QoS-Aware Active Queue Management for Multimedia Services over the Internet

I-Shyan Hwang, *Bor-Jiunn Hwang, Pen-Ming Chang, Cheng-Yu Wang

Abstract—**Recently, the multimedia services such as IPTV, video conference emerges to be the main traffic source. When UDP coexists with TCP, it induces not only congestion collapse but also unfairness problem. In this paper, a new Active Queue Management (AQM) algorithm, called Traffic Sensitive Active Queue Management (TSAQM), is proposed for providing multimedia services. The TSAQM comprise Dynamic Weight Allocate Scheme (DWAS) and Service Guarantee Scheme (SGS), the purpose of DWAS is to allocate resource with fairness and high end-user utility, and the purpose of SGS is to determine the satisfactory threshold (TH) and threshold region (TR). Several objectives of this proposed scheme include achieving high end-user utility for video service, considering the multicast as well as unicast proprieties to meet inter-class fairness and achieving the QoS requirement by adjusting the thresholds adaptively based on traffic situations.**

Index Terms—**AQM, QoS-Aware, Multicast.**

I. INTRODUCTION

In order to improve the congestion collapse problem, the early TCP protocol has prompted the study of end-to-end congestion avoidance and control algorithms [1]. Recently, several applications, such as IPTV and VoIP, using User Datagram Protocol (UDP) without employing end-to-end flow and congestion control, are being increasingly deployed over the Internet. When UDP coexists with TCP, it not only induces congestion collapse problem but also unfairness problem that each flow cannot get the same treatment and causes the Internet unstable and lower link utilization. The congestion control methodologies can be categorized as - *the Primal* and *the Dual* [2]. The Primal congestion control is that the source node dynamically adjusts sending rate or window sizes depending on the indication information fed back from the Internet. The *Primal* methodology has two types, which is classified based on the way of reacting to congestion, adjusting the congestion window size, called *Window-Based*; or the packet transmission gap, called *Rate-Based.* The Rate-based is more suitable for delivering real-time traffic due to it can provide more smooth transmission rate and it has no need to wait an ACK message from receiver [3]. The *Primal* methodologies [4] use the fluid

model to analyze the Internet traffic load or use probing-based methods including probe gap model (PGM) and probe rate model (PRM) to estimate residue bandwidth in the bottleneck [5]. In essence, those algorithms regarding the amount of packet loss and value of RTT's variation imply as the network congestion occurs. However, the packets loss is not only due to congestion occurrence but also the environment interference, i.e. fading or interference in wireless channel or high bandwidth delay environment. Due to the limitations of Prime methodology, the Dual plays a more important actor to assist by providing more accurate and quick feedback. The congest control algorithm for *Dual* is implemented in routers gathering traffic flow information, such as flow numbers and traffic load, and sends implicitly or explicitly feedback to sender or receiver node for revising sending rate or making active queue management.

The Dual methodology, Active Queue Management (AQM), can be divided into two main categories including *closed-loop control* and *open-loop control* depending on whether the algorithm uses feedback information or not. For closed-loop control, the most well-known proposals are RED, Adaptive-RED (ARED) [6] and BLUE [7]. The RED's main idea is that using two predefined thresholds, minimum and maximum thresholds, to separate the queue length into the three congestion grades and adjusts the packet dropping rate according to different situations. The ARED dynamically adjust RED's thresholds based on the observed queue length and try to maintain the queuing delay within a target range. BLUE [7] uses packets loss and link-idle events as the critical factors to adjust packet dropping probability instead of the queue length. In the open-loop control, the major promising proposals are RAP [8], XCP [9] and its extended researches [10]. The main objective of this category is to achieve the incoming data rate that is equal to the output link capacity of router, and each traffic flow is allocated the same bandwidth and also ensures the lower queue sizes at the same time. This category can eliminate the high bandwidth-delay product network effect on the TCP's throughput, which is inversely proportional to the RTT, to satisfy the *TCP-friendly property* [3]. However, the above congestion control algorithms only adopt homogeneous fairness resource allocation method.

The works [11-13] alleviate this problem by modifying AQM architecture. In [11], the proposed algorithm rearranges the order of packets in the queue of router, dynamically adjusts packet dropping rate and the target queuing average size based on the packet arrival time, incoming traffic's requirements and delay hint. The work, in [12], uses three levels of RED to emulate the class-based architecture that each level sets parameters according to different traffic requirements and based on that to determine the incoming packet is accepted or not. Research in [13], it provides different dropping rate adjusting algorithms for TCP and UDP with TCP-Friendly property for

the diversity traffic characteristic. However, above surveyed algorithms cannot satisfy the delay and throughput requirements at the same time since it only adopt one-queue architecture for all types of traffic.

Recently, the multimedia streaming applications, such as IPTV and Video conference, emerge to be the one of main traffic sources with low tolerant of delay and jitter. Usually, the scalable layered coding (SVC) [14] technique is used to increase the end-user utility under diversified environments. The SVC is an extension of H.264/AVC which uses layered structure scheme to generate multilayer with one base layer and several enhancement layers. Therefore, a receiver can subscribe an appropriate one scenario based on the network status and required transmission quality. In order to ensure the efficient use of network resources, this kind of applications adapt the multicast technique to deliver the contents. Besides, the multicast service over wireless environment results in not only enhancing resource efficiency but also reducing transmission power consumption due to the *wireless multicast advantage* [15] property. Accompanying the wireless technique is mature enough to be the last mile solution; the IPTV multicast services under wire and wireless environment such as the integration EPON and WiMAX [16] will become a trend. However, all of the proposed active queue management mechanisms do not take account of the multicast services, and the proposed algorithms with an assumption that the same weight for unicast and multicast connections; however this is unfair for the multicast connection which will cause the poor system performance in the light of the entire network average video quality.

Several researches [17-19] utilize the video coding technique to improve throughput and end-user utility when congestion occurs. In view of the video coding technique, the literature [17], one of XCP extending research, adds addition header field to record how many resource has been assigned to each flow so that the sender can know which layers should be delivered. In the literatures [18-19], they support different QoS by using priority dropping queue management and packet marking technique. In [20], the author adopts the SCED+ scheduler for guaranteeing delay requirement. In these researches [12,19-20], the proposed various algorithms satisfy the QoS requirements by utilizing scheduler and marking technique; however, too complex and incurring additional process overhead to the router induces the impact on bottleneck obviously.

In summary, current AQM algorithms have the following problems: 1) Most of algorithms cannot achieve the delay and throughput requirements at the same time. On the other hand, some AQM algorithms can satisfy each traffic type's requirement, but those algorithms are too complex and unsuitable for high traffic load which causes the heavy computing overhead. 2) Above mentioned algorithms barely consider the video traffic characteristic that only adopt homogeneous fairness bandwidth allocation policy. 3) They do not take account of multicast service property, thus it leads to low bandwidth efficiency and poor system average video quality. 4) Current AQM algorithms only utilize adjusting packet dropping rate to overcome the congestion problem; however, it should not only adjust packet dropping rate but also consider congestion level and the AQM will be more efficient for reacting to various traffic loads, 5) Most AQM algorithms do not have the adaptability, and those algorithms have to be

trained or adjust a set of parameters to meet the diverse traffic load and router link capacity. It is a challenge to overcome congestion problem to consider the video coding technique, bandwidth efficiency and different traffic's QoS requirement for more outstanding performance.

In this paper, the Traffic Sensitive Active Queue Management (TSAQM) scheme is proposed to overcome those problems. Several objectives of this proposed scheme are described as follows: *first*, a Dual methodology congestion control algorithm is proposed to meet the QoS requirement of different services by using multi-queues multi-thresholds mechanism cooperating with weight-based scheduler algorithm; *second*, it achieves high end-user utility for video service; *third*, it considers the multicast as well as unicast proprieties to meet inter-class fairness; *fourth*, it has the ability to adaptively adjust parameters of TSAQM according as time-varying traffic loads. The rest of paper is organized as follows. Section 2 presents the system architecture. The system performance is analyzed and discussed in section 3. Finally, the paper ends with our conclusions in section 4.
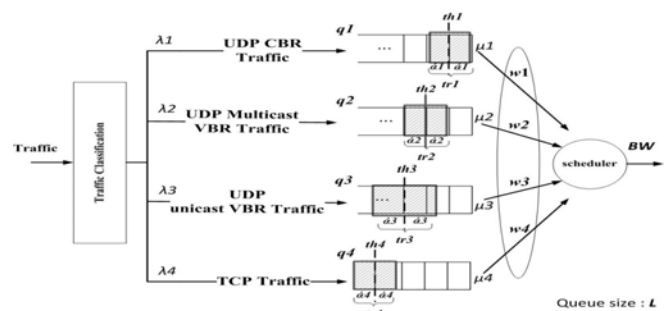
II. SYSTEM ARCHITECTURE

Fig. 1 System architecture

The system architecture, as shown in Fig. 1, has four types of traffic including UDP CBR (constant bit rate) traffic, UDP VBR (variable bit rate) unicast traffic (UVBR), UDP multicast traffic with VBR (MVBR) and TCP traffic. The Traffic Sensitive Active Queue Management (TSAQM) with Dynamic Weight Allocate Scheme (DWAS) and Service Guarantee Scheme (SGS) is proposed for QoS-aware active queue management.

A. System Environment

Based on Fig. 1, the four-queues with four-thresholds and weight-based scheduler are proposed; in addition, four individual FIFO queues, $Q=\{q_1, q_2, q_3, q_4\}$, are set for different traffic classes, $T=\{t_1, t_2, t_3, t_4\}$, respectively, where the traffic class t_1 is the UDP traffic with CBR (B_{CBR}), the traffic classes t_2 and t_3 are the multicast and the unicast UDP traffic with VBR, and the traffic class t_4 is TCP traffic. For traffic types of VBR, each flow contains VL video layers and the bandwidth of each layer denotes as $LB=\{lb_1, lb_2, lb_3, …, lb_N\}$. The arrival rates and service rates for different traffic classes are $\lambda=\{\lambda_1, \lambda_2, \lambda_3, \lambda_4\}$ and $\mu=\{\mu_1, \mu_2, \mu_3, \mu_4\}$, and QoS requirement vector denotes as $R=\{r_1, r_2, r_3, r_4\}$, including delay, packet dropping rate and throughput.

Due to the performance of GRED-I is better than both RED and GRED [21,22], each queue applies GRED-I buffer management with threshold TH and threshold region TR for different traffic classes in the proposed TSAQM scheme, where threshold TH and threshold region TR denote the vector of each queue's threshold and threshold region, respectively. The purpose of threshold for different traffic classes, $TH=\{th_1, th_2, th_3, th_4\}$, is estimated to determine the packet dropping rate, and the threshold region for different traffic classes, $TR=\{tr_1, tr_2, tr_3,$

tr_4}, where the $tr_i=(th_i-\sigma_i,\ th_i+\sigma_i)$ with threshold range σ_i for different traffic classes, i=1, 2, 3, 4, is cooperated with *TH* to estimate suitable parameters for current traffic condition. Furthermore, in order to achieve effective resource utilization, the dynamic weight-based scheduler is adopted with weight for different traffic classes, $W=\{w_1,\ w_2,\ w_3,\ w_4\}$, as a scheduler mechanism.

B. Traffic Sensitive Active Queue Management (TSAQM)

The TSAQM has two main tasks: one is to allocate resource with fairness and high end-user utility in the Dynamic Weight Allocate Scheme (DWAS), and the other is to determine the satisfactory threshold (*TH*) and threshold region (*TR*) in the Service Guarantee Scheme (SGS).The DWAS is used to allocate bandwidth and adjust the weights mechanism of *W* for different traffic classes to achieve better resource utilization. Differential service fairness delimitation, called *Differ-TCP-Friendly*, is proposed to provide the minimum requirement of each class first and then distribute residue bandwidth for TSAQM. Then, the thresholds (*TH*) and threshold regions (*TR*) are determined by a one-dimension Markov-chain model in the SGS to adjust thresholds precisely to meet the QoS requirement of each traffic class.

Fig. 2 DWAS algorithm

Fig. 3 DRBS algorithm

C. Dynamic Weight Allocate Scheme (DWAS)

The DWAS has two phases: the first phase is to satisfy the minimum throughput requirement of each traffic class, and the second phase is to use the DRBS (Distribute Residue Bandwidth Scheme) to distribute the residue bandwidth with Differ-TCP-Friendly to all traffic classes, except the CBR traffic.

The DWAS distributes bandwidth to traffics $T=\{t_1, t_2, t_3, t_4\}$ based on the traffic priority and current active connections, $N=\{n_1, n_2, n_3, n_4\}$, for different traffic classes. The traffic classes t_1, t_2, and t_3 have the property that the data rate is constant or staircase-like bit rates, and traffic class, t_4, is throughput sensitive without minimum throughput requirement. However, in order to satisfy the *Differ-TCP-Friendly*, the DWAS allocates bandwidth to traffic class, t_4, by the assumption that the minimum requirement of traffic class, t_4, is the maximum throughput requirement of CBR and VBR.

The basic bandwidth allocation unit for VBR traffic is the bandwidth of each layer of SVC. While all layer's bandwidth are met or the residue bandwidth is not enough for any class's requirement, the resource will be divided equally to all traffic classes, except CBR traffic, based on the proportion of current active connection(s). The detail procedure of DRBS algorithm is shown in the Fig. 3.

D. Service Guarantee Scheme (SGS)

Fig. 4 SGS algorithm

Fig. 5 QV and MBL functions

The algorithm of SGS is shown Fig. 4. If the incoming traffic class, t_i, is delay sensitive traffic, it checks the trend flag, tf_i, is in decreasing trend (higher than the upper bound) or increasing trend (less than lower bound of threshold region). When the trend flag indicates that the situation is in decreasing, then the threshold th_i, subtracts ε_{delay}; otherwise, adds ε_{delay}, where ε_{delay} is the adjusting *TH* unit; then, the SGS verifies the adjustment outcome using the Quality Verification (QV) function to verify current threshold setting whether meets the required QoS or not. The details of QV function is shown in Fig. 5. When the traffic class is throughput sensitive, it uses the Modify BLUE LIKE (MBL) function, shown in Fig. 5, to be

responsive to current traffic load by adjusting the packet dropping rate. According to the QV function, it compares the QoS requirements of ith traffic class, r_i, to *TP*, *DT* and *PD*, respectively, for verifying current *TH* setting. In case of the requirements cannot be met, the SGS chooses the minimum value as the *TH* setting value for guaranteeing delay requirement. According to the MBL function, if the current queue size is longer than *TR* or equal to *L*, the th_i subtracts $\varepsilon_{throughput}$; otherwise, adds $\varepsilon_{throughput}$ while there is no packet arrival in *Freeze_time*, where $\varepsilon_{throughput}$ is the adjusting *TH* unit. Finally, the variation of connection (CV) is used as a main critical factor based on varying packet queue for each connection to determine the threshold range (σ_i).

$$CV = \frac{1}{PN} \sum_{k=1}^{PN} (x_k - \delta_i)^2 \times \frac{1}{PN} \sum_{k=1}^{PN} (\theta_k - \rho_i)^2 \qquad (1)$$

where δ_i and ρ_i are the average number of connection and service rate of traffic class i, respectively, χ_k and θ_k are the number of current active connection and arrival rate of kth record, respectively, and *PN* is the history data quantity from previous update to the present time.

The TSAQM monitors the system condition, and based on the result of threshold region information determines the proper moment to update the system parameters. This can avoid unnecessary initiation, since there is no additional bandwidth for lower priority traffic, and the initial timing is defined in Fig. 6.
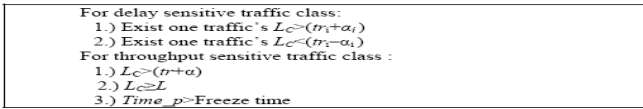

For delay sensitive traffic class:
  1.) Exist one traffic's $L_C > (r_i + \alpha_i)$
  2.) Exist one traffic's $L_C < (r_i - \alpha_i)$
For throughput sensitive traffic class :
  1.) $L_C > (r_i + \alpha)$
  2.) $L_C \geq L$
  3.) $Time\_p > Freeze\ time$

Fig. 6 Initial timing

E. Dynamic Weight Allocate Scheme (DWAS)

The one-dimension Markov-chain model, shown in Fig. 7, is adopted to estimate the throughput (*TP*), delay time (*DT*) and packet dropping rate (*PD*), which is a *M/M/1/L/th* queuing system under the First-In-First-Out (FIFO) service discipline. The traffic arrival follows a Poisson process with an average arrival rate λ and the service time is exponentially distributed with mean $1/\mu$ and the total system capacity is *L* with one threshold.
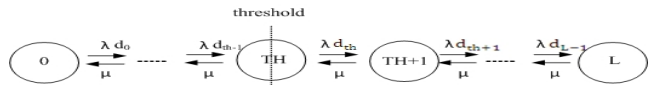

Fig. 7 One-dimension Markov-chain model

$$d_i = \begin{cases} 1, & 0 \leq i \leq th \\ 1 - \left(1 - \frac{i - th + 1}{L - th + 1}\right) d_{max} & th \leq i \leq L \end{cases} \qquad (2)$$

Refer to [23], the packet dropped behavior can be regarded as the trend to decrease arrival rate. A linear dropping equation, d_i, (obtained from Eq. 2) is used to represent as packet dropped behavior and the maximum dropping probabilities, d_{max}, is 1. Let P_i is the probability of state i, $0 \leq i \leq L$, and based on the Fig. 8, the balance equations, Eq. 3, Eq. 4 and Eq. 5 can be obtained.

Based on M/M/1/L model and Little's formula, the throughput, delay time and packet dropping rate can be obtained from Eq. 9, Eq. 10, and Eq. 11.

$$d_0 \times \lambda \times P_0 = \mu \times P_1 \qquad (3)$$

$$(d_i \times \lambda + \mu) \times P_i = (d_{i-1} \times \lambda) \times P_{i-1} + \mu \times P_{i+1} \quad 1 \leq i \leq L \qquad (4)$$

$$\mu \times P_L = (d_{L-1} \times \lambda) \times P_{L-1} \qquad (5)$$

$$\sum_{i=0}^{L} P_i = 1 \qquad (6)$$

The probability P_0 and P_i can be expressed as follows:

$$P_i = \left(\prod_{k=0}^{i-1} \frac{d_i \times \lambda}{\mu}\right) P_0 \qquad 1 \leq i \leq L \qquad (7)$$

$$P_0 = \left[1 + \sum_{i=1}^{L} \left(\prod_{j=0}^{i-1} \frac{d_j \times \lambda}{\mu}\right)\right]^{-1} \qquad (8)$$

$$TP = \sum_{i=0}^{L-1} (P_i \times d_i \times \lambda) \qquad (9)$$

$$DT = \left(\sum_{i=0}^{L} P_i \cdot i\right) \Big/ ((1 - P_0) \times \mu) \qquad (10)$$

$$PD = \sum_{i=0}^{L} P_i \times (1 - d_i) \qquad (11)$$

III. PERFORMANCE ANALYSIS

Table 1 System Environment Parameters

Environment Variable	value
Router Queue Size	100 (packet size)
Router Number	2
Node Number	10
Link Capacity	10Mbps
Simulation time	1200 seconds
Maximum Dropping Rate	1.0
ε_{delay}	1
$\varepsilon_{throughput}$	1
Scheduler	Weighted fair queuing

Table 2 Parameters of traffic class

Traffic class	Mean of duration(s)	Data rate (kbps)	Latency Guideline(ms)	Dropping Guideline
CBR	210	64	150	0.03
Multicast VBR	360	46~240	150	N/A
Unicast VBR	360	46~240	150	N/A
FTP	180	N/A	N/A	N/A

Table 3 Video information

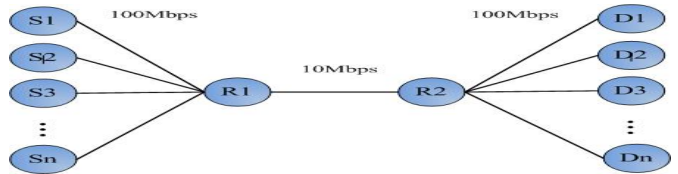| Layer	Frame size	Frame rate (frame/sec)	Data rate (kbps)
1	176×144	0.9375	46
2	176×144	1.8750	26
3	176×144	3.7500	38
4	176×144	7.5000	54
5	176×144	15.0000	76

Fig. 8 Simulation topology

In this section, the network simulator 2 (NS-2) is used to estimate the performance of TSAQM, and adopt the dumbbell topology as the simulation topology, shown in Fig. 8, which there are n sources, n destinations, and two routers [6]. The bandwidth between source (or destination) and router is with 100Mbps, and the bandwidth between routers is with 10 Mbps. The buffer space at router is set to 100 packets as shown in Table 1. The traffic arrival rate follows the Poisson process and the data rate of the CBR [24], the VBR video source is the "HARBOUR" which is generated by JSVM [25] and the TCP traffic is generated as the FTP Traffic Model [24] are shown in Table 2 and Table 3.

Based on Fig. 8, the router R1 is chosen to evaluate system performance in terms of packet dropping rate, average delay time and connection throughput for two simulation cases for different CBR and MVBR traffic arrival rates, respectively.

A. TSAQM for different CBR traffic arrival rates

In the case, the arrival rate of CBR is varied from 0.06 to 0.14(flows/sec), and the others are fixed and set to be 0.065(flows/sec).Fig. 9(a), 9(b) and 9(c) show the average packet dropping rate, delay time and connection throughput, respectively, for different CBR arrival rates.

In Fig. 9(a), it shows that the packet dropping rate of the CBR, MVBR, and UVBR for different CBR arrival rates. The average packet dropping rate of the CBR is always lower than the others and is maintained at about 0.005. This shows that the proposed TSAQM can achieve the dropping guideline of CBR

traffic. The packet dropping rate of MVBR is lower than UVBR due to the DRBS distributing residue bandwidth to MVBR through threshold adjustment. When the UVBR dropping rate is about 15%, it means that the DRBS does not allocate the bandwidth to the 5th layer video stream. In the case of the arrival rate of CBR is between 0.085(flows/sec) and 0.095(flows/sec), the UVBR dropping rates is about 23% which means the DRBS does not allocate the bandwidth to the 4th layer video stream. The UVBR dropping rate is between 23% and 30%, in the case of the arrival rate of CBR is between 0.15(flows/sec) and 0.105(flows/sec), which means the DRBS does not allocate the bandwidth to the 3rd layer video stream. Similarly, in the case of the arrival rate of CBR is higher than 1.0(flows/sec), the 5th layer video stream will be dropped for MVBR.
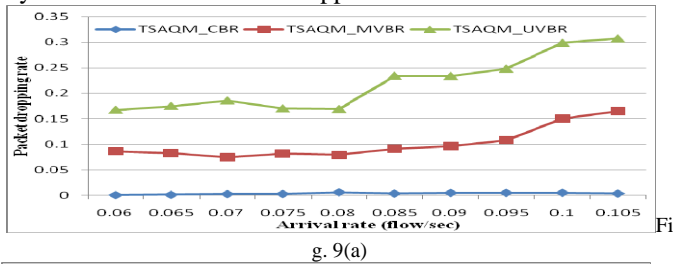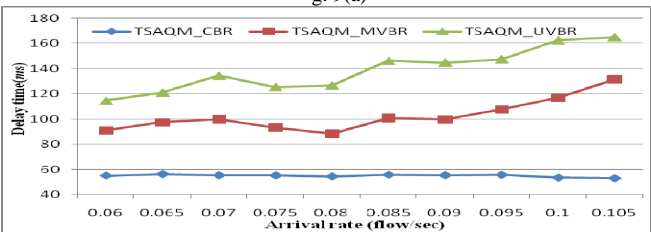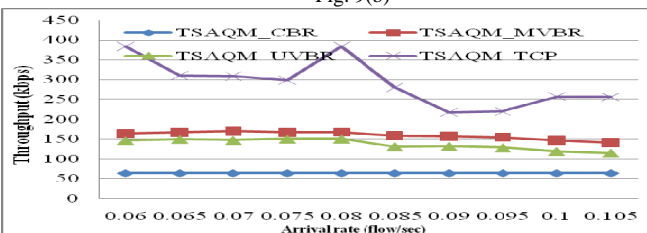

Fig. 9(a)

Fig. 9(b)

Fig. 9(c)

Fig. 9 (a) Packet dropping rate, (b) Delay time of the CBR, MVBR and UVBR (c) Throughput of the CBR, MVBR, UVBR and TCP for different CBR arrival rates.

Fig. 9(b) shows the delay time of the CBR, MVBR and UVBR for different CBR arrival rates that the proposed TSAQM can achieve the latency guideline of CBR and MVBR traffic. For the same reason, the delay time of CBR is the lowest and UVBR is the highest by the DRBS distributing strategy. When the arrival rate of CBR is higher than 0.1(flows/sec), the delay time of UBVR is slightly higher than 150*ms*. Besides, there are two reasons for the unstable delay time. First, the frame variation of the "HARBOUR" is more intense that means the variation of entering queue rate is higher than smooth one. Secondly, the proposed TSAQM uses the TR to avoid the reinitiate since burst traffic arriving that will result in the higher TR value and cause the higher delay than the TSAQM estimated especially for heavy load case.

Fig. 9(c) shows the average connection throughput of CBR, MVBR and UVBR, and total throughput of TCP for different CBR arrival rates. This shows that the proposed TSAQM can achieve the required transmission rate for CBR, MVBR and UVBR. The mean throughput of CBR is about 64kbps for different CBR arrival rates. Besides, in the case of the arrival rate of CBR is 0.085(flows/sec), the throughput of TCP increases obviously due to the 4th layer packets of UVBR are

dropped referring to Fig. 10(a). There is the same phenomenon for the arrival rate of CBR is 0.1(flows/sec) case.

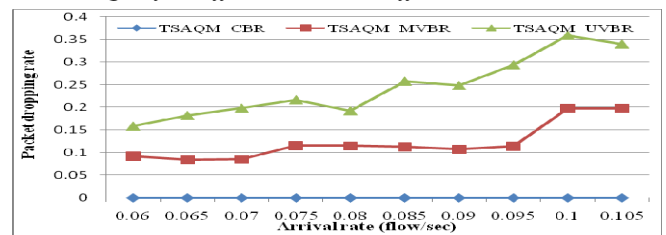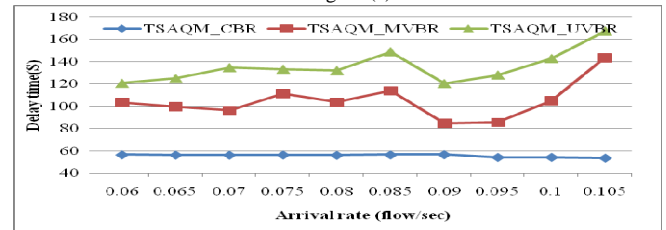### B. TSAQM for different MVBR traffic arrivals

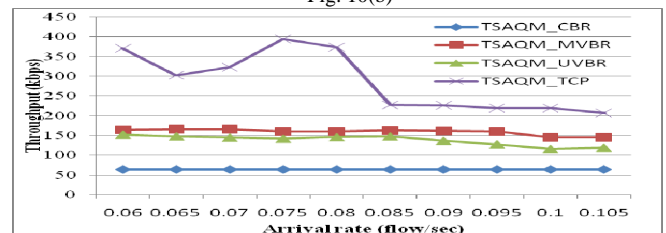Fig. 10(a)

Fig. 10(b)

Fig. 10(c)

Fig. 10 (a) Packet dropping rate, (b) Delay time of the CBR, MVBR and UVBR (c) Throughput for different MVBR arrival rates.

In the case, the arrival rate of MVBR is varied from 0.06 to 0.14(flows/sec), and the others are fixed and set to be 0.065(flows/sec). Fig. 10(a), 10(b) and 10(c) show the average packet dropping rate, delay time and connection throughput, respectively, for different MVBR arrival rates for the proposed TSAQM and GRED-I.

Comparing Fig. 10(a) with Fig. 9(a), the packet dropping rate of CBR, MVBR and UVBR in Fig. 9(a) is higher than that in Fig. 10(a) which is due to the data rate of MVBR is higher than CBR. Besides, the packet dropping rate increases more rapidly than in Fig. 9(a) for the UVBR when the MVBR arrival rate is increased. However, the impact on MVBR is slight for increasing MVBR arrival rate. Fig. 9(a) also signifies that in the cases of the arrival rate of MVBR at 0.085(flows/sec) and 0.1(flows/sec), the DRBS does not allocate the bandwidth to the 4th and the 3rd layer video stream, respectively, for MVBR.

Fig. 10(b) shows the delay time of the CBR, MVBR, and UVBR for different MVBR arrival rates. This shows that the proposed TSAQM can achieve the latency guideline of CBR and MVBR traffic through DRBS distributes residue bandwidth to them first. Comparing Fig. 10(b) with Fig. 10(b), it has unstable results in Fig. 10(b) in case of arrival rate between 0.08(flows/sec) and 0.1(flows/sec). The reason is the same as varying CBR arrival rate case that is the impact on frame variation and the TR will be obvious since the MVBR traffic increasing. Due to the DWRR adopts packet based scheduler, the DWAS will be impacted since the packet size is various greatly, and it is more obvious than in Case 1.

Fig. 10(c) shows the average connection throughput of CBR, MVBR and UVBR, and total throughput of TCP for different MVBR arrival rates. This also shows that the proposed TSAQM can achieve the required transmission rate for CBR, MVBR and UVBR. The mean throughput of CBR is about 64

kbps for different MVBR arrival rates. According to Fig. 11(a), most of packets of the 5th layer video stream for UVBR are dropped when arrival rate is 0.070(flows/sec), therefore, TCP got more bandwidth. In the case of 0.075(flows/sec), due to a few of packets of the 4th layer video stream for UVBR and more packets of the 5th layer video stream for MVBR are dropped, the TCP got the highest throughput. Since the arrival rate is higher than 0.085(flows/sec), more packets of MVBR and UVBR are dropped, the throughput of TCP is decreased due to increasing total UVBR as the UVBR arrival rate increases.
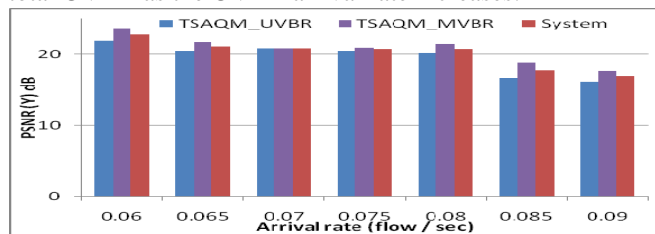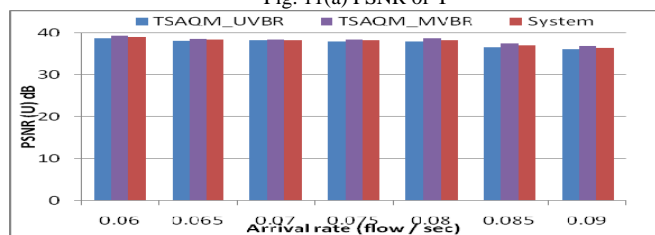

Fig. 11(a) PSNR of Y
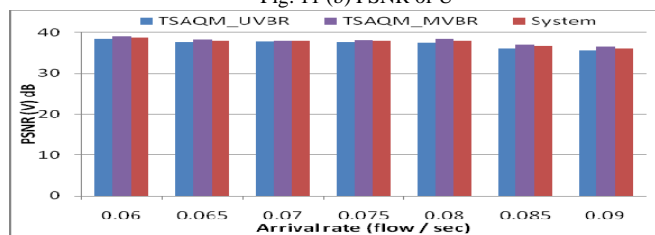

Fig. 11 (b) PSNR of U

Fig. 11 (c) PSNR of V
Fig. 11 the PSNR of (a) Y, (b) U and (c) V for MVBR, UVBR and system for different MVBR rates.

C. Results of peak of SNR (PSNR)

In order to estimate the video quality, the arrival rate of MVBR is varied from 0.06 to 0.14 (flows/sec), the others are fixed and set to be 0.065(flows/sec).

Fig. 11(a), (b), and (c) show the peak of SNR (PSNR) of Y, U and V, respectively, for MVBR, UVBR and system for different MVBR rates. According to Fig. 11(b) and (c), the variation of PSNR for U and V is about 2.5dB (i.e. between 36.5dB and 39dB). The decreased is more obvious for Y under increasing CBR arrival rate, and the variation is about 6dB, referring to Fig. 11. In addition, the values of MVBR are higher than UVBR for all cases due to more packets of UVBR are dropped.

IV. CONCLUSION

In this paper, the Traffic Sensitive Active Queue Management (TSAQM) is proposed to overcome problems of current AQM algorithms. Based on simulation results, several objectives of this proposed scheme are achieved including by using multi-queues multi-thresholds mechanism cooperating with weight-based scheduler algorithm to meet the QoS requirement, high end-user utility for video service, considers the multicast, and adaptively adjust parameters of TSAQM according as time-varying traffic loads. Also it shows that the TSAQM can achieve the QoS requirement in time-varying Internet by adjusting the thresholds adaptively based on traffic situations.

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