwidth allocation algorithm

A^k : denotes the set of SS links flow not saturated at the beginning of step k.
P^k : denotes the set of SS not passing through any saturated link at the beginning of step k.
n_α^k : denotes the number of SSs that use BS link α and are in P^k.
$\tilde{\alpha}^k$: denotes the rate increment added to all of the SSs in P^k at the k^{th} step.

1: Initial conditions:$k = 1, F_\alpha^0 = 0, r_p^0 = 0, P^1 = P$ and $A^1 = A$
2: n_α^k= the number of SSs with a different QOS service $p \in P^k$ crossing link $_\alpha$.
3: **while** $n_\alpha^k \neq NULL$ **do**
4: $\tilde{\alpha}^k$=min$_{\alpha \in A^k} \frac{C_\alpha - F_\alpha^{k-1}}{n_\alpha^k}$.
5: **if** $p \in P^k$ **then**
6: build a high priority link $_\alpha$
7: **else**
8: {the bit rate over the constrain set up bandwidth for system allocation mechanism}.
9: **end if**
10:

$$r_p^k = \begin{cases} r_p^{k-1} + \tilde{r}^k & for p \in P^k \\ r_p^{k-1} & otherwise \end{cases} \quad (5)$$

11: $F_\alpha^k = \sum_{p \times \alpha} r_p^k$(for those p, which use link$_\alpha$.)
12: $A^{k+1} = \{a | C_\alpha - F_\alpha^k > 0\} \because A^{k+1}$denotes the set of SS links not saturated at the beginning of iteration k+1 under a BS bandwidth allocation condition
13: $P^{k+1} = \{p|$do not cross any SS link $a \in A^{k+1}\}$
14: $k := k + 1$
15:
16: **if** P^k is empty **then**
17: $STOP$
18: **else**
19: go to 1
20: **end if**
21: return per k
22: set of paths P^k and flows $\forall_{p_i^j \epsilon P_i} f(p_i^j)$
23: **end while**

have the same bottleneck link (1-2), and services B and E, which also have the same bottleneck link (4-5), require an equal amount of bandwidth (5.55 and 11.1 Mbps, respectively). As has already been shown, the computation of a max-min fair rate vector depends on an eight-step algorithm whose iterations are equal to the number of network links.

3.3 Bottlenecked and active flows

For notational convenience, define $c_0 = \mathcal{C}$ and $c_{N+1} = 0$. If

$$\sum_{i=1}^{n} x_i c_i > \mathcal{C}$$

fair queueing realizes max-min fair sharing with fair rate θ given by

$$\theta = \frac{\mathcal{C} - \sum_{i>J} x_i c_i}{\sum_{i>J} x_i}$$

where $J = J(x)$ is the unique integer, $1 \leq J \leq N$, such that $c_J \geq \theta \geq > c_{J+1}$. Variable flows of peak rate greater than or equal to c_J are bottlenecked and realize the fair rate θ while the others preserve their peak rate through the scheduler. If

$$\sum_{i=1}^{n} x_i c_i \leq \mathcal{C}$$

all flows preserve their peak rate. Let $J(x) = 0$ in this case. Note that J defines a partition of the state space. Consider now the operation of the RTPS algorithm. All bottlenecked flows are included in CongestList. In addition, any non-bottlenecked flow having emitted a packet in the recent past, such that its finish tag is less than VirtualTime, is also included. To evaluate the BASE Station distribution of the total Bandwidth number we must make some assumptions about packet lengths and the arrival process of packets from non-bottlenecked flows. The most convenient assumption is to suppose all packets of bottlenecked flows have maximum size MTU for real-time Polling service. This is reasonable given the packet size statistics of high peak rate flows reported above and is arguably a worst case assumption for the distribution of CongestList size. Given this assumption, and taking MTU as the unit of virtual time, it is easy to see that VirtualTime takes only integer values. The periods when at least one flow is bottlenecked and VirtualTime is constant constitute "busy cycles". Any packet from a non-bottlenecked flow arriving in a busy cycle acquires the time stamp VirtualTime and further perpetuates the cycle for the duration of its own transmission. This flow remains in CongestList until the end of the cycle. It is then removed since its finish tag cannot be greater than $VirtualTime + 1$ (it has size $l < 1$ in MTU units). Busy cycles also occur when there are no bottlenecked flows in progress. In this case a busy cycle is initiated by a packet arriving to an empty system. VirtualTime remains equal to zero in such cycles. This is because VirtualTime changes more rarely so that small non-backlogged packets remain longer in the list. The size of CongestList is largest just before the end of a busy cycle. To evaluate the stationary distribution of this number MaxList, we assume each non-bottlenecked flow emits at most one packet in a cycle. To model the arrival process we assume each flow independently emits its packet between time t and $t + dt$ after the start of the busy cycle with probability $\alpha dt + o(dt)$. This finite source Poisson process facilitates analysis while accurately accounting for the number of contributing flows$\sum_{i>J} x_i$.From the above description it may be recognized that an estimate for the number of nonbottlenecked flows contributing to MaxList can be derived from an evaluation of the busy period of a queue with exceptional first service. The latter corresponds to the transmission of one MTU sized packet from each of the $\sum_{i\leq J} x_i$ bottlenecked flows. The arrival process results from $N = \sum_{i>J} x_i$ flows independently emitting at most one packet according to the process described at the end of the last section. The service time distribution derives from that of the packet length F(s). We assume packet lengths are $i.i.d.$ for all non-bottlenecked flows. Let packet length be measured in units of MTU and denote the mean by σ.The assumed per-flow arrival rate α is the average rate for all non-bottlenecked flows: $\alpha = R = (N/\sigma)$ where $R = \sum_{i>J} x_i c_i$ is their overall bit rate. The probability k packets arrive in an interval of length $T_{on} = 1/\lambda$ and silence in an interval of length $T_{on} = 1/\mu$ from the start of the busy cycle is then:

$$\Lambda_N(k,u) = \left(\begin{array}{c} N \\ k \end{array} \right) (1 - e^{-\alpha 1/\lambda})^k e^{(N-k)\alpha\mu} \mathbb{I}_{(0,+\infty)}(u) \tag{6}$$

The conditional distribution of MaxList is given by the following proposition.

4 Simulation Results

4.1 Simulation Model

This paper have integrated into the existing implementation of WiMAX QoS parameters, QoS classes, uni-

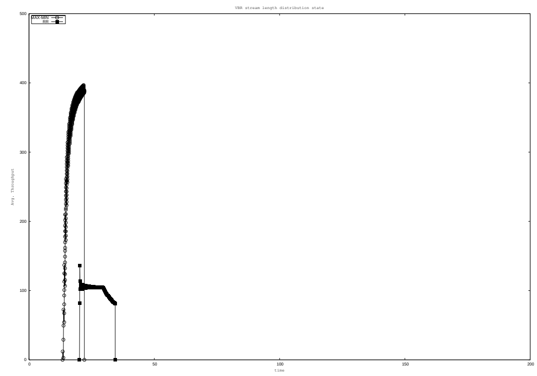

Figure 6: RTPS scheduling throughput simulation VBR transmit situation

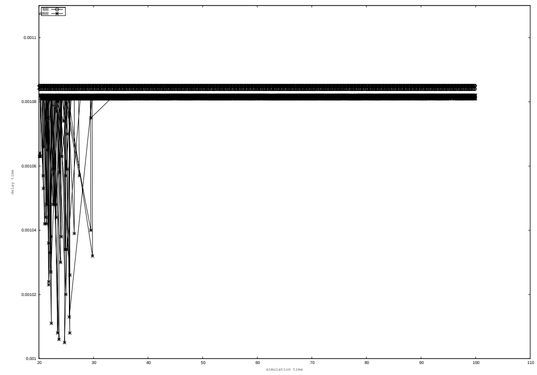

Figure 7: delay time of RTPS-RR scheduling versus RTPS-WRR scheduling service time for simulation VBR transmit situation

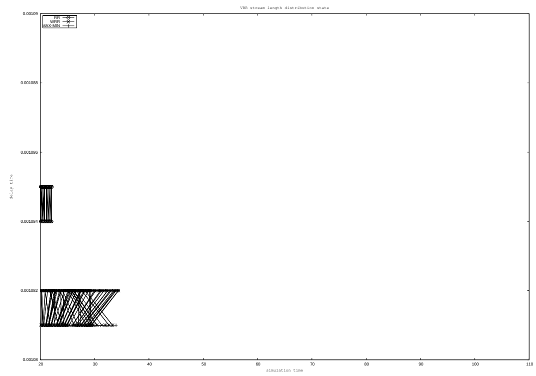

Figure 8: delay time of RTPS-RR scheduling versus RTPS-MAX-MIN scheduling service time for simulation VBR transmit situation

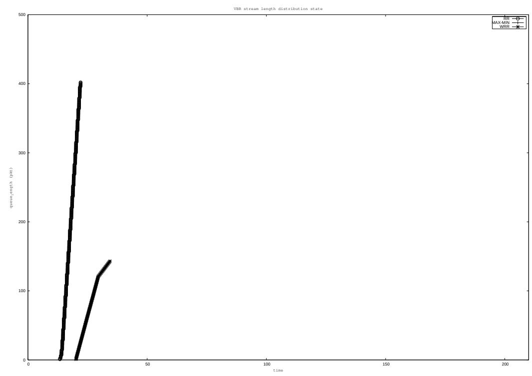

Figure 9: RTPS scheduling queue length simulation VBR transmit situation

cast and contention request opportunities, three rtPS schedulers, and simple UGS and BE schedulers. The main parameters of the simulation model are represented in Table 1. We consider nine rtPS, and two BE subscribers. The subscribers can use the QPSK 1/2, QPSK 3/4, 16-QAM 1/2, 16-QAM 3/4, 64-QAM 2/3, and 64-QAM 3/4 MCSs.

4.2 Performance of rtPS scheduling evaluation for VBR stream transmit

In this section, we study the behavior of the rtPS schedulers for VBR stream transmit. We consider the scheduler of the existing module as well as real-time VBR stream implemented schedulers: RR (round robin), max-min RR, WRR (Weight round robin). Each UGS SS generates Constant Bit Rate (CBR) traffic with a rate of 160 Kbit/s s. We have also nine rePS SSs that generate VBR stream traffic. Fig. 6. shows the throughput of the rtPS connections as a function of the rtPS traffic load submitted in the network. This Figure shows the low efficiency of the Existing RR scheduler. Such a scheduler, in the quest of the simplifying of the scheduling steps, throttles the network traffic. Indeed, this scheduler allocates all the symbols to one SS even if it has not data to send. We note that the the rtPS throughput of the Existing RR scheduler is nearly seven times worse than that of other implemented rtPS schedulers(such as MAX-MIN scheduling ,WRR scheduling,...etc). We observe that the MAX-MIN schedulers nicely outperform the other schedulers with a maximum throughput of 4 Mbit/s. The schedulers favor SSs having employee highest values and then the most efficient. We also observe that the RR scheduler provides rtPS throughput less than that of the MAX-MIN schedulers and WRR schedulers. This is due to the fact that the channel quality of the diRerent SSs in not taken into consideration. Fig. 7 and Fig. 8 shows the mean delay time of the rtPS connections as a function of the rtPS traffic load submitted in the network. The mean delay time is a vital parameter for the real-time applications. We note that the Existing RR scheduler requires a large average delay to deliver a data packet. This is because this scheduler does not provide sufficient symbols to the SSs. It periodically allocates all the remaining

Table 1: System Parameters in Simulations

Parameter	Value
Frequency	5Mhz
Propagation	Two Ray Ground
Antenna model	Omni antenna
Antenna height	1.5m
Transmit antenna gain	1
Transmit power	0.25
Frame duration	20ms
Cyclic prefix (CP)	0.0625ms
Packet length	1024 bytes
Simulation duration	100s

symbols of an uplink subframe to one rtPS SS. This period is equal to the frame duration multiplied by the number of all the SSs.However,This average queue length includes the packet in the BS server (physically, the packet being transmitted). The Queue-Monitor object gives the number of VBR stream packets waiting in the queue, without counting the packet in the BS server. From fig. 9. the will notice that there is a difference compute index between the theoretical value and the measured value. The compute index is the average number of packets occupying the BS resource of the queue.The fig. 9. show that rtps-MAX-MIN scheduling can balanced the packet load which it is maximum no over 2700 queue size buffer under BS server. Above all,the validate this VBR stream that it consider including throughput, delay latency service time and queue length value using simulation. The evaluation level will be a determine BS performance factor for distribution bandwidth resource that this large VBR stream queue request packet from SSs.

5 Conclusion

In this paper, we proposed an adaptive VBR real-time WiMax system structure, adopted weighting fairness max-min transmission packet for streaming VBR over a wireless broadband system. The purpose of research is comparable to real-time streaming by Polling service method for VBR. The system can rapidly adjust to variable bit rate due to bandwidth transferred allocation resources to other service queue users within the wireless broadband system. Current work is dedicated to the extension of the approach to a multi-channel selective for efficient bandwidth management and improved overload resulting in many subscriber station contention bandwidth resources. The paper proposed structure to real digital television and video signal for service user under wireless multimedia network that can be expected in future.

References:

[1] G.B. Scorza,C. Sacchi, F. Granelli, "An Adaptive MAC-PHY Approach for Medium Access Control in VBR MC-CDMA Systems," in *Multimedia Services Access Networks, 1st International Conference*,pp. 96–100, 13-15 June 2005.

[2] "The design and implementation of wimax module for ns-2 simulator," in $http://ndsl.csie.cgu.edu.tw/wimax\_ns2.php$, September 2007.

[3] J. Chen, C.-C. Wang, C.-W. Chang, S.-S Liu, J. Guo, W.-J. Lien, J.-H. Sum, and C.-H. Hung "The design and implementation of wimax module for ns-2 simulator," in *ACM Valuetools*, Pisa, Italy, October 10, 2006.

[4] D. Bertsekas and R. Gallager, *Data Networks*, Prentice Hall, 1987.

[5] J. Rawls, *A Theory of Justice*, Belknap Press, 1971.

[6] L. Tassiulas and S. Sarkar, "Maxmin Fair Scheduling in Wireless Networks," in *Proc. IEEE INFOCOM'02*, vol. 2, pp. 763–772, New York, NY, USA, June 2002.

[7] P. Camarda, D. Striccoli, "Queueing networks approach for bandwidth estimation of smoothed VBR video streams," in *Elsevier Science Publishers B. V.*,vol. 57, pp. 1–18,Issue 1, May 2004.

[8] P. Camarda and D. Striccoli, "Queueing networks approach for bandwidth estimation of smoothed VBR video streams," in *Performance Evaluation, Elsevier'04*, vol. 57, pp. 1–18,August 2004.

[9] A. Belghith, L. Nuaymi, "Design and Efficiency based Packet Scheduling Algorithm for OFDMA wireless System," in *Proc. IEEE ICC'05*,Vol. 4 , pp. 2779–2785,May 2005.

[10] E. L. Hahne, "Round-robin scheduling for max-min fairness in data networks," in *IEEE Journal on Selected Areas in Communications*, vol. 9, no. 7, pp. 1024-V1039, 1991. vol. 46, no. 6, pp. 784V798, 1998. Switzerland, 2003.

[11] A. -L. Beylot, "Models De traffic et commutateurs pour l'evaluation de la perte et du delai dans les reseaux ATM," these de doctorat de l'Universite Paris 6, September, 1993.

[12] F. P. Kelly, "Charging and rate rate control for elastic traffic," in *European Transactions on Telecommunications*, vol. 8, no. 1, pp. 33–37, 1997.

[13] F. P. Kelly, A. K. Maulloo, and D. K. H. Tan, "Rate control in communication networks: shadow prices, proportional fairness and stability," in *Journal of the Operational Research Society*, vol. 49, no. 3, pp. 237–252, 1998.

[14] B. Radunovic and J.-Y. Le Boudec, "Why max-min fairness is not suitable for multi-hop wireless networks," Document, Ecole Polytechnique Fédérale de Lausanne (EPFL), Lausanne, Switzerland, 2003.